

**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.

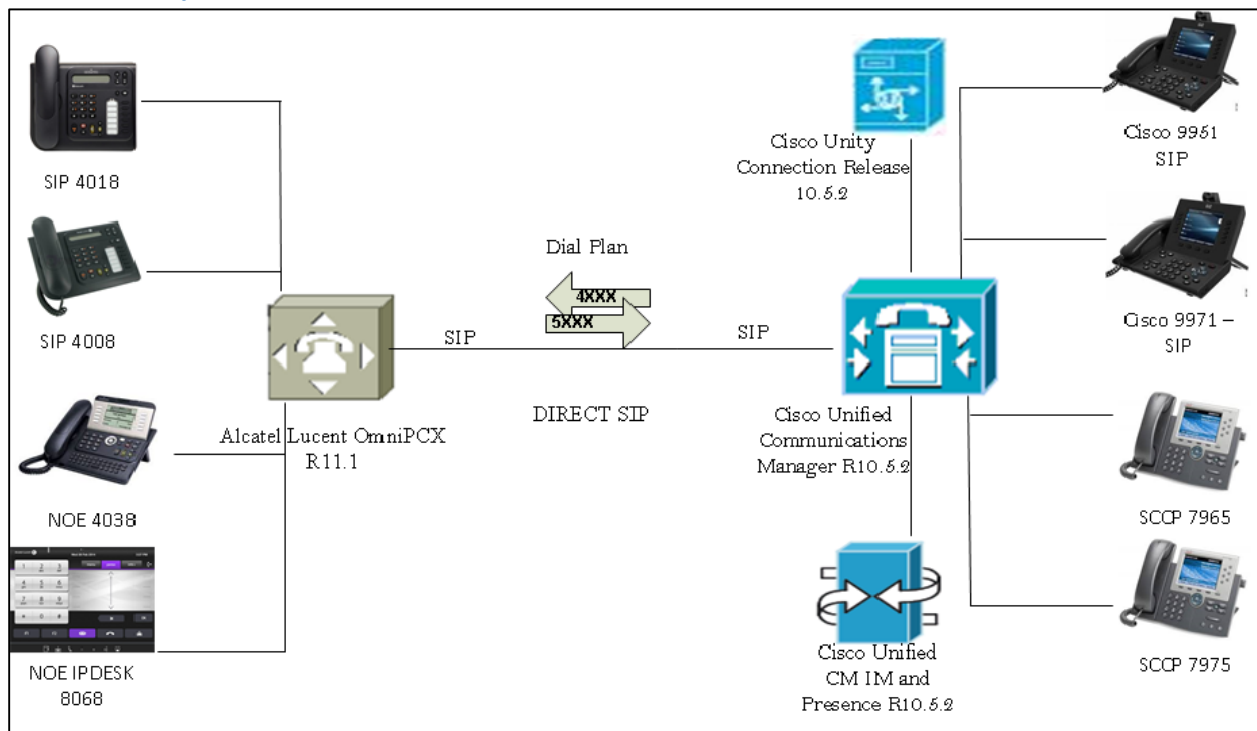
- For call forwarding scenarios where the Alcatel station is the forwarding station, it is required that the Cisco UCM have the “Redirect by Application” checkbox enabled in the SIP Profile used by the SIP Trunk to the Alcatel PBX.
- Retrieval of a parked call on the Alcatel PBX from the Cisco UCM is 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



## 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

```
# The role of the CPU is MAIN
Application software identity
R11.1-11.301-27-us-c0s1
Business identification: R11.1
Release:
DELIVERY 11.301
Patch identification: 27
Dynamic patch identification: none
Country: us
Cpu: c0s1
ACD VERSION
  release : 11
  bug_fixing : 2
  protocol_id : 92
  version_dy_hr_stat : 12
(101)cpua>
```

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.

```
(101)cpua> spadmin
ooo Reading string-file /DHS3bin/oneshot/mtcl/objects.US0 ...
ooo Reading string-file /DHS3bin/oneshot/mtcl/spadmin.dct ...
ooo Reading string-file /DHS3bin/oneshot/mtcl/err2lov.dct ...

Display current counters ..... 1
Display active file ..... 2
Check active file coherency ..... 3
Install a new file ..... 4
Read the system CPUID ..... 5
CPU-Ids management ..... 6
Display active and new file ..... 7
Display OPS limits ..... 8
Display ACK code ..... 9
Exit ..... 0
choice :
```

### 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.

```
lqSelect an objectqqqqqqqqqqqqqqqqqqqqqqk
x
x > Shelf x
x Media Gateway x
x PWT/DECT System x
x System x
x Translator x
x Classes of Service x
x Attendant x
x Users x
x Users by profile x
x Set Profile x
x Groups x
x Speed Dialing x
x Phone Book x
x Entities x
x Trunk Groups x
x External Services x
x Inter-Node Links x
x X25 x
x DATA x
x Applications x
x Specific Telephone Services x
x ATM x
x Events Routing Discriminator x
x Security and Access Control x
x IP x
x SIP x
x DHCP Configuration x
x Alcatel-Lucent 8&9 Series x
x SIP Extension x
x Encryption x
x Passive Com. Server x
x SNMP Configuration x
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

## Configure IP Domain

Navigation: IP → IP domain

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

Click ctrl+v to complete.

```
lqReview/Modify: IP domainqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                                               x
x           Node Number (reserved) : 101                                                       x
x           Instance (reserved) : 1                                                            x
x           IP Domain Number : 0                                                                x
x                                                                                               x
x           IP Domain Name : lab.tekvizion.com         x
x           Country + Default                                                                   x
x           Intra-domain Coding Algorithm + Without Compression                                x
x           Extra-domain Coding Algorithm + Without Compression                                x
x FAX/MODEM Intra domain call transp + YES                                                    x
x FAX/MODEM Extra domain call transp + YES                                                    x
x           G722 allowed in Intra-domain + NO                                                  x
x           G722 allowed in Extra-domain + NO                                                  x
x           Tandem Primary Domain : -1                                                         x
x           Domain Max Voice Connection : -1                                                  x
x           IP Quality of service : 0                                                           x
x           Contact Number : -----                                                          x
x           Backup IP address : -----                                                         x
x           Trunk Group ID : -1                                                                 x
x IP recording quality of service : 0                                                           x
x           Time Zone Name + System Default                                                    x
x           Calling Identifier : -----                                                        x
x           Supplement. Calling Identifier : -----                                           x
x           SIP Survivability Mode + NO                                                         x
x                                                                                               x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

## 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

```

lgReview/Modify: Trunk Groupsqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x      Node Number (reserved) : 101                                x
x      Trunk Group ID : 1                                          x
x
x      Trunk Group Type + T2                                       x
x      Trunk Group Name : SIP                                     x
x      UTF-8 Trunk Group Name : -----                          x
x      Number Compatible With : -1                               x
x      Remote Network : 15                                       x
x      Shared Trunk Group + False                                 x
x      Special Services + Nothing                               x
x      Node number : 1                                           x
x      Transcom Trunk Group + False                             x
x      Auto.reserv.by Attendant + False                       x
x      Overflow trunk group No. : -1                            x
x      Tone on seizure + True                                  x
x      Private Trunk Group + False                             x
x      Q931 Signal variant + ABC-E                             x
x      SS7 Signal variant + No variant                         x
x      Number Of Digits To Send : 0                             x
x      Channel selection type + Quantified                     x
x      Auto.DTMF dialing on outgoing call + YES                x
x      T2 Specification + SIP                                  x
x      Homogenous network for direct RTP + NO                  x
x      Public Network COS : 31                                  x
x      DID transcoding + False                                 x
x      Can support UUS in SETUP + True                         x
x      Associated Ext SIP gateway : 2                           x
x
x      Implicit Priority                                         x
x
x      Activation mode : 0                                       x
x      Priority Level : 0                                         x
x
x      Preempter + NO                                           x
x      Incoming calls Restriction COS : 10                     x
x      Outgoing calls Restriction COS : 10                     x
x      Callee number mpt1343 + NO                               x
x      Overlap dialing + NO                                     x
x      Call diversion in ISDN + NO                              x
x
x      ]qqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq[

```

### 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.

```
lq[ 20 ] Instances: Trusted IP Addressesqk
x                                     x
x > 10.70.50.2                       x
x 10.70.50.6                         x
x 10.70.50.15                        x
x 10.70.50.67                        x
x 172.16.29.83                       x
x 10.64.202.53                       x
x 172.16.31.93                       x
x 10.80.10.2                         x
x 172.16.31.94                       x
x 172.16.26.30                      x
x 10.80.10.3                         x
x 10.80.10.4                         x
x 172.16.29.210                     x
x 10.70.50.7                         x
x 172.16.31.169                     x
x 10.64.1.163                       x
x 10.70.50.133                      x
x 10.80.10.23                       x
x 10.80.10.5                         x
x 206.165.51.46                     x
x                                     x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqq
```

### 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

```

lqReview/Modify: SIP Gateway
Node Number (reserved) : 101
Instance (reserved) : 1
Instance (reserved) : 1

SIP Subnetwork : 10
SIP Trunk Group : 1
IP Address : 10.70.50.6
Machine name - Host : tekvisionoxe
SIP Proxy Port Number : 5060
SIP Subscribe Min Duration : 1800
SIP Subscribe Max Duration : 86400
Session Timer : 1800
Min Session Timer : 1800
Session Timer Method + RE INVITE
DNS local domain name : lab.tekvision.com
DNS type + DNS A
SIP DNS1 IP Address : 10.64.1.3
SIP DNS2 IP Address : -----
SDP in 18x + True
Cac SIP-SIP + True
INFO method for remote extension + False
Dynamic Payload type for DTMF : 101

```

### **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

```
lqReview/Modify: SIP Ext Gatewayqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqkk
x
x Node Number (reserved) : 101 x
x Instance (reserved) : 1 x
x SIP External Gateway ID : 2 x
x
x Gateway Name : CUCM x
x SIP Remote domain : 10.80.10.3 x
x PCS IP Address : ----- x
x SIP Port Number : 5060 x
x Transport type + UDP x
x Belonging Domain : lab.tekvizion.com x
x Registration ID : ----- x
x Registration ID P_Asserted + False x
x Registration timer : 0 x
x SIP Outbound Proxy : ----- x
x Supervision timer : 0 x
x Trunk group number : 1 x
x Pool Number : -1 x
x Outgoing realm : ----- x
x Outgoing username : ----- x
x Outgoing Password : ----- x
x Confirm : ----- x
x Incoming username : ----- x
x Incoming Password : ----- x
x Confirm : ----- x
x RFC 3325 supported by the distant + True x
x DNS type + DNS A x
x SIP DNS1 IP Address : ----- x
x SIP DNS2 IP Address : ----- x
x SDP in 18x + True x
x Minimal authentication method + SIP None x
x INFO method for remote extension + False x
x To EMS + False x
x SRTP + RTP only x
x Ignore inactive/black hole + True x
x Contact with IP address + False x
x Dynamic Payload type for DTMF : 101 x
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```









### 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

```
lqReview/Modify: Network Routing Tableqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x      Node Number (reserved) : 101      x
x      Instance (reserved) : 1          x
x      Network Number : 10              x
x
x      Rank of First Digit to be Sent : 1 x
x      Incoming identification prefix : ----- x
x      Protocol Type + ABC_F          x
x      Numbering Plan Descriptor ID : 10  x
x      ARS Route list : 5              x
x      Schedule number : -1            x
x      ATM Address ID : -1            x
x      Network call prefix : -----    x
x      City/Town Name : -----        x
x      Send City/Town Name + False    x
x      Associated Ext SIP gateway : 2    x
x      Enable UTF8 name sending + True x
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

### 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.

```
lqReview/Modify: Prefix Planqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x      Node Number (reserved) : 101      x
x      Instance (reserved) : 1          x
x      Number : 5                        x
x
x      Prefix Meaning + Routing No.     x
x      Network Number : 10              x
x      Node Number/ABC-F Trunk Group : 1 x
x      Number of Digits : 4             x
x      Number With Subaddress (ISDN) + NO x
x      Default X25 ID.pref. + NO       x
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

**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.

```
lqReview/Modify: Prefix Planqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x      Node Number (reserved) : 101      x
x      Instance (reserved) : 1          x
x      Number : 7                        x
x
x      Prefix Meaning + Routing No.     x
x      Network Number : 10              x
x      Node Number/ABC-F Trunk Group : 1 x
x      Number of Digits : 4             x
x      Number With Subaddress (ISDN) + NO x
x      Default X25 ID.pref. + NO       x
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

## 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

```
lqReview/Modify: Dynamic State Userqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x      Node Number (reserved) : 101      x
x      Directory Number : 4004          x
x      Instance (reserved) : 1          x
x
x      Forward + No forward            x
x      Forward Directory Number : 4004   x
x      Secondary Line Forward + No forward x
x      Secondary Line Number Forward : 4004 x
x      Do Not Disturb + False          x
x      Lock + False                    x
x      Busy Camp-on + True              x
x      Overflow on associate + False    x
x      Overfl.busy to assoc.set + false x
x      Associated Set No. : 4004        x
x      Reset Charge Counter + False    x
x      Consistent with Identity + True  x
x      User Type + Administrative       x
x      State of the set + Out of order  x
x      SIP Survivability Mode + None    x
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

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)

```
Review/Modify: Entities
Node Number (reserved) : 101
Entity Number : 0
Name : ENTITY_0
UTF-8 Name : -----
Attendant Group Manager : -1
Priority : NO
Emergency call to attd : NO
Traffic Overflow : Disallowed
Installation No. (ISDN) : -----
Supplement.Install.No. (ISDN) : 3038356006
Caller ID Secret : Yes
AdvOfCharg2 requests (AOC2) : NO
AdvOfCharg3 requests (AOC3) : NO
Auto. Locking : 0
Voice Mail Box No.for attendt : -----
Trunk Group ID : 0
External Callback Table : 0
Call Distribution
```

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.

```

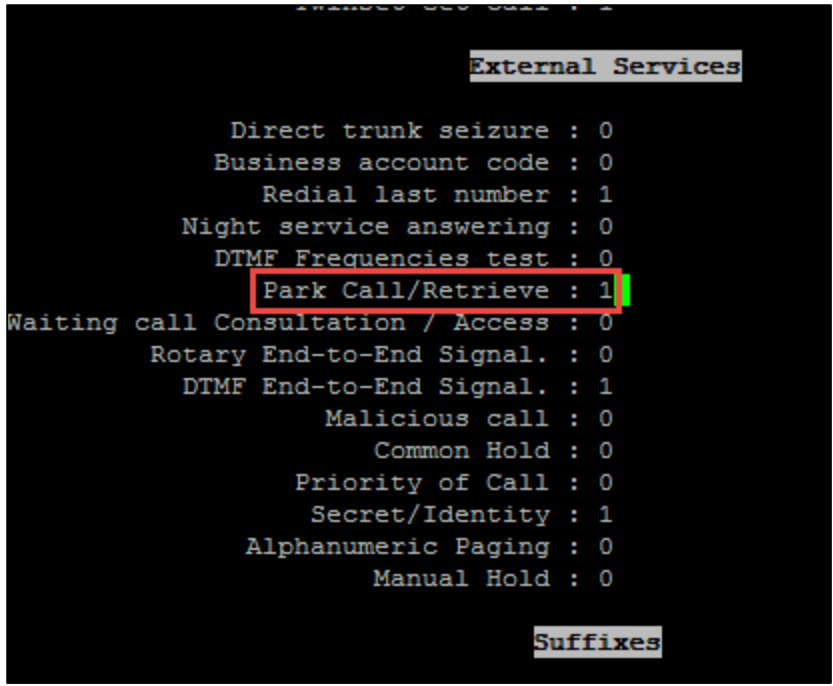
lgReview/Modify: Phone Features COSqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x      Node Number (reserved) : 101      x
x      Instance (reserved) : 1          x
x      Phone Features COS : 1           x
x
x      Phone COS Name : █-----      x
x
x      Rights
x
x      Prot.against dir.call pickup : 0   x
x      Protected against all barge-in : 0 x
x      Protected against set barge-in : 0 x
x      Outgoing calls only : 0           x
x      Forward to external No. : 1       x
x      Prot.against multi-1 ringing : 0   x
x      Protected against forwarding : 0   x
x      Protected (against barge-in, etc.) : 0 x
x      Prot.against call announc. : 0     x
x      Remote wake-up/appointment : 0    x
x      Auto call back on busy trk-grp : 0 x
x      Transfer on no answer : 1         x
x      ISDN remote charge service : 0    x
x      Bypass on forwarding : 0          x
x      Prot.against bypass onforward : 0  x
x      Interphony : 1                   x
x      Secret Code, Repertory Key : 0    x
x      Night Serv.Answ.Pick up : 0       x
x      Night Serv.Direct call pick-up : 0 x
x      Attendant Call Privil.on PAI : 0  x
x      Busy priv.to public overfl. : 0   x
x      Server-Minitel PC : 0             x
x      Prot.against Priv.Call : 0        x
x      Prot.against.Rem.Forward. : 0     x
x      Beep On Ext.Call : 0             x
x      O/S private to public overflow : 0 x
x      Transfer outgoing - incoming : 1  x
x      Transfer Outgoing-Outgoing : 1    x
x      PCX Calls Follow Ext. forwarding : 1 x
x      Mask ID.name Only for ext.calls : 0 x
x      Ringing tone In Handset : 0      x
x
mqppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppppj

```

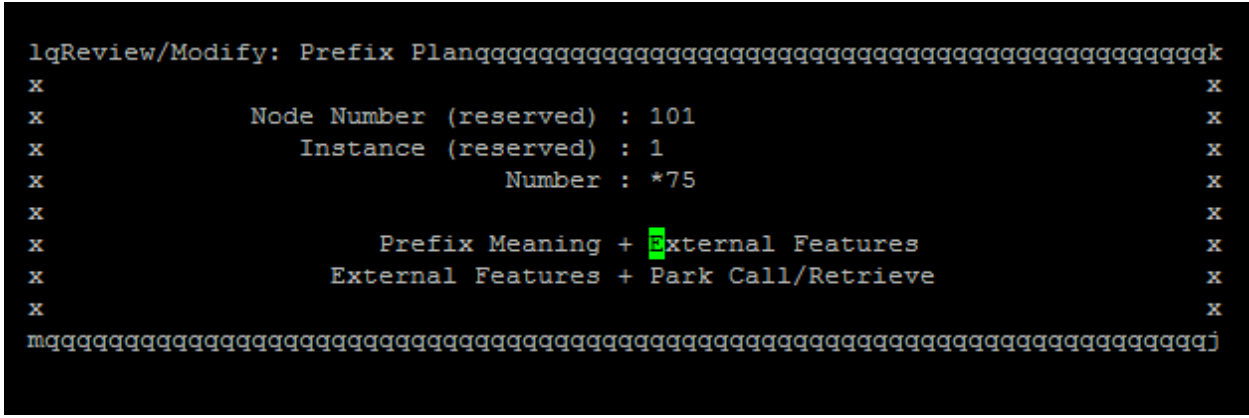
**Configure Call Park**

- 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

```

lqReview/Modify: IP domainqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x
x          Node Number (reserved) : 101              x
x          Instance (reserved) : 1                 x
x          IP Domain Number : 0                    x
x
x          IP Domain Name : lab.tekvizion.com        x
x              Country + Default                   x
x      Intra-domain Coding Algorithm + Without Compression x
x      Extra-domain Coding Algorithm + With Compression x
x FAX/MODEM Intra domain call transp + YES         x
x FAX/MODEM Extra domain call transp + YES         x
x          G722 allowed in Intra-domain + NO       x
x          G722 allowed in Extra-domain + NO       x
x          Tandem Primary Domain : -1              x
x          Domain Max Voice Connection : -1        x
x          IP Quality of service : 0               x
x          Contact Number : -----                x
x          Backup IP address : -----             x
x          Trunk Group ID : -1                     x
x IP recording quality of service : 0              x
x          Time Zone Name + System Default         x
x          Calling Identifier : -----            x
x          Supplement. Calling Identifier : ----- x
x          SIP Survivability Mode + NO             x
x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj

```

### 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 examples

```
Node Number (reserved) : 101
Directory Number : 4004

Directory name : NOE_USER1
Directory FIRST Name : -----
UTF-8 Directory Name : -----
UTF-8 Directory First Name : -----
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 : ****
Confirm : ****

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
```

## Configure SIP user

**Navigation:** Users → Create

Node Number (reserved): 101

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

```
Node Number (reserved) : 101
Directory Number : 4003

Directory name : ALU_SIP1
Directory First Name : -----
UTF-8 Directory Name : -----
UTF-8 Directory First Name : -----
Location Node : 1
Shelf Address : 255
Board Address : 255
Equipment Address : 255
Set Type + SIP extension
Entity Number : 1
Set Function + Default
Profile Name : -----
Key Profiles + None
Domain Identifier : 0
Language ID : 1

Secret Code : ****
Confirm : ****

Associated Set No. : 4003
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
```

URL Username: 4003

URL Domain: tekvisionoxe

SIP Authentication: 4003

SIP Password: \*\*\*\*\*

Confirm: \*\*\*\*\*

Explicit Priority

```
Activation mode : 0
Priority Level : 0

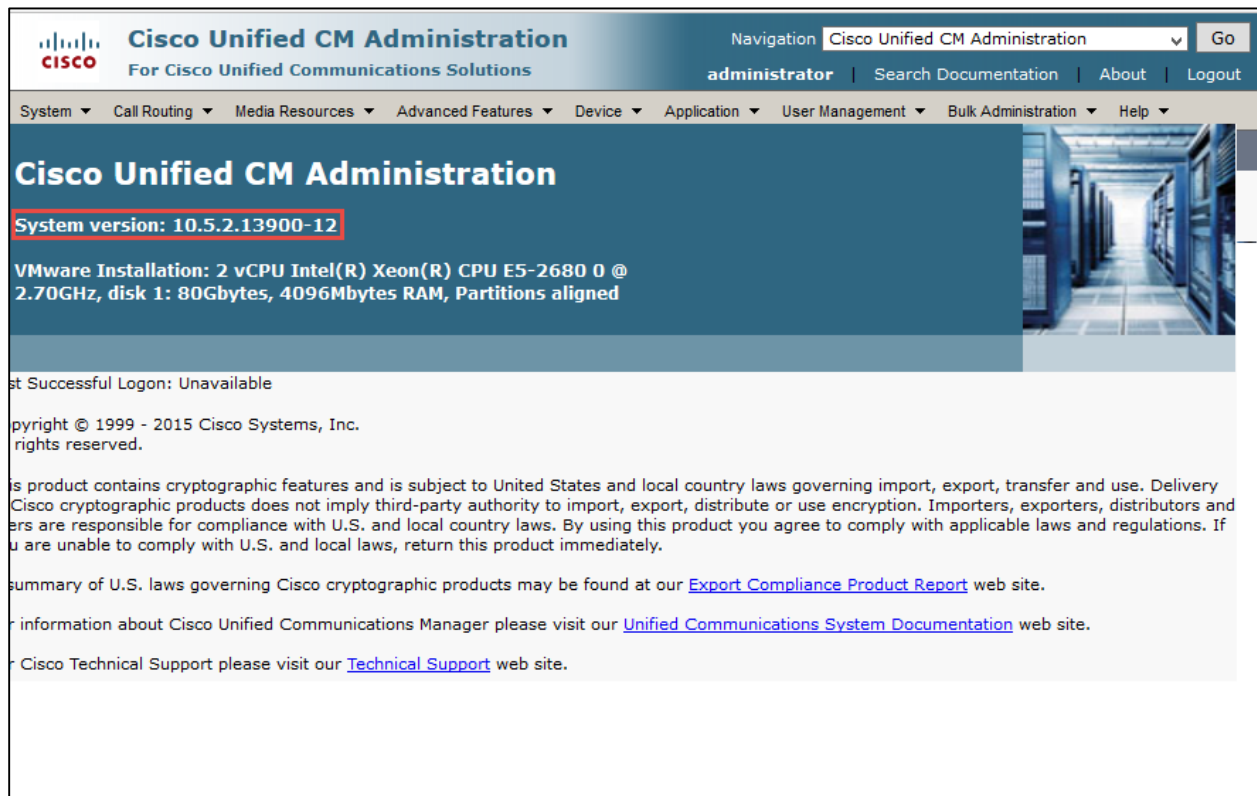
Pre-emptable Primary Inc. Line + NO
Pre-emptable Secondary Inc. Line + NO
Priority Presentation + NO
Ith Service type + Not Valid
CUG List Number : -1
Preferential CUG : -1
CUG Outgoing Access + False
CUG Incoming Access + False
Automatic reconfiguration + CTQ Forbidden - Connection TO
Associated RSI : -----
URL UserName : 4003
URL Domain : tekvizionoxe
SIP Authentication : 4003

SIP Passwd : *****
Confirmation : *****

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



The screenshot displays the Cisco Unified CM Administration web interface. At the top, the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions" are visible. A navigation bar includes a search field with "Cisco Unified CM Administration" and a "Go" button, along with links for "administrator", "Search Documentation", "About", and "Logout". A secondary navigation bar lists various system components: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area features a large blue header with the title "Cisco Unified CM Administration" and a sub-header "System version: 10.5.2.13900-12" which is highlighted with a red box. Below this, the hardware configuration is listed: "VMware Installation: 2 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 4096Mbytes RAM, Partitions aligned". A message states "Last Successful Logon: Unavailable". The footer contains copyright information for 1999-2015 Cisco Systems, Inc., and several links for legal compliance, documentation, and technical support.

**Cisco Unified CM Administration**

**System version: 10.5.2.13900-12**

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

Last 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.

For information about Cisco Unified Communications Manager please visit our [Unified Communications System Documentation](#) web site.

For Cisco Technical Support please visit our [Technical Support](#) web site.

## 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.

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The page title is "SIP Trunk Security Profile Configuration". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

The configuration form is titled "SIP Trunk Security Profile Information" and contains the following fields and options:

- Name\*: ALU Non Secure SIP Trunk Profile
- Description: Non Secure SIP Trunk Profile authenticated by null String
- Device Security Mode: Non Secure
- Incoming Transport Type\*: TCP+UDP
- Outgoing Transport Type: UDP
- Enable Digest Authentication
- Nonce Validity Time (mins)\*: 600
- X.509 Subject Name: (empty)
- Incoming Port\*: 5060
- Enable Application level authorization
- Accept presence subscription
- Accept out-of-dialog refer\*\*
- Accept unsolicited notification
- Accept replaces header
- Transmit security status
- Allow charging header
- SIP V.150 Outbound SDP Offer Filtering\*: Use Default Filter

At the bottom of the form, there are buttons for Save, Delete, Copy, Reset, Apply Config, and Add New.

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.

The screenshot shows the Cisco Unified CM Administration interface for configuring a SIP Trunk Security Profile. The page title is "SIP Trunk Security Profile Configuration". The navigation bar includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The user is logged in as "administrator".

The configuration form is titled "SIP Trunk Security Profile Information" and contains the following fields and options:

- Name\*: SIP Trunk Profile for Unity Connection
- Description: Non Secure SIP Trunk Profile authenticated by null String
- Device Security Mode: Non Secure
- Incoming Transport Type\*: TCP+UDP
- Outgoing Transport Type: TCP
- Enable Digest Authentication
- Nonce Validity Time (mins)\*: 600
- X.509 Subject Name: (empty)
- Incoming Port\*: 5060
- Enable Application level authorization
- Accept presence subscription
- Accept out-of-dialog refer\*\*
- Accept unsolicited notification
- Accept replaces header
- Transmit security status
- Allow charging header
- SIP V.150 Outbound SDP Offer Filtering\*: Use Default Filter

Buttons at the bottom include Save, Delete, Copy, Reset, Apply Config, and Add New.

## 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**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

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

### SIP Profile Configuration

Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Status**

- 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-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, anc
Confidential Access Level Headers*	Disabled

Redirect by Application  
 Disable Early Media on 180

Outgoing T.38 INVITE include audio mline  
 Use Fully Qualified Domain Name in SIP Requests  
 Assured Services SIP conformance

## Cisco Unified Communications Manager SIP Profile (Continued)

These values are default.

The screenshot shows the Cisco Unified CM Administration interface for SIP Profile Configuration. The page includes a navigation bar with the Cisco logo and 'Cisco Unified CM Administration For Cisco Unified Communications Solutions'. The user is logged in as 'administrator'. The main content area is titled 'SIP Profile Configuration' and includes a toolbar with 'Save', 'Delete', 'Copy', 'Reset', 'Apply Config', and 'Add New' buttons. Below the toolbar, there are two main sections: 'SDP Information' and 'Parameters used in Phone'.

**SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default

Require SDP Inactive Exchange for Mid-Call Media Change  
 Allow RR/RS bandwidth modifier (RFC 3556)

**Parameters used in Phone**

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup

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

All other values are default.

## Cisco Unified Communications Manager SIP Profile (Continued)

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

Navigation: Cisco Unified CM Administration

administrator | Search Documentation | About | Logout

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


### SIP Profile Configuration

Related Links: Back To Find/List

Call Pickup Group URI*	<input type="text" value="x-cisco-serviceuri-gpickup"/>
Meet Me Service URI*	<input type="text" value="x-cisco-serviceuri-meetme"/>
User Info*	<input type="text" value="None"/>
DTMF DB Level*	<input type="text" value="Nominal"/>
Call Hold Ring Back*	<input type="text" value="Off"/>
Anonymous Call Block*	<input type="text" value="Off"/>
Caller ID Blocking*	<input type="text" value="Off"/>
Do Not Disturb Control*	<input type="text" value="User"/>
Telnet Level for 7940 and 7960*	<input type="text" value="Disabled"/>
Resource Priority Namespace	<input type="text" value="&lt; None &gt;"/>
Timer Keep Alive Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Delta (seconds)*	<input type="text" value="5"/>
Maximum Redirections*	<input type="text" value="70"/>
Off Hook To First Digit Timer (milliseconds)*	<input type="text" value="15000"/>
Call Forward URI*	<input type="text" value="x-cisco-serviceuri-cfwdall"/>
Speed Dial (Abbreviated Dial) URI*	<input type="text" value="x-cisco-serviceuri-abbrdial"/>

Conference Join Enabled  
 RFC 2543 Hold  
 Semi Attended Transfer

## Cisco Unified Communications Manager SIP Profile (Continued)

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

Navigation: Cisco Unified CM Administration

administrator | Search Documentation | About | Logout

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

**SIP Profile Configuration** Related Links: Back To Find/List

Enable VAD  
 Stutter Message Waiting  
 MLPP User Authorization

**Normalization Script**

Normalization Script:

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>

**Incoming Requests FROM URI Settings**

Caller ID DN:   
Caller Name:

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on\*:

RSVP Over SIP\*:

Resource Priority Namespace List:

Fall back to local RSVP

SIP Rel1XX Options\*:

Video Call Traffic Class\*:

## 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.

Allow multiple codecs in answer SDP

Save Delete Copy Reset Apply Config Add New

**i** \*- indicates required item.

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

Navigation Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

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

**SIP Profile Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Mandatory (insert MTP if needed)

Enable ANAT  
 Deliver Conference Bridge Identifier  
 Allow Passthrough of Configured Line Device Caller Information  
 Reject Anonymous Incoming Calls  
 Reject Anonymous Outgoing Calls  
 Send ILS Learned Destination Route String

**SIP OPTIONS Ping**

<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

**SDP Information**

Send send-receive SDP in mid-call INVITE  
 Allow Presentation Sharing using BFCP  
 Allow iX Application Media

## 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.

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The configuration page includes a "SIP Trunk Status" section showing "Service Status: Full Service" and "Duration: Time In Full Service: 0 day 2 hours 20 minutes". The "Device Information" section contains the following fields:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	ALU
Description	SIP Trunk to ALU
Device Pool*	G711 Preferred
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_NoMTP
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

Red boxes highlight the fields: Device Name\*, Device Pool\*, Call Classification\*, and Media Resource Group List.

## Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". A navigation menu contains links for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Trunk Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this, there are action buttons: Save, Delete, Reset, and Add New. The configuration options are as follows:

- Media Termination Point Required
- Retry Video Call as Audio
- Path Replacement Support
- Transmit UTF-8 for Calling Party Name
- Transmit UTF-8 Names in QSIG APDU
- Unattended Port
- SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.
- Consider Traffic on This Trunk Secure\*:
- Route Class Signaling Enabled\*:
- Use Trusted Relay Point\*:
- PSTN Access
- Run On All Active Unified CM Nodes

Below these options are three sections:

- Intercompany Media Engine (IME)**: E.164 Transformation Profile
- MLPP and Confidential Access Level Information**:
  - MLPP Domain:
  - Confidential Access Mode:
  - Confidential Access Level:

All other values are default.

## Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

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

Navigation Cisco Unified CM Administration

administrator | Search Documentation | About | Logout

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

**Trunk Configuration** Related Links: Back To Find/List

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
Asserted-Type\*   
SIP Privacy\*

**Inbound Calls**

Significant Digits\*   
Connected Line ID Presentation\*   
Connected Name Presentation\*   
Calling Search Space   
AAR Calling Search Space   
Prefix DN   
 Redirecting Diversion Header Delivery - Inbound



## Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

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

Navigation Cisco Unified CM Administration Go

administrator | [Search Documentation](#) | [About](#) | [Logout](#)

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

Related Links: Back To Find/List Go

Save Delete Reset Add New

**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.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<span style="border: 1px solid black; padding: 2px;">&lt; None &gt;</span>	<input checked="" type="checkbox"/>

**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.

Clear Prefix Settings
Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<span style="border: 1px solid black; padding: 2px;">&lt; None &gt;</span>	<input checked="" type="checkbox"/>

**Connected Party Settings**

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS

## Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

Check Redirecting Diversion Header Delivery - Outbound

All other values are default.

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

Navigation Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

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

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**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

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

Navigation Cisco Unified CM Administration Go

administrator Search Documentation About Logout

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

**Trunk Configuration** Related Links: Back To Find/List Go

Save Delete Reset Add New

**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	10.70.50.6		5060

MTP Preferred Originating Codec\* 711ulaw

BLF Presence Group\* Standard Presence group

SIP Trunk Security Profile\* ALU Non Secure SIP Trunk Profile

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* ALU - Standard SIP Profile [View Details](#)

DTMF Signaling Method\* No Preference

**Normalization Script**

Normalization Script < None >

Enable Trace

### Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)

Parameter Name	Parameter Value
1	

**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

Save Delete Reset Add New

**i** \*- indicates required item.

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

## 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.

The screenshot displays the 'Trunk Configuration' page in Cisco Unified CM Administration. The 'Device Information' section is expanded, showing the following configuration details:

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	UnityConnection
Description	UnityConnection
Device Pool*	Default
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	< None >
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes
ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0

Additional options at the bottom of the form include:

- Media Termination Point Required
- Retry Video Call as Audio
- Path Replacement Support
- Transmit UTF-8 for Calling Party Name

## 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.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation menu with "Cisco Unified CM Administration" selected. Below the navigation bar is a breadcrumb trail: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The main content area is titled "Trunk Configuration" and includes a "Related Links" section with "Back To Find/List". Below this are action buttons: Save, Delete, Reset, and Add New. The configuration area contains several checkboxes and dropdown menus: "Transmit UTF-8 Names in QSIG APDU" (unchecked), "Unattended Port" (unchecked), "SRTP Allowed" (checked), "Consider Traffic on This Trunk Secure\*" (dropdown: "When using both sRTP and TLS"), "Route Class Signaling Enabled\*" (dropdown: "Default"), "Use Trusted Relay Point\*" (dropdown: "Default"), "PSTN Access" (checked), and "Run On All Active Unified CM Nodes" (checked and highlighted with a red box). Below this is the "Intercompany Media Engine (IME)" section with an "E.164 Transformation Profile" dropdown set to "< None >". The "MLPP and Confidential Access Level Information" section includes "MLPP Domain", "Confidential Access Mode", and "Confidential Access Level" dropdowns, all set to "< None >".

## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

Check Redirecting Diversion Header Delivery - Inbound

All other values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes navigation menus for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

The "Inbound Calls" section is expanded, showing the following configuration options:

- Significant Digits\*: All
- Connected Line ID Presentation\*: Default
- Connected Name Presentation\*: Default
- Calling Search Space: < None >
- AAR Calling Search Space: < None >
- Prefix DN: (empty field)
- Redirecting Diversion Header Delivery - Inbound (highlighted with a red box)

The "Incoming Calling Party Settings" section includes a descriptive paragraph and two buttons: "Clear Prefix Settings" and "Default Prefix Settings". Below this is a table with the following data:

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

The "Incoming Called Party Settings" section also includes a descriptive paragraph and two buttons: "Clear Prefix Settings" and "Default Prefix Settings".

Check Redirecting Diversion Header Delivery – Outbound

## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

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

Navigation: Cisco Unified CM Administration

administrator | [Search Documentation](#) | [About](#) | [Logout](#)

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

### Trunk Configuration

Related Links: [Back To Find/List](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**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

## 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.

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and it includes a navigation menu at the top with options like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The user is logged in as "administrator".

The configuration is divided into two main sections: "Caller Information" and "SIP Information".

**Caller Information:**

- Caller ID DN: [Text Input Field]
- Caller Name: [Text Input Field]
- Maintain Original Caller ID DN and Caller Name in Identity Headers

**SIP Information:**

**Destination:**

- Destination Address is an SRV
- 1 \* [Destination Address: 10.80.10.5] [Destination Address IPv6: ] [Destination Port: 5060]

**Other SIP Parameters:**

- MTP Preferred Originating Codec\*: 711ulaw
- BLF Presence Group\*: Standard Presence group
- SIP Trunk Security Profile\*: UnityConnectionTrunkSecurityProfile
- Rerouting Calling Search Space: < None >
- Out-Of-Dialog Refer Calling Search Space: < None >
- SUBSCRIBE Calling Search Space: < None >
- SIP Profile\*: Standard SIP Profile
- DTMF Signaling Method\*: No Preference

A "View Details" link is visible next to the SIP Profile dropdown.



## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)

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

Navigation: Cisco Unified CM Administration

administrator | Search Documentation | About | Logout

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

**Trunk Configuration** Related Links:

**Normalization Script**

Normalization Script:

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>

**Recording Information**

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

**Geolocation Configuration**

Geolocation:   
Geolocation Filter:   
 Send Geolocation Information

**i** \*- indicates required item.  
**i** \*\*- 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.

The screenshot shows the Cisco Unified CM Administration web interface. The page title is "Service Parameter Configuration" and it is for the "Cisco Call Manager (Active)" service. The "Duplex Streaming Enabled" parameter is highlighted with a red box, showing its current value as "True".

Parameter Name	Value	Default Value
Stop Hunting on Out of Bandwidth Flag *	False	False
Use Pickup Group Of Line Group Member DN *	True	False
External QoS Enabled *	False	False
Default Network Hold MOH Audio Source ID *	1	1
Default User Hold MOH Audio Source ID *	1	1
<b>Duplex Streaming Enabled *</b>	<b>True</b>	<b>False</b>
Media Exchange Interface Capability Timer *	8	8
Send Multicast MOH in H.245 OLC Message *	True	True
Media Exchange Timer *	12	12
Media Exchange Stop Streaming Timer *	8	8
Open Video Channel Response Timer for SIP Interop *	500	500
Port Received Timer After Call Connection *	500	500
Media Resource Allocation Timer *	12	12
MTP and Transcoder Resource Throttling Percentage	95	95

**\*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.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". A secondary navigation menu lists various system components. The main content area is titled "Find and List Device Pools" and includes a search bar and a table of device pools. The table has columns for Name, Cisco Unified CM Group, Region, Date/Time Group, and Copy. Three device pools are listed: "Default", "G711 Preferred", and "G729 Preferred". The "G711 Preferred" and "G729 Preferred" rows are highlighted with a red border. Below the table are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

	Name ^	Cisco Unified CM Group	Region	Date/Time Group	Copy
<input type="checkbox"/>	<a href="#">Default</a>	<a href="#">Default</a>	<a href="#">Default</a>	<a href="#">CMLocal</a>	
<input type="checkbox"/>	<a href="#">G711 Preferred</a>	<a href="#">Default</a>	<a href="#">G711 Preferred</a>	<a href="#">CMLocal</a>	
<input type="checkbox"/>	<a href="#">G729 Preferred</a>	<a href="#">Default</a>	<a href="#">G729 Preferred</a>	<a href="#">CMLocal</a>	

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

Navigation Cisco Unified CM Administration Go  
administrator Search Documentation About Logout

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

**Device Pool Configuration** Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Device Pool Settings**

Device Pool Name\* G711 Preferred

Cisco Unified Communications Manager Group\* Default

Calling Search Space for Auto-registration < None >

Adjunct CSS < None >

Reverted Call Focus Priority Default

Intercompany Media Services Enrolled Group < None >

**Roaming Sensitive Settings**

Date/Time Group\* CMLocal

Region\* G711 Preferred

Media Resource Group List MRGL\_NoMTP

Location < None >

Network Locale < None >

SRST Reference\* Disable

Connection Monitor Duration\*\*\*

Single Button Barge\* Default

Join Across Lines\* Default

Physical Location < None >

Device Mobility Group < None >

Wireless LAN Profile Group < None > [View Details](#)

### Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.

Cisco Unified CM Administration  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator Search Documentation About Logout

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

Device Pool Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**Local Route Group Settings**  
Standard Local Route Group < None >

**Device Mobility Related Information\*\*\*\***  
 Device Mobility Calling Search Space < None >  
 AAR Calling Search Space < None >  
 AAR Group < None >  
 Calling Party Transformation CSS < None >  
 Called Party Transformation CSS < None >

**Geolocation Configuration**  
 Geolocation < None >  
 Geolocation Filter < None >

**Call Routing Information**

**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.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

### Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.

Cisco Unified CM Administration  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go  
administrator Search Documentation About Logout

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

Device Pool Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

**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.

Clear Prefix Settings Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

**Phone Settings**

Caller ID For Calls From This Phone  
Calling Party Transformation CSS < None >

**Connected Party Settings**

Connected Party Transformation CSS < None >

**Redirecting Party Settings**

Redirecting Party Transformation CSS < None >

Save Delete Copy Reset Apply Config Add New

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

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

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

**Device Pool Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

**Device Pool Settings**

Device Pool Name\* | 6729 Preferred

Cisco Unified Communications Manager Group\* | Default

Calling Search Space for Auto-registration | < None >

Adjunct CSS | < None >

Reverted Call Focus Priority | Default

Intercompany Media Services Enrolled Group | < None >

**Roaming Sensitive Settings**

Date/Time Group\* | CMLocal

Region\* | G729 Preferred

Media Resource Group List | MRGL

Location | < None >

Network Locale | < None >

SRST Reference\* | Disable

Connection Monitor Duration\*\*\* |

Single Button Barge\* | Default

Join Across Lines\* | Default

Physical Location | < None >

Device Mobility Group | < None >

Wireless LAN Profile Group | < None > | [View Details](#)

## Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.

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

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

**Device Pool Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Reset | Apply Config | Add New

Calling Party Transformation CSS | < None >

Called Party Transformation CSS | < None >

**Geolocation Configuration**

Geolocation | < None >

Geolocation Filter | < None >

**Call Routing Information**

**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.

Clear Prefix Settings | Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

**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.

Clear Prefix Settings | Default Prefix Settings

## Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.

The screenshot displays the Cisco Unified CM Administration interface for Device Pool Configuration. The page title is "Device Pool Configuration" and it includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is divided into several sections:

- Incoming Called Party Settings:** A section with a descriptive paragraph and two buttons: "Clear Prefix Settings" and "Default Prefix Settings". Below this is a table with four columns: "Number Type", "Prefix", "Strip Digits", and "Calling Search Space".
- Phone Settings:** A section with a sub-section "Caller ID For Calls From This Phone" containing a dropdown menu for "Calling Party Transformation CSS" set to "< None >".
- Connected Party Settings:** A section with a dropdown menu for "Connected Party Transformation CSS" set to "< None >".
- Redirecting Party Settings:** A section with a dropdown menu for "Redirecting Party Transformation CSS" set to "< None >".

At the bottom of the configuration area, there are buttons for "Save", "Delete", "Copy", "Reset", "Apply Config", and "Add New". Below these buttons, there are four informational icons with text:

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

## 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.



The screenshot shows the Cisco Unified CM Administration interface. At the top, there is a navigation bar with the Cisco logo and the text 'Cisco Unified CM Administration For Cisco Unified Communications Solutions'. Below this is a menu bar with options like 'System', 'Call Routing', 'Media Resources', etc. The main content area is titled 'Find and List Regions'. It includes a status bar showing '3 records found' and a table of regions. The table has a column for checkboxes and a column for region names. The regions listed are 'Default', 'G711 Preferred', and 'G729 Preferred'. There are also buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected' at the bottom of the table.

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

Navigation: Cisco Unified CM Administration [Go]  
administrator | Search Documentation | About | Logout

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

Region Configuration | Related Links: Back To Find/List [Go]

Save | Delete | Reset | Apply Config | Add New

**Region Information**  
Name\* G711 Preferred

**Region Relationships**

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G711 G729	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G711 Preferred	G711 G729	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps

NOTE: Regions not displayed | Use System Default | Use System Default | Use System Default | Use System Default

**Modify Relationship to other Regions**

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default G711 Preferred G729 Preferred	Keep Current Setting	Keep Current Setting <input type="radio"/> <input type="text" value=""/> kbps	Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps

Save | Delete | Reset | Apply Config | Add New

**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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

**Region Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Reset | Apply Config | Add New

**Region Information**  
Name\*: G729 Preferred

**Region Relationships**

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default	G729 G711	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
G729 Preferred	G729 G711	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps

NOTE: Regions not displayed | Use System Default | Use System Default | Use System Default | Use System Default

**Modify Relationship to other Regions**

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
Default G711 Preferred G729 Preferred	Keep Current Setting	Keep Current Setting <input type="radio"/> <input type="text"/> kbps	Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps	Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps

Save | Delete | Reset | Apply Config | Add New

## Cisco Unified Communications Manager Media Resource Group

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

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

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

**Find and List Media Resource Groups**

+ Add New | Select All | Clear All | Delete Selected

**Status**  
2 records found

**Media Resource Group (1 - 2 of 2)** | Rows per Page: 50

Find Media Resource Group where: Name | begins with | Find | Clear Filter

	Name ^	Description	Multi-cast	Copy
<input type="checkbox"/>	<a href="#">MRG</a>		false	
<input type="checkbox"/>	<a href="#">MRG_NoMTP</a>		false	

Add New | Select All | Clear All | Delete Selected

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group. The page title is "Media Resource Group Configuration". The navigation menu at the top includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area is divided into several sections:

- Media Resource Group Status:** Shows "Media Resource Group: MRG (used by 5 devices)".
- Media Resource Group Information:** Contains fields for "Name\*" (set to "MRG") and "Description".
- Devices for this Group:** Contains two lists:
  - Available Media Resources\*\*:** An empty list.
  - Selected Media Resources\*:** A list containing "ANN\_2 (ANN)", "ANN\_3 (ANN)", "ANN\_4 (ANN)", "CFB\_2 (CFB)", and "CFB\_3 (CFB)".
- Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available):** A checkbox that is currently unchecked.

At the bottom of the page, there are buttons for "Save", "Delete", "Copy", and "Add New".

## 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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

### Media Resource Group Configuration

Related Links: Back To Find/List | Go

Save ~~X~~ Delete Copy + Add New

**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\*  
ANN\_2 (ANN)  
ANN\_3 (ANN)  
ANN\_4 (ANN)  
CFB\_2 (CFB)  
CFB\_3 (CFB)

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

Save Delete Copy Add New

## Cisco Unified Communications Manager Media Resource Group List

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

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

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

### Find and List Media Resource Group Lists

+ Add New Select All Clear All ~~X~~ Delete Selected

**Status**  
2 records found

**Media Resource Group List (1 - 2 of 2)** Rows per Page 50

Find Media Resource Group List where Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^	Copy
<input type="checkbox"/>	<a href="#">MRGL</a>	
<input type="checkbox"/>	<a href="#">MRGL_NoMTP</a>	

+ Add New Select All Clear All Delete Selected

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

The screenshot displays the Cisco Unified CM Administration web interface for configuring a Media Resource Group List (MRGL). The page title is "Media Resource Group List Configuration". The navigation bar includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The user is logged in as "administrator".

Key sections of the configuration page include:

- Status:** Shows "Status: Ready".
- Media Resource Group List Status:** Indicates "Media Resource Group List: MRGL (used by 5 devices)".
- Media Resource Group List Information:** The "Name\*" field is set to "MRGL".
- Media Resource Groups for this List:** This section contains two lists:
  - Available Media Resource Groups:** A list box containing "MRG\_NoMTP".
  - Selected Media Resource Groups:** A list box containing "MRG".

At the bottom of the configuration area, there are buttons for "Save", "Delete", "Copy", and "Add New".

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

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation menu with "Navigation" set to "Cisco Unified CM Administration" and a "Go" button. Below this, a secondary navigation bar shows "administrator" and links for "Search Documentation", "About", and "Logout". A main menu bar contains various system categories like "System", "Call Routing", "Media Resources", etc. The current page is titled "Media Resource Group List Configuration" and includes a "Related Links" section with "Back To Find/List" and a "Go" button. Below the title bar, there are icons for "Save", "Delete", "Copy", and "Add New". The main content area is divided into several sections:
 

- Status:** Shows "Status: Ready" with an information icon.
- Media Resource Group List Status:** Displays "Media Resource Group List: MRGL\_NoMTP (used by 12 devices)".
- Media Resource Group List Information:** Contains a "Name\*" field with the value "MRGL\_NoMTP".
- Media Resource Groups for this List:** Features two list boxes. The "Available Media Resource Groups" list contains "MRG". The "Selected Media Resource Groups" list contains "MRG\_NoMTP". Arrows between the lists indicate the ability to move items between them.

 At the bottom of the page, there are buttons for "Save", "Delete", "Copy", and "Add New".

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.

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

Navigation Cisco Unified CM Administration Go  
administrator Search Documentation About Logout

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

**Route Pattern Configuration** Related Links: Back To Find/List Go

Save Delete Copy Add New

**Status**  
Status: Ready

**Pattern Definition**

Route Pattern\* 4XXX

Route Partition < None >

Description ALU

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence\* Default

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain < None >

Route Class\* Default

Gateway/Route List\* ALU (Edit)

Route Option  
 Route this pattern  
 Block this pattern No Error

Call Classification\* OffNet

External Call Control Profile < None >

Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority

Require Forced Authorization Code

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

Navigation: Cisco Unified CM Administration [Go]  
administrator | Search Documentation | About | Logout

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

**Route Pattern Configuration** Related Links: Back To Find/List [Go]

Save [X] Delete Copy Add New

**Calling Party Transformations**

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask: XXXX

Prefix Digits (Outgoing Calls):

Calling Line ID Presentation\*: Default

Calling Name Presentation\*: Default

Calling Party Number Type\*: Cisco CallManager

Calling Party Numbering Plan\*: Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation\*: Default

Connected Name Presentation\*: Default

**Called Party Transformations**

Discard Digits: < None >

Called Party Transform Mask:

Prefix Digits (Outgoing Calls):

Called Party Number Type\*: Cisco CallManager

Called Party Numbering Plan\*: Cisco CallManager

## 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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

**Phone Configuration** | Related Links: Back To Find/List

Save | Delete | Copy | Reset | Apply Config | Add New

1	Line [1] - 5004 (no partition)	Product Type: Cisco 8961
2	Line [2] - Add a new DN	Device Protocol: SIP
3	Add a new SD	<b>Real-time Device Status</b>
4	Add a new SD	Registration: Registered with Cisco Unified Communications Manager clus20sub1
5	Add a new SD	IPv4 Address: 172.16.26.49
6	Unassigned Associated Items	Active Load ID: sip8961.9-4-2SR2-2
7	All Calls	Inactive Load ID: sip8961.9-4-2-13
8	Answer Oldest	Download Status: None
9	Add a new BLF Directed Call Park	<b>Device Information</b>
10	Call Pickup	<input checked="" type="checkbox"/> Device is Active
11	CallBack	<input checked="" type="checkbox"/> Device is trusted
12	Do Not Disturb	MAC Address*: 081FF3625DD1
13	Group Call Pickup	Description: SEP081FF3625DD1
14	Hunt Group Logout	Device Pool*: G711 Preferred
15	Intercom [1] - Add a new Intercom	Common Device Configuration: < None >
16	Malicious Call Identification	Phone Button Template*: Standard 8961 SIP
17	Meet Me Conference	Softkey Template: Standard User
		Common Phone Profile*: Standard Common Phone Profile
		Calling Search Space: < None >

### 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.

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

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

**Phone Configuration** | Related Links: Back To Find/List

Save | Delete | Copy | Reset | Apply Config | Add New

19	Other Pickup	Media Resource Group List: < None >
20	Quality Reporting Tool	User Hold MOH Audio Source: < None >
21	Redial	Network Hold MOH Audio Source: < None >
22	Add a new SURL	Location*: Hub_None
23	Services	AAR Group: < None >
24	Add a new BLF SD	User Locale: < None >
25	Call Park	Network Locale: < None >
26	Record	Built In Bridge*: Default
27	Alerting Calls	Privacy*: Default
28	Queue Status	Device Mobility Mode*: Default
29	Privacy	Owner: <input type="radio"/> User <input checked="" type="radio"/> Anonymous (Public/Shared Space)
30	None	Owner User ID: < >
		Phone Personalization*: Default
		Services Provisioning*: Default
		Phone Load Name: < >
		Use Trusted Relay Point*: Default
		BLF Audible Alert Setting (Phone Idle)*: Default
		BLF Audible Alert Setting (Phone Busy)*: Default

javascript:popUpWindow('/ccmadmin/serviceUrlEdit.do?fkDevice=4e6200b7-55be-73fa-7c42-09584dc7ef66&tkClass=1&tkProduct=427')

## Cisco Unified Communications Manager SIP Phone Ext. 5004 Device Level Configuration (Continued)

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation menu with "Cisco Unified CM Administration" selected. Below the navigation bar is a menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Phone Configuration" and includes a "Related Links" section with a dropdown menu set to "Back To Find/List" and a "Go" button. Below this is a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New. The configuration area is divided into two sections. The first section contains several dropdown menus: "BLF Audible Alert Setting (Phone Busy)\*" set to "Default", "Always Use Prime Line\*" set to "Default", "Always Use Prime Line for Voice Message\*" set to "Default", "Geolocation" set to "< None >", and "Feature Control Policy" set to "< None >". Below these are several checkboxes: "Ignore Presentation Indicators (internal calls only)" (unchecked), "Allow Control of Device from CTI" (checked), "Logged Into Hunt Group" (checked), "Remote Device" (unchecked), "Protected Device\*\*\*\*" (unchecked), and "Require off-premise location" (unchecked). The second section is titled "Number Presentation Transformation" and contains a sub-section "Caller ID For Calls From This Phone" with a dropdown menu for "Calling Party Transformation CSS" set to "< None >" and a checked checkbox for "Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)".

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

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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration [Go]  
administrator | Search Documentation | About | Logout

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

**Phone Configuration** Related Links: Back To Find/List [Go]

Save Delete Copy Reset Apply Config Add New

**Remote Number**

Calling Party Transformation CSS: < None >  
 Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

**Protocol Specific Information**

Packet Capture Mode\*: None  
 Packet Capture Duration: 0  
 BLF Presence Group\*: Standard Presence group  
 SIP Dial Rules: < None >  
 MTP Preferred Originating Codec\*: 711ulaw  
 Device Security Profile\*: Cisco 8961 - Standard SIP Non-Secure Profile  
 Rerouting Calling Search Space: < None >  
 SUBSCRIBE Calling Search Space: < None >  
 SIP Profile\*: Standard SIP Profile [View Details](#)  
 Digest User: < None >

Media Termination Point Required  
 Unattended Port

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

All other values are default.

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

Navigation: Cisco Unified CM Administration [Go]  
administrator | Search Documentation | About | Logout

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

**Phone Configuration** Related Links: Back To Find/List [Go]

Save Delete Copy Reset Apply Config Add New

Require DTMF Reception

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

Certificate Operation\*: No Pending Operation  
 Authentication Mode\*: By Null String  
 Authentication String:   
  
 Key Size (Bits)\*: 2048  
 Operation Completes By: 2015 5 15 12 (YYYY:MM:DD:HH)  
 Certificate Operation Status: None  
 Note: Security Profile Contains Addition CAPF Settings.

**Expansion Module Information**

Module 1: < None >  
 Module 1 Load Name:   
 Module 2: < None >  
 Module 2 Load Name:

**External Data Locations Information (Leave blank to use default)**

Information:   
 Directory:

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

All values are default.

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

Navigation Cisco Unified CM Administration Go

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

**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

**Phone Configuration**

**Related Links:** Back To Find/List Go

Save ✖ Delete Copy Reset Apply Config Add New

Secure Information URL

↑

Secure Messages URL

Secure Services URL

**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**

	Param	Override Common Settings
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
PC Port *	Enabled	
Back USB Port *	Enabled	<input type="checkbox"/>

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

All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation dropdown menu set to "Cisco Unified CM Administration" with a "Go" button. Below this, a secondary navigation bar shows the user role "administrator" and links for "Search Documentation", "About", and "Logout". A main menu bar contains various system categories like "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The current page is titled "Phone Configuration" and features a "Related Links" dropdown menu set to "Back To Find/List" with a "Go" button. Below the navigation is a toolbar with icons for "Save", "Delete", "Copy", "Reset", "Apply Config", and "Add New".

The main configuration area is divided into several sections:

- Require DTMF Reception:** A checkbox that is currently unchecked.
- Certification Authority Proxy Function (CAPF) Information:**
  - Certificate Operation\*: No Pending Operation (dropdown)
  - Authentication Mode\*: By Null String (dropdown)
  - Authentication String: (text input field)
  - Generate String: (button)
  - Key Size (Bits)\*: 2048 (dropdown)
  - Operation Completes By: 2015 5 15 12 (YYYY:MM:DD:HH)
  - Certificate Operation Status: None
  - Note: Security Profile Contains Addition CAPF Settings.
- Expansion Module Information:**
  - Module 1: < None > (dropdown)
  - Module 1 Load Name: (text input field)
  - Module 2: < None > (dropdown)
  - Module 2 Load Name: (text input field)
- External Data Locations Information (Leave blank to use default):**
  - Information: (text input field)
  - Directory: (text input field)

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

All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface for SIP Phone Device Level Configuration. The page title is "Phone Configuration" and it includes a navigation menu with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

At the top, there are navigation links: "Navigation Cisco Unified CM Administration" and "Go". Below that, the user role "administrator" is shown along with "Search Documentation", "About", and "Logout".

The main configuration area is titled "Phone Configuration" and includes a "Related Links" dropdown set to "Back To Find/List" with a "Go" button. Below this is a toolbar with icons for Save, Delete, Copy, Reset, Apply Config, and Add New.

The configuration fields are organized into sections:

- MLPP Preemption\***: Default (dropdown)
- Confidential Access Mode**: < None > (dropdown)
- Confidential Access Level**: < None > (dropdown)
- Do Not Disturb**:
  - Do Not Disturb
  - DND Option\***: Use Common Phone Profile Setting (dropdown)
  - DND Incoming Call Alert**: < None > (dropdown)
- Secure Shell Information**:
  - Secure Shell User**: (text input)
  - Secure Shell Password**: (password input)
- Product Specific Configuration Layout**:
  - Disable Speakerphone
  - Disable Speakerphone and Headset
  - PC Port \***: Enabled (dropdown)
  - Back USB Port\***: Enabled (dropdown)
  - Side USB Port\***: Enabled (dropdown)

On the right side of the "Product Specific Configuration Layout" section, there is a column for "Override Common Settings" with checkboxes for each parameter.

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

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 CM Administration**  
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

administrator Search Documentation About Logout

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

Phone Configuration Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

Cisco Camera*	Enabled	<input checked="" type="checkbox"/>
Console Access*	Disabled	<input type="checkbox"/>
Video Capabilities*	Enabled	<input checked="" type="checkbox"/>
Enable/Disable USB Classes	<ul style="list-style-type: none"> <li>Mass Storage</li> <li>Human Interface Device</li> <li>Audio Class</li> </ul>	<input type="checkbox"/>
Bluetooth *	Enabled	<input type="checkbox"/>
Bluetooth Profiles*	<ul style="list-style-type: none"> <li>Handsfree</li> <li>Human Interface Device</li> </ul>	<input type="checkbox"/>
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	<input type="checkbox"/>
PC Voice VLAN Access*	Enabled	<input type="checkbox"/>
Web Access*	Disabled	<input type="checkbox"/>
Show All Calls on Primary Line*	Disabled	<input type="checkbox"/>
Days Display Not Active	<ul style="list-style-type: none"> <li>Sunday</li> <li>Monday</li> <li>Tuesday</li> </ul>	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>
Display Idle Timeout	01:00	<input type="checkbox"/>

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

All values are default.



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

Navigation Cisco Unified CM Administration

administrator | Search Documentation | About | Logout

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

**Phone Configuration** Related Links:

HTTPS Server*	<input type="text" value="http and https Enabled"/>	<input type="checkbox"/>
Enable Power Save Plus	<input type="text" value="Sunday Monday Tuesday"/>	<input type="checkbox"/>
Phone On Time	<input type="text" value="00:00"/>	<input type="checkbox"/>
Phone Off Time	<input type="text" value="24:00"/>	<input type="checkbox"/>
Phone Off Idle Timeout*	<input type="text" value="60"/>	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain	<input type="text"/>	<input type="checkbox"/>
EnergyWise Endpoint Security Secret	<input type="text"/>	<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	<input type="text" value="Disabled"/>	
Logging Display*	<input type="text" value="Disabled"/>	
Load Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Load Server	<input type="text"/>	<input type="checkbox"/>
Recording Tone*	<input type="text" value="Disabled"/>	
Recording Tone Local Volume*	<input type="text" value="100"/>	
Recording Tone Remote Volume*	<input type="text" value="50"/>	

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

The screenshot displays the Cisco Unified CM Administration interface for SIP Phone Device Level Configuration. The page title is "Phone Configuration" and the user is logged in as "administrator". The "RTCP\*" setting is highlighted with a red box, showing it is set to "Enabled".

Setting	Value	Checkbox
Remote Volume*		
Recording Tone Duration		
Display On When Incoming Call*	Enabled	<input type="checkbox"/>
<b>RTCP*</b>	<b>Enabled</b>	<input checked="" type="checkbox"/>
Log Server		<input type="checkbox"/>
IPv6 Log Server		<input type="checkbox"/>
Remote Log*	Disabled	<input type="checkbox"/>
Log Profile	Default	<input type="checkbox"/>
Advertise G.722 and iSAC Codecs*	Use System Default	
Wideband Headset UI Control*	Enabled	
Wideband Headset*	Enabled	
Peer Firmware Sharing*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>

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

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration Go

**administrator** | [Search Documentation](#) | [About](#) | [Logout](#)

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

**Phone Configuration** Related Links: Back To Find/List Go

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Discovery			
Protocol - Media			
Endpoint			
Discover			
(LLDP-MED):			
Switch Port*			
Link Layer	Enabled	▾	<input type="checkbox"/>
Discovery			
Protocol (LLDP):			
PC Port*			
LLDP Asset ID			
LLDP Power	Unknown	▾	
Priority*			
802.1x	User Controlled	▾	<input type="checkbox"/>
Authentication*			
FIPS Mode*	Disabled	▾	<input type="checkbox"/>
Detect Unified	Normal	▾	<input type="checkbox"/>
CM Connection			
Failure*			
Switch Port	Disabled	▾	<input type="checkbox"/>
Remote			
Configuration*			
PC Port Remote	Disabled	▾	<input type="checkbox"/>
Configuration*			
Automatic Port	Disabled	▾	<input type="checkbox"/>
Synchronization*			
Power	Enabled	▾	<input type="checkbox"/>
Negotiation*			
SSH Access*	Disabled	▾	<input type="checkbox"/>

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

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

Navigation: Cisco Unified CM Administration


administrator | Search Documentation | About | Logout

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

**Phone Configuration** Related Links: Back To Find/List

Incoming Call Toast Timer*	<input type="text" value="5"/>	<input type="checkbox"/>
Provide Dial Tone from Release Button*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Hide Video By Default*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Background Image	<input type="text"/>	<input type="checkbox"/>
Simplified New Call UI*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Enable VXC VPN for MAC	<input type="text"/>	<input type="checkbox"/>
VXC VPN Option*	<input type="text" value="Dual Tunnel"/>	<input type="checkbox"/>
VXC Challenge*	<input type="text" value="Challenge"/>	<input type="checkbox"/>
VXC-M Servers	<input type="text"/>	<input type="checkbox"/>
Revert to All Calls*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
RTCP for Video*	<input type="text" value="Enabled"/>	<input type="checkbox"/>
Record Call Log from Shared Line*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Show Remote Private Calls*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Record Call Log For Remote Private Calls*	<input type="text" value="Enabled"/>	<input type="checkbox"/>

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

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

Navigation: Cisco Unified CM Administration

administrator | Search Documentation | About | Logout

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

Phone Configuration Related Links:

Show Call History for Selected Line Only.*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Actionable Incoming Call Alert*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
DF bit*	<input type="text" value="0"/>	<input type="checkbox"/>
Default Line Filter	<input type="text"/>	
Separate Audio and Video Mute*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
Softkey Control*	<input type="text" value="Feature Control Policy"/>	<input type="checkbox"/>
Start Video Port	<input type="text"/>	<input type="checkbox"/>
Stop Video Port	<input type="text"/>	<input type="checkbox"/>
Lowest Alerting Line State Priority*	<input type="text" value="Disabled"/>	<input type="checkbox"/>
TLS Resumption Timer*	<input type="text" value="3600"/>	<input type="checkbox"/>
Audio EQ*	<input type="text" value="Default : Default"/>	<input type="checkbox"/>

## 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.

The screenshot displays the Cisco Unified CM Administration interface for configuring a phone device. The page title is "Phone Configuration" and the navigation path is "Device → Phone". The interface includes a top navigation bar with "Cisco Unified CM Administration" and "administrator" roles. A secondary navigation bar shows "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main configuration area is divided into two panels. The left panel, titled "Modify Button Items", lists 17 items for configuration, including "Line [1] - 5001 (no partition)", "Line [2] - Add a new DN", and various "Add a new SD" and "Add a new BLF SD" options. The right panel, titled "Device Information", shows the following configuration details:

- Product Type:** Cisco 7971
- Device Protocol:** SCCP
- Real-time Device Status:**
  - Registration: Registered with Cisco Unified Communications Manager clus20sub1
  - IPv4 Address: 172.16.31.254
  - Active Load ID: SCCP70.9-4-2SR1-1S
  - Download Status: None
- Device Information:**
  - Device is Active:
  - Device is trusted:
  - MAC Address\*: 001DA21A0B85
  - Description: SEP001DA21A0B85
  - Device Pool\*: G711 Preferred (View Details)
  - Common Device Configuration: < None > (View Details)
  - Phone Button Template\*: Standard 7971 SCCP
  - Softkey Template: Standard User
  - Common Phone Profile\*: Standard Common Phone Profile (View Details)
  - Calling Search Space: < None >
  - AAR Calling Search Space: < None >
  - Media Resource Group List: < None >

Red boxes in the screenshot highlight the MAC Address\*, Description, Device Pool\*, Phone Button Template\*, and Common Phone Profile\* fields in the Device Information section.

## Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

All other values are default.

The screenshot displays the Cisco Unified CM Administration web interface for configuring an SCCP phone device. The page title is "Phone Configuration" and it includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

On the left side, there is a list of configuration options for the phone device, numbered 24 through 36:

- 24 Meet Me Conference
- 25 Mobility
- 26 New Call
- 27 Other Pickup
- 28 Quality Reporting Tool
- 29 Record
- 30 Redial
- 31 Remove Last Participant
- 32 Add a new SURL
- 33 Add a new BLF SD
- 34 Queue Status
- 35 Privacy
- 36 None

The main configuration area on the right contains the following settings:

- Built In Bridge\*: Default
- Privacy\*: Default
- Device Mobility Mode\*: Default (with a [View Current Device](#) link)
- Owner:  User  Anonymous (Public/Shared Space)
- Owner User ID: [Empty text field]
- Phone Personalization\*: Default
- Services Provisioning\*: Default
- Phone Load Name: [Empty text field]
- Use Trusted Relay Point\*: Default
- BLF Audible Alert Setting (Phone Idle)\*: Default
- BLF Audible Alert Setting (Phone Busy)\*: Default
- Always Use Prime Line\*: Default
- Always Use Prime Line for Voice Message\*: Default
- Geolocation: < None >
- Ignore Presentation Indicators (internal calls only)
- Allow Control of Device from CTI
- Logged Into Hunt Group
- Remote Device
- Protected Device\*\*\*\*
- Hot line Device\*\*\*\*\*
- Require off-premise location

## 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.

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and user information "administrator" with links for "Search Documentation", "About", and "Logout". A secondary navigation bar lists menu items: "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main content area is titled "Phone Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this is a toolbar with icons for "Save", "Delete", "Copy", "Reset", "Apply Config", and "Add New".

The configuration is organized into three sections:

- Remote Number:** Contains a dropdown for "Calling Party Transformation CSS" set to "< None >" and a checked checkbox for "Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)".
- Protocol Specific Information:** Contains several dropdown menus: "Packet Capture Mode\*" (None), "Packet Capture Duration" (0), "BLF Presence Group\*" (Standard Presence group), "Device Security Profile\*" (Cisco 7971 - Standard SCCP Non-Secure Profile), and "SUBSCRIBE Calling Search Space" (< None >). It also includes three unchecked checkboxes: "Unattended Port", "Require DTMF Reception", and "RFC2833 Disabled".
- Certification Authority Proxy Function (CAPF) Information:** Contains dropdown menus for "Certificate Operation\*" (No Pending Operation) and "Authentication Mode\*" (By Null String). It features a text input for "Authentication String" with a "Generate String" button, a dropdown for "Key Size (Bits)\*" (2048), and a date selector for "Operation Completes By" (2016 6 19 12 (YYYY:MM:DD:HH)).



## Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

These values are default.

The screenshot displays the Cisco Unified CM Administration web interface for configuring an SCCP phone device. The page title is "Phone Configuration" and it includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

Below the navigation menu, there are action buttons: Save, Delete, Copy, Reset, Apply Config, and Add New. A "Related Links" section contains a "Back To Find/List" link.

The main configuration area is divided into three sections:

- External Data Locations Information (Leave blank to use default):** This section contains a list of fields, each with a corresponding text input box:
  - 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
- Extension Information:** This section includes:
  - An unchecked checkbox for "Enable Extension Mobility".
  - A dropdown menu for "Log Out Profile" set to "-- Use Current Device Settings --".
  - Fields for "Log in Time" and "Log out Time", both set to "< None >".
- MLPP and Confidential Access Level Information:** This section includes:
  - A dropdown menu for "MLPP Domain" set to "< None >".
  - A dropdown menu for "MLPP Indication\*" set to "Default".

## Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

Set Video Capabilities\* = Enabled

These values are default.

The screenshot displays the Cisco Unified CM Administration interface for configuring an SCCP Phone Device. The main configuration area is titled "Phone Configuration" and includes a toolbar with "Save", "Delete", "Copy", "Reset", "Apply Config", and "Add New" buttons. The configuration is organized into several sections:

- Do Not Disturb:** Includes checkboxes for "Do Not Disturb", "DND Option\*" (set to "Use Common Phone Profile Setting"), and "DND Incoming Call Alert" (set to "< None >").
- Secure Shell Information:** Includes fields for "Secure Shell User" and "Secure Shell Password".
- Product Specific Configuration Layout:** A table-like structure with columns for "Parameter Value" and "Override Common Settings".

Parameter	Value	Override Common Settings
<input type="checkbox"/> Disable Speakerphone		
<input type="checkbox"/> Disable Speakerphone and Headset		
PC Port *	Enabled	
Settings Access*	Enabled	<input type="checkbox"/>
Gratuitous ARP*	Disabled	
PC Voice VLAN Access*	Enabled	
Bluetooth *	Enabled	
<b>Video Capabilities*</b>	<b>Enabled</b>	<input checked="" type="checkbox"/>
Advertise G.722 Codec*	Use System Default	
Web Access*	Disabled	<input type="checkbox"/>
Days Display Not Active	Sunday Monday Tuesday	<input type="checkbox"/>
Display On Time	07:30	<input type="checkbox"/>
Display On Duration	10:30	<input type="checkbox"/>

## Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

These values are default.

The screenshot displays the Cisco Unified CM Administration web interface for configuring an SCCP phone device. The page title is "Phone Configuration" and it includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

The configuration settings shown are:

Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>
Phone On Time	00:00	<input type="checkbox"/>
Phone Off Time	24:00	<input type="checkbox"/>
Phone Off Idle Timeout*	60	<input type="checkbox"/>
<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>
EnergyWise Domain		<input type="checkbox"/>
EnergyWise Endpoint Security Secret		<input type="checkbox"/>
<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
Span to PC Port*	Disabled	<input type="checkbox"/>
Logging Display*	PC Controlled	<input type="checkbox"/>
Load Server		<input type="checkbox"/>
Recording Tone*	Disabled	<input type="checkbox"/>
Recording Tone Local Volume*	100	<input type="checkbox"/>
Recording Tone Remote Volume*	50	<input type="checkbox"/>
Recording Tone Duration		<input type="checkbox"/>
RTCP for Audio*	Disabled	<input type="checkbox"/>
RTCP for Video*	Enabled	<input checked="" type="checkbox"/>
"more" Soft Key Timer	5	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): Switch Port*	Enabled	<input type="checkbox"/>
Cisco Discovery Protocol (CDP): PC Port*	Enabled	<input type="checkbox"/>
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled	<input type="checkbox"/>

## Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

These values are default.

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

Navigation Cisco Unified CM Administration Go

administrator
| Search Documentation | About | Logout

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

**Phone Configuration**
Related Links: Back To Find/List Go

Save Delete Copy Reset Apply Config Add New

IPv6 Load Server	<input type="text"/>	<input type="checkbox"/>
IPv6 Log Server	<input type="text"/>	<input type="checkbox"/>
802.1x Authentication*	<span style="border-bottom: 1px solid black; padding: 0 5px;">User Controlled</span>	<input type="checkbox"/>
Detect Unified CM Connection Failure*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Normal</span>	<input type="checkbox"/>
Minimum Ring Volume*	<span style="border-bottom: 1px solid black; padding: 0 5px;">0-Silent</span>	<input type="checkbox"/>
Headset Sidetone Level*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Use Phone Default</span>	<input type="checkbox"/>
HTTPS Server*	<span style="border-bottom: 1px solid black; padding: 0 5px;">http and https Enabled</span>	<input type="checkbox"/>
Enbloc Dialing*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Enabled</span>	<input type="checkbox"/>
Switch Port Remote Configuration*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>
PC Port Remote Configuration*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>
Automatic Port Synchronization*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>
SSH Access*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>
LOGIN Access*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Enabled</span>	<input type="checkbox"/>
FIPS Mode*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>
80-bit SRTCP*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>

Save Delete Copy Reset Apply Config Add New

Simplified New Call UI*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>
TLS Resumption Timer*	<input type="text" value="3600"/>	<input type="checkbox"/>
Minimum Ring Volume*	<span style="border-bottom: 1px solid black; padding: 0 5px;">0-Silent</span>	<input type="checkbox"/>
Log Server	<input type="text"/>	<input type="checkbox"/>
Remote Log*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Disabled</span>	<input type="checkbox"/>
Auto Save Volume During Call*	<span style="border-bottom: 1px solid black; padding: 0 5px;">True</span>	<input type="checkbox"/>
Peer Firmware Sharing*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Enabled</span>	<input type="checkbox"/>
Detect Unified CM Connection Failure*	<span style="border-bottom: 1px solid black; padding: 0 5px;">Normal</span>	<input type="checkbox"/>

Save Delete Copy Reset Apply Config Add New

\*- indicates required item.

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

\*\*\*Note: Security Profile Contains Addition CAPF Settings.

\*\*\*\*Note: A Protected device means it is capable of playing Secure and Non-Secure Tones. When the checkbox is checked, the user will hear a Secure or Non-Secure Tone when the call is connected.

\*\*\*\*\*Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.

## 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.

The screenshot shows the Cisco Unified CM Administration interface for Service Parameter Configuration. The page title is "Service Parameter Configuration" and it includes navigation menus for System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "administrator".

The main configuration area displays a list of parameters. The parameter "Accept Audio Codec Preferences in Received Offer" is highlighted with a red box and is set to "Off". Other parameters include "iSAC Codec Enabled", "Default Intra-region Max Audio Bit Rate", "Default Inter-region Max Audio Bit Rate", "Default Intra-region Max Video Call Bit Rate (Includes Audio)", "Default Inter-region Max Video Call Bit Rate (Includes Audio)", "Default Intra-region Max Immersive Video Call Bit Rate (Includes Audio)", "Default Inter-region Max Immersive Video Call Bit Rate (Includes Audio)", "Use Video Bandwidth Pool for Immersive Video Calls", "Default Intra-region and Inter-region Link Loss Type", "Default Audio Codec List between Regions", "Default Audio Codec List within Region", and "G.Clear Bandwidth Override".

Parameter Name	Current Value	Default Value
iSAC Codec Enabled *	Enabled for All Devices	Enabled for All Devices
Default Intra-region Max Audio Bit Rate *	64 kbps (G.722, G.711)	64 kbps (G.722, G.711)
Default Inter-region Max Audio Bit Rate *	8 kbps (G.729)	8 kbps (G.729)
Default Intra-region Max Video Call Bit Rate (Includes Audio) *	384	384
Default Inter-region Max Video Call Bit Rate (Includes Audio) *	384	384
Default Intra-region Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Default Inter-region Max Immersive Video Call Bit Rate (Includes Audio) *	2000000000	2000000000
Use Video Bandwidth Pool for Immersive Video Calls *	True	True
Default Intra-region and Inter-region Link Loss Type *	Low Loss	Low Loss
Default Audio Codec List between Regions *	Factory Default low loss	Factory Default low loss
Default Audio Codec List within Region *	Factory Default low loss	Factory Default low loss
Accept Audio Codec Preferences in Received Offer *	Off	Off
G.Clear Bandwidth Override *	False	False

Clusterwide Parameters (System - CCM Automated Alternate Routing)

Parameter Name	Current Value	Default Value
Automated Alternate Routing Enable *	False	False

## 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.

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user role "administrator". Below the navigation bar, a breadcrumb trail shows the path: System > Call Routing > Media Resources > Advanced Features > Device > Application > User Management > Bulk Administration > Help. The main heading is "Find and List Audio Codec Preference Lists". Below this heading, there are action buttons: "Add New", "Select All", "Clear All", and "Delete Selected". A status box indicates "5 records found". The main content area is titled "Audio Codec Preference Lists (1 - 5 of 5)" and includes a search filter: "Find Audio Codec Preference Lists where Name begins with". Below the search filter is a table with the following data:

<input type="checkbox"/>	Name ^	Description	Copy
<input type="checkbox"/>	<a href="#">Factory Default lossy</a>	Lossy Codec List	
<input type="checkbox"/>	<a href="#">Factory Default low loss</a>	Low Loss Codec List	
<input type="checkbox"/>	<a href="#">G711 G729</a>	G711 prefer	
<input type="checkbox"/>	<a href="#">G729 G711</a>	G729 prefer	
<input type="checkbox"/>	<a href="#">Uniqy</a>	Low Loss Codec List	

At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

## 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.

The screenshot displays the Cisco Unified CM Administration web interface. The page title is "Audio Codec Preference List Configuration". The navigation bar includes "Navigation Cisco Unified CM Administration" and "Go". The user is logged in as "administrator". The main content area shows the configuration for an audio codec preference list. The "Status" is "Ready". The "Audio Codec Preference List Information" section is highlighted with a red box and contains the following fields:

- Name\*: G711 G729
- Description\*: G711 prefer
- Codecs in List\*: A list of audio codecs, with G.711 U-Law 64k and G.729 8k selected.

The list of codecs includes: G.711 U-Law 64k, G.711 A-Law 64k, G.729 8k, G.729a 8k, G.729b 8k, G.729ab 8k, AMR-WB (7K-24K), AMR (5k-13k), MP4A-LATM 128k, AAC-LD (MP4A Generic), MP4A-LATM 64k, MP4A-LATM 56k, L16 256k, MP4A-LATM 48k, G.722 64k, ISAC 32k, MP4A-LATM 32k, G.722.1 32k, G.722 56k, G.722.1 24k, G.722 48k, MP4A-LATM 24k, and G.711 U-Law 56k.

## Cisco Unified Communications Manager Audio Codec Preference List Configuration (Continued)

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

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

Set Codec in List\*= G.729 8k. First choice for this example.

Set Codec in List\*= G.711 U-Law 64. Second choice for this example.

All other values are default.

The screenshot shows the Cisco Unified CM Administration interface for configuring an Audio Codec Preference List. The page title is "Audio Codec Preference List Configuration". The status is "Ready". The configuration fields are: Name\* (G729 G711), Description\* (G729 G711), and Codecs in List\* (a list of codecs including G.729 8k, G.711 U-Law 64k, G.729a 8k, G.729b 8k, G.711 A-Law 56k, AMR-WB (7k-24k), AMR (5k-13k), MP4A-LATM 128k, AAC-LD (MP4A Generic), MP4A-LATM 64k, MP4A-LATM 56k, L16 256k, MP4A-LATM 48k, G.722 64k, ISAC 32k, MP4A-LATM 32k, G.722.1 32k, G.722 56k, G.722.1 24k, G.722 48k, MP4A-LATM 24k, G.711 A-Law 64k, G.711 U-Law 56k, ILBC 16k, G.728 16k, GSM Enhanced Full Rate 13k, GSM Full Rate 13k, G.729ab 8k, GSM Half Rate 6k, and G.723.1 7k). The 'Name\*', 'Description\*', and the first two items in the 'Codecs in List\*' are highlighted with a red box.

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

Navigation: Cisco Unified CM Administration | Go  
 administrator | Search Documentation | About | Logout

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

**End User Configuration** | Related Links: Back to Find List Users | Go

Save | Delete | Add New

**User Information**

User Status: Enabled Local User

User ID\*: user1

Password: [Redacted] | Edit Credential

Confirm Password: [Redacted]

Self-Service User ID: 5007

PIN: [Redacted] | Edit Credential

Confirm PIN: [Redacted]

Last name\*: user1

Middle name: [Redacted]

First name: [Redacted]

Title: [Redacted]

Directory URI: user1@lab.tekvizion.com

Telephone Number: [Redacted]

Home Number: [Redacted]

Mobile Number: [Redacted]

Pager Number: [Redacted]

Mail ID: [Redacted]

Manager User ID: [Redacted]

**Cisco UCM end user Configuration (Continued)**

Pager Number: [Redacted]

Mail ID: [Redacted]

Manager User ID: [Redacted]

Department: [Redacted]

User Locale: English, United States

Associated PC: [Redacted]

Digest Credentials: [Redacted]

Confirm Digest Credentials: [Redacted]

User Profile: Use System Default( "Standard (Factory Default) U" | View Details

**Service Settings**

Home Cluster

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

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

[Presence Viewer for User](#)

UC Service Profile: Use System Default( "Jabber\_Services" ) | View Details

Set Controlled Devices = CTI1. This is used for this example.

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The main menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The current page is "End User Configuration", with a "Related Links" section containing "Back to Find List Users".

The "Device Information" section is expanded and contains the following fields:

- Controlled Devices:** A dropdown menu with "CTI1" selected and highlighted by a red box.
- Available Profiles:** An empty dropdown menu.
- CTI Controlled Device Profiles:** An empty dropdown menu.
- Device Association:** A button.
- Line Appearance Association for Presence:** A button.

Below the "Device Information" section is the "Extension Mobility" section, which contains an "Available Profiles" dropdown menu.

### Cisco UCM end user Configuration (Continued)

Check Allow Control of Device from CTI

Select the Primary Extension for this user.5007 is used for this example.

The screenshot shows the Cisco Unified CM Administration interface, continuing from the previous page. The "End User Configuration" page is shown with the "Extension Mobility" section expanded.

The "Extension Mobility" section contains the following fields:

- Available Profiles:** An empty dropdown menu.
- Controlled Profiles:** An empty dropdown menu.
- Default Profile:** A dropdown menu with "-- Not Selected --" selected.
- BLF Presence Group\*:** A dropdown menu with "Standard Presence group" selected.
- SUBSCRIBE Calling Search Space:** A dropdown menu with "< None >" selected.
- Allow Control of Device from CTI:** A checkbox that is checked and highlighted by a red box.
- Enable Extension Mobility Cross Cluster:** An unchecked checkbox.

Below the "Extension Mobility" section is the "Directory Number Associations" section, which contains a "Primary Extension" dropdown menu with "5007" selected and highlighted by a red box.

## Check Enable Mobility

**Mobility Information**

Enable Mobility

Enable Mobile Voice Access

Maximum Wait Time for Desk Pickup\* 10000

Remote Destination Limit\* 4

Remote Destination Profiles

[View Details](#)

**Mutilevel Precedence and Preemption Authorization**

MLPP User Identification Number

MLPP Password

Confirm MLPP Password

MLPP Precedence Authorization Level Default

**CAPF Information**

Associated CAPF Profiles

[View Details](#)

Add the following permissions for Standard Users:

- Standard CCM End-Users
- Standard CTI Enabled
- Standard CCMUSER Administration

**Permissions Information**

Groups

Standard CCM End Users
Standard CTI Enabled

[View Details](#)

Roles

Standard CCM End Users
Standard CCMUSER Administration
Standard CTI Enabled

[View Details](#)

[Add to Access Control Group](#)

[Remove from Access Control Group](#)

[Save](#) [Delete](#) [Add New](#)

**i** \* - indicates required item.

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

Navigation: Cisco Unified CM Administration [Go]  
administrator | Search Documentation | About | Logout

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

**Phone Configuration** | Related Links: Back To Find/List [Go]

Save | Delete | Copy | Reset | Apply Config | Add New

**Association**

- Line [1] - 5007 (no partition)
- Line [2] - Add a new DN

**Phone Type**  
Product Type: CTI Remote Device

**Real-time Device Status**  
Registration: Registered with Cisco Unified Communications Manager clus20sub1  
IPv4 Address:

**Device Information**

Device is Active  
 Device is not trusted

Active Remote Destination: none

Owner User ID\*: user1

Device Name\*: CTI1

Description: CTI1

Device Pool\*: G711 Preferred [View Details](#)

Calling Search Space: < None >

User Hold MOH Audio Source: < None >

Network Hold MOH Audio Source: < None >

Location\*: Hub\_None

### Cisco UCM CTI remote device Configuration (Continued)

User Locale: English, United States

Network Locale: United States

Ignore Presentation Indicators (internal calls only)

**Number Presentation Transformation**

**Caller ID For Calls From This Phone**

Calling Party Transformation CSS: < None >

Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

**Remote Number**

Calling Party Transformation CSS: < None >

Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

**Protocol Specific Information**

BLF Presence Group\*: Standard Presence group

SUBSCRIBE Calling Search Space: < None >

Rerouting Calling Search Space: < None >

Set RD\*= 4004. This is used for this example.4004 is the Alcatel extension.

**Associated Remote Destinations**

Route calls to all remote destinations when client is not connected

Name	Destination Number
JabberRD	4004

[Add a New Remote Destination](#)

**Do Not Disturb**

Do Not Disturb

DND Option\*

### Remote Destination Configuration

Set Destination Number\*= 4004. This is used for this example.  
Check Enable Extend and Connect.

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

Navigation: Cisco Unified CM Administration | administrator | Search Documentation | About | Logout

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

**Remote Destination Configuration** | Related Links: Back To Find/List | Go

Save | Delete | Copy | Add New

Line	Line Association
Line [1] - 5007 (no partition)	<input checked="" type="checkbox"/>

**Remote Destination Information**

Name: JabberRD  
Destination Number\*: 4004  
Owner User ID\*: user1

Enable Unified Mobility features  
Remote Destination Profile\*: -- Not Selected --  
Single Number Reach Voicemail Policy\*: Use System Default

Enable Single Number Reach  
Ring this phone and my business phone at the same time when my business line(s) is dialed.

Enable Move to Mobile  
If this is a mobile phone, transfer active calls to this phone when the mobility button on your Cisco IP Phone is pressed.

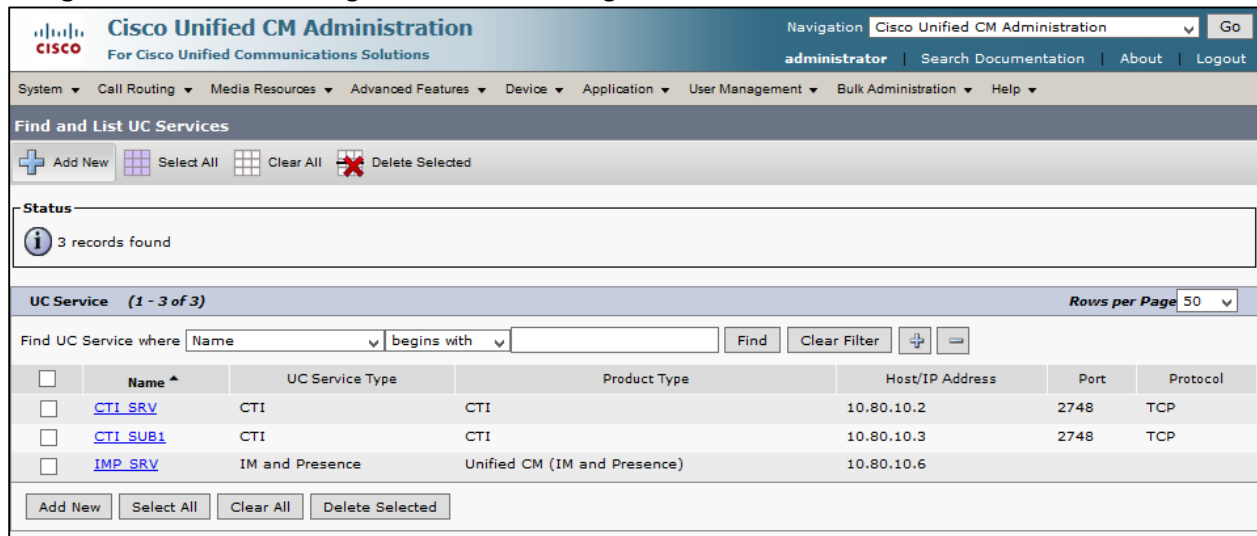
Enable Extend and Connect  
Allow this phone to be controlled by CTI applications (e.g. Jabber)  
CTI Remote Device\*: CTI1

**Timer Information**

Wait\* 4.0 seconds before ringing this phone when my business line is dialed.\*  
Prevent this call from going straight to this phone's voicemail by using a time delay of\* 1.5 seconds to detect calls go straight to voicemail.\*  
Stop ringing this phone after\* 19.0 seconds to avoid connecting to this phone's voicemail.\*

## Cisco UCM UC service Configuration

Navigation Path: User Management → User setting → UC Service



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

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

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

### Find and List UC Services

+ Add New Select All Clear All Delete Selected

Status  
3 records found

UC Service (1 - 3 of 3) Rows per Page 50

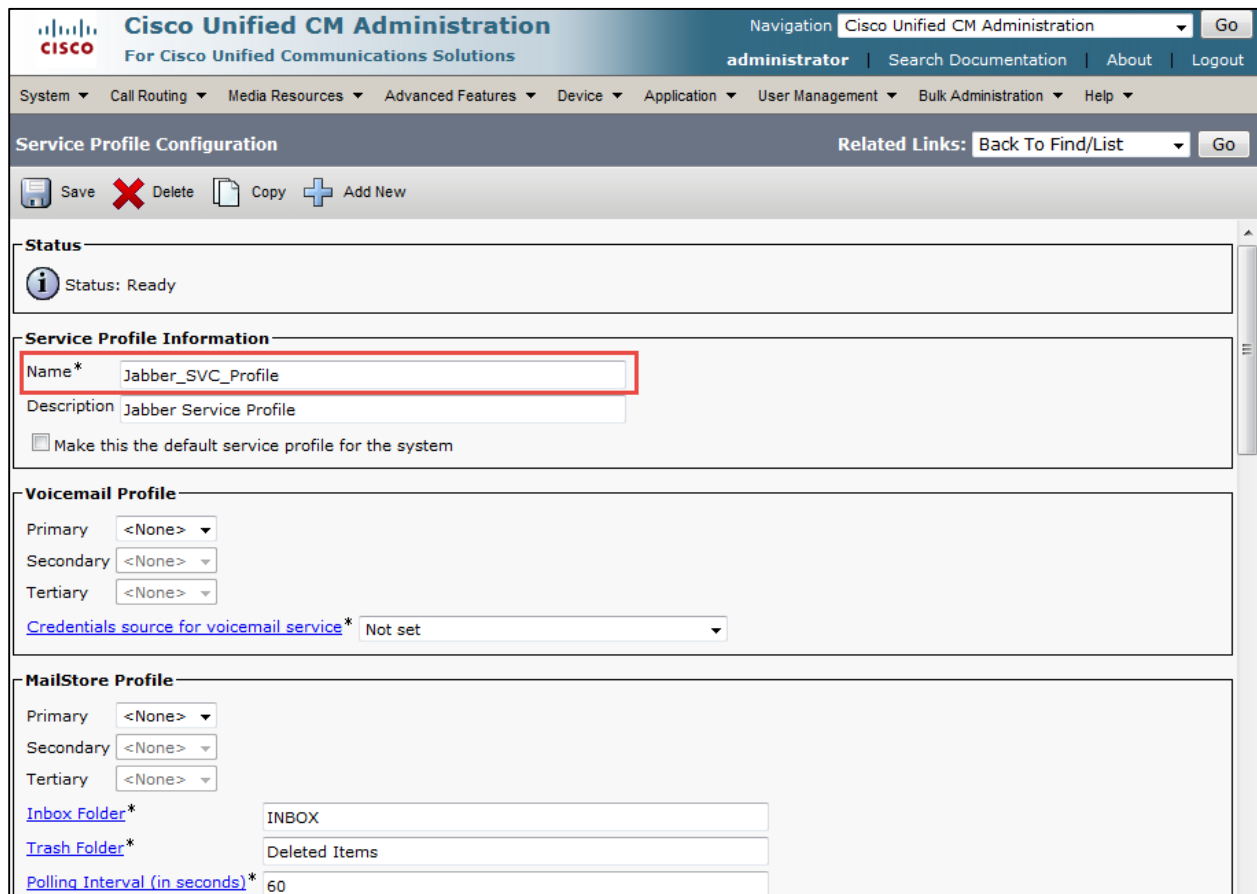
Find UC Service where Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^	UC Service Type	Product Type	Host/IP Address	Port	Protocol
<input type="checkbox"/>	<a href="#">CTI_SRV</a>	CTI	CTI	10.80.10.2	2748	TCP
<input type="checkbox"/>	<a href="#">CTI_SUB1</a>	CTI	CTI	10.80.10.3	2748	TCP
<input type="checkbox"/>	<a href="#">IMP_SRV</a>	IM and Presence	Unified CM (IM and Presence)	10.80.10.6		

Add New Select All Clear All Delete Selected

## Cisco UCM service Profile Configuration

Navigation Path: User Management → User setting → Service Profile



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

Navigation: Cisco Unified CM Administration Go  
administrator | Search Documentation | About | Logout

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

### Service Profile Configuration

Related Links: Back To Find/List Go

Save Delete Copy Add New

Status  
Status: Ready

**Service Profile Information**

Name\* Jabber\_SVC\_Profile  
Description Jabber Service Profile  
 Make this the default service profile for the system

**Voicemail Profile**

Primary <None>  
Secondary <None>  
Tertiary <None>  
Credentials source for voicemail service\* Not set

**MailStore Profile**

Primary <None>  
Secondary <None>  
Tertiary <None>  
Inbox Folder\* INBOX  
Trash Folder\* Deleted Items  
Polling Interval (in seconds)\* 60

## Cisco UCM service profile Configuration (Continued)

[Allow dual folder mode](#)

---

**Conferencing Profile**

Primary

Secondary

Tertiary

Server Certificate Verification

[Credentials source for web conference service\\*](#)

---

**Directory Profile**

Primary

Secondary

Tertiary

[Use UDS for Contact Resolution](#)

[Use Logged On User Credential](#)

[Username](#)

[Password](#)

[Search Base 1](#)

[Search Base 2](#)

[Search Base 3](#)

[Recursive Search on All Search Bases](#)

[Search Timeout \(seconds\)\\*](#)

[Base Filter \(Only used for Advance Directory\)](#)

[Predictive Search Filter \(Only used for Advance Directory\)](#)

---

**IM and Presence Profile**

Primary

Secondary

Tertiary

---

**CTI Profile**

Primary

Secondary

Tertiary

---

**Video Conference Scheduling Portal Profile**

Primary

Secondary

Tertiary

## Cisco Unified CM IM Presence – CCMCIP Profile Configuration

**Navigation Path:** Application → Legacy Clients → CCMCIP Profile

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.

The screenshot displays the Cisco Unified CM IM and Presence Administration interface. The main content area is titled "CCMCIP Profile Configuration". At the top, there are navigation tabs: System, Presence, Messaging, Application, Bulk Administration, Diagnostics, and Help. Below the tabs, there are action buttons: Save, Delete, and Add New. The "Status" section shows "Status: Ready". The "CCMCIP Profile Settings" section contains the following fields:

- Name\*: remotedesk
- Description: (empty)
- Primary CCMCIP Host\*: 10.80.10.2
- Backup CCMCIP Host\*: 10.80.10.3
- Server Certificate Verification\*: Any Certificate
- Make this the default CCMCIP Profile for the system.

The "Users in Profile" section shows a table with the following data:

	User ID	Firstname	Lastname	Department
<input type="checkbox"/>	user1		user1	
<input type="checkbox"/>	user2		user2	

Below the table are buttons: Add Users to Profile, Select All, Clear All, Delete Selected, and a "Rows per Page" dropdown set to 50.



## 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.

The screenshot displays the Cisco Unified CM Administration web interface for configuring a SIP Trunk. The page title is "Trunk Configuration" and the user is logged in as "administrator". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The "Trunk Configuration" section includes a "Related Links" dropdown set to "Back To Find/List" and a "Go" button. Below this are icons for Save, Delete, Reset, and Add New. The configuration is organized into several sections:

- Status:** Shows "Status: Ready".
- SIP Trunk Status:** Shows "Service Status: Unknown - OPTIONS Ping not enabled" and "Duration: Unknown".
- Device Information:** A table of configuration parameters:

Field	Value
Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	IMPTrunk
Description	
Device Pool*	G711 Preferred
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL
Location*	Hub_None
AAR Group	< None >
Tunneled Protocol*	None
QSIG Variant*	No Changes

## Cisco Unified Communications Manager SIP Trunk to CUP Configuration (Continued)

All other values are default.

ASN.1 ROSE OID Encoding*	No Changes
Packet Capture Mode*	None
Packet Capture Duration	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

**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.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**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.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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

Navigation: Cisco Unified CM Administration | Go  
administrator | Search Documentation | About | Logout

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

### Trunk Configuration

Related Links: Back To Find/List | Go

Save | Delete | Reset | Add New

#### SIP Information

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.80.10.6		5060

MTP Preferred Originating Codec\* | 711ulaw |  
BLF Presence Group\* | Standard Presence group |  
SIP Trunk Security Profile\* | Non Secure SIP Trunk Profile |  
Rerouting Calling Search Space | < None > |  
Out-Of-Dialog Refer Calling Search Space | < None > |  
SUBSCRIBE Calling Search Space | < None > |  
SIP Profile\* | Standard SIP Profile | [View Details](#)  
DTMF Signaling Method\* | No Preference |

#### Normalization Script

Normalization Script | < None > |  
 Enable Trace

	Parameter Name	Parameter Value
1		

#### Normalization Script

Normalization Script | < None > |  
 Enable Trace

	Parameter Name	Parameter Value
1		

#### 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

Save | Delete | Reset | Add New

# Cisco Unity Connection

## Cisco Unity Connection Telephony Integration

**Navigation:** Telephony Integrations → Phone system

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

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree with 'Phone System' selected under 'Telephony Integrations'. The main content area is titled 'Phone System Basics (pbxinterop)'. It includes a search bar for phone systems and a 'Related Links' section with 'Add Port Group'. Below this are navigation buttons: Save, Delete, Previous, and Next. The 'Phone System' section contains a text field for 'Phone System Name\*' with the value 'pbxinterop'. There are three checkboxes: 'Default TRAP Phone System' (unchecked), 'Send Message Counts' (checked), and 'Use Same Port for Enabling and Disabling MWIs' (unchecked). A 'Run' button is next to 'Synchronize All MWIs on This Phone System'. The 'Call Loop Detection by Using DTMF' section has two checkboxes (unchecked) and a 'DTMF Tone To Use' dropdown set to 'A'. The 'Guard Time' is set to 2500 milliseconds. The 'Call Loop Detection by Using Extension' section has one checked checkbox: 'Enable for Forwarded Message Notification Calls (by Using Extension)'.

The screenshot shows the 'Phone View Settings' and 'Outgoing Call Restrictions' sections. The left sidebar is the same as the previous screenshot. The 'Phone View Settings' section has an 'Enable Phone View' checkbox (unchecked). Below it are two text fields for 'CTI Phone Access' labeled 'Username' and 'Password'. The 'Outgoing Call Restrictions' section has three radio buttons: 'Enable outgoing calls' (selected), 'Disable all outgoing calls immediately', and 'Disable all outgoing calls between'. The 'Beginning Time' is set to 12:00 AM and the 'Ending Time' is set to 12:00 AM. At the bottom are 'Save', 'Delete', 'Previous', and 'Next' buttons, and a note: 'Fields marked with an asterisk (\*) are required.'

## Port Group

**Navigation:** Telephony Integration → Port Group

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

Check Register with SIP server

The screenshot shows the Cisco Unity Connection Administration interface. The left sidebar contains a navigation tree with 'Port Group' selected under 'Telephony Integrations'. The main content area is titled 'Port Group Basics (pbxinterop-1)'. It includes a 'Display Name\*' field containing 'pbxinterop-1', an 'Integration Method' dropdown set to 'SIP', and a 'Reset Status' section with a 'Reset' button. Below this is the 'Session Initiation Protocol (SIP) Settings' section, which has two unchecked checkboxes: 'Register with SIP Server' and 'Authenticate with SIP Server'. There are also input fields for 'Authentication Username', 'Authentication Password', and 'Contact Line Name'. The 'SIP Security Profile' is set to '5060' and the 'SIP Transport Protocol' is set to 'UDP'. At the bottom, there is an 'Advertised Codec Settings' section with a 'Change Advertising' button and a table with columns 'Display Name' and 'Packet Size'. The table contains one entry: 'G.711 mu-law' with a 'Packet Size' of '20'.

This screenshot shows the 'Message Waiting Indicator Settings' section of the configuration page. It features a 'Change Advertising' button at the top. The 'Enable Message Waiting Indicators' checkbox is checked. Below it are four input fields: 'Delay between Requests' (0 milliseconds), 'Maximum Concurrent Requests' (0), 'Retries After Successful Attempt' (0), and 'Retry Interval After Successful Attempt' (5 milliseconds). At the bottom of this section are 'Save', 'Delete', 'Previous', and 'Next' buttons. A note at the very bottom states: 'Fields marked with an asterisk (\*) are required.'

## Cisco Unity Telephony integration Configuration (Continued)

Navigation Path: Telephony Integration → Port Group → Edit → Servers

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows the navigation tree with 'Port Group' selected under 'Telephony Integrations'. The main content area is titled 'Port Group' and includes 'Edit', 'Refresh', and 'Help' options. A 'Save' button is at the top left. The 'SIP Servers' section contains a table with one server entry highlighted by a red box. The 'TFTP Servers' section also contains a table with one server entry highlighted by a red box. At the bottom, the 'IPv6 Addressing Mode' section has two dropdown menus set to 'IPv4' and a 'Save' button.

Order	IPv4 Address or Host Name	IPv6 Address or Host Name	Port	TLS Port
0	10.80.10.2		5060	5061

Order	IPv4 Address or Host Name	IPv6 Address or Host Name
0	10.80.10.2	

IPv6 Addressing Mode  
Preference for Signaling: IPv4  
Preference for Media: IPv4

## 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

The screenshot displays the Cisco Unity Connection Administration web interface. The left sidebar shows a navigation tree with 'Port' selected under 'Telephony Integrations'. The main content area is titled 'Port Basics (pbxinterop-1-001)'. It features a 'Phone System Port' section with a red border, containing the following fields: 'Enabled' (checked), 'Port Name' (pbxinterop-1-001), 'Phone System' (pbxinterop), 'Port Group' (pbxinterop-1), and 'Server' (clus20unity.lab.tekvizion.com). Below this is the 'Port Behavior' section with 'Answer Calls', 'Perform Message Notification', 'Send MWI Requests', and 'Allow TRAP Connections' all checked. Navigation buttons for 'Save', 'Delete', 'Previous', and 'Next' are present at the top and bottom of the configuration area.

## 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)

The screenshot shows the 'Edit User Basics' configuration page for user 4001. The left sidebar contains a navigation tree with categories like Users, Class of Service, Templates, Contacts, Distribution Lists, Call Management, Message Storage, and Networking. The main content area has a header with 'Edit User Basics (4001)' and navigation buttons (Save, Delete, Previous, Next). The form fields are as follows:

Name	
Alias*	4001
First Name	4001
Last Name	
Display Name	4001
SMTP Address	4001@clus20unity.lab.tekvizion.com
Initials	
Title	
Employee ID	

**LDAP Integration Status**

- Integrate with LDAP Directory
- Do Not Integrate with LDAP Directory

**Phone**

Extension*	4001
Cross-Server Transfer Extension or URI	
Outgoing Fax Number	

The screenshot shows the 'Phone System' configuration page for user 4001. The left sidebar is the same as in the previous screenshot. The main content area has a header with 'Partition' and 'Search Scope' dropdowns. The form fields are as follows:

Partition	clus20unity Partition
Search Scope	clus20unity Search Space
Phone System	pbxinterop
Class of Service	Voice Mail User COS
Active Schedule	Weekdays

Set for Self-enrollment at Next Sign-In

List in Directory

Send Non-Delivery Receipts on Failed Message Delivery

Skip PIN When Calling From a Known Extension  
**Caution!** Security risk. See Help for This Page for details.

Use Short Calendar Caching Poll Interval

Recorded Name

**Location**

Address	
Building	
City	
State	
Postal Code	
Country	United States

Use System Default Time Zone

Time Zone

Language  Use System Default Language

## Cisco Unity Connection User Configuration (Continued)

All values are default.

The screenshot displays the Cisco Unity Connection Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unity Connection Administration", and the tagline "For Cisco Unified Communications Solutions". The user is logged in as "administrator". The left sidebar shows a navigation tree with "Users" selected. The main content area displays the "Location" configuration page for a user. The fields are as follows:

- Address:
- Building:
- City:
- State:
- Postal Code:
- Country:
- Use System Default Time Zone
- Time Zone:
- Use System Default Language
- 
- Department:
- Manager:
- Billing ID:
- Corporate Email Address:
- Generate SMTP Proxy Address From Corporate Email Address
- Directory URI:
- Corporate Phone Number:

At the bottom of the form, there are buttons for "Save", "Delete", "Previous", and "Next". A note at the bottom states: "Fields marked with an asterisk (\*) are required."

## Acronyms

<b>Acronym</b>	<b>Definition</b>
AAR	Automatic Alternate route
CCNR	Call Completion on No Reply
CFB	Call Forwarding on Busy
CFNA	Call Forwarding No Answer
CFU	Call Forwarding Unconditional
Cisco UCM	Cisco Unified Communications Manager
CLIP	Calling Line (Number) Identification Presentation
CLIR	Calling Line (Number) Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line (Number) Identification Presentation
COLR	Connected Line (Number) Identification Restriction
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
CUP	Cisco Unified Presence
DNS	Domain Name Server
EXT	Extension
FAC	Feature Access Code
FQDN	Fully Qualified Domain Name
MRGL	Media Resource Group List
MTP	Media Termination Point
MWI	Message Waiting Indicator
NOE	New office environment Protocol
PBX	Private Branch Exchange
PSTN	Public Switched Telephone Network
RTP	Real Time Protocol
SCCP	Skinny Client Control Protocol
SIP	Session Initiated Protocol
UDP	Uniform Dial Plan
VM	Voice Mail