# 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.

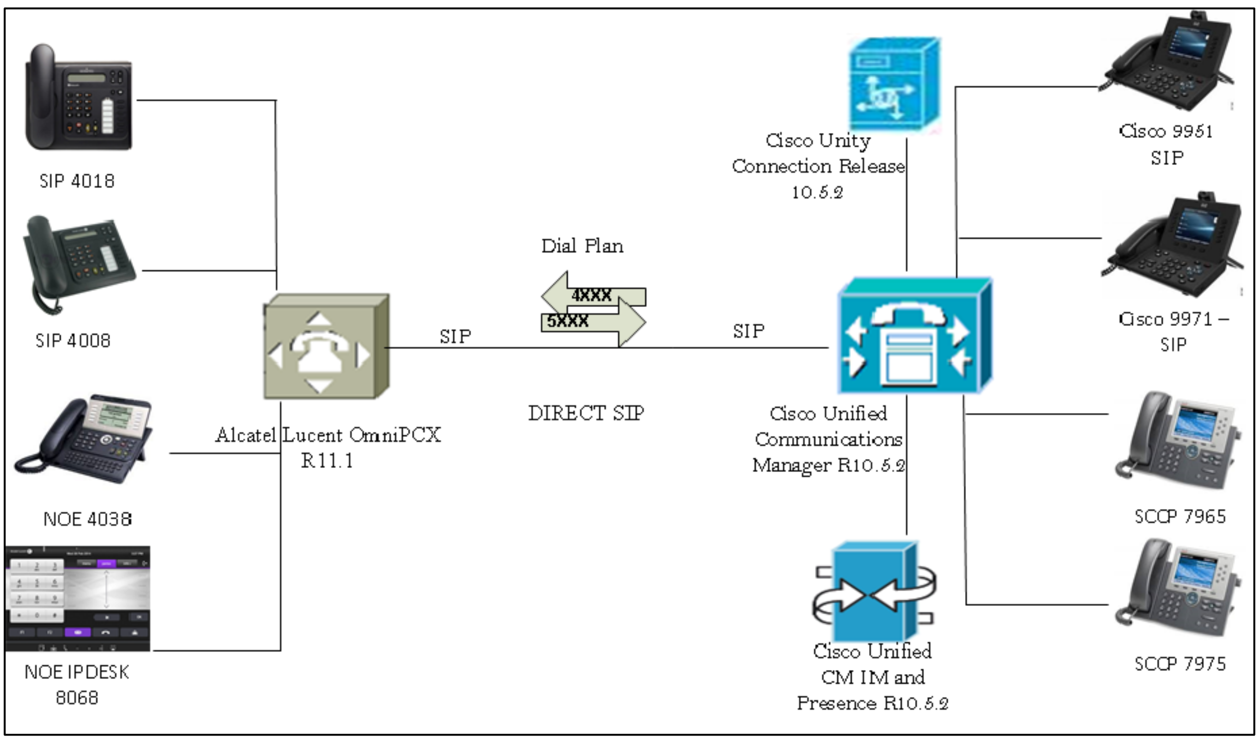
- For call forwarding scenarios where the Alcatel station is the forwarding station, it is required that the Cisco UCM have the "Redirect by Application" checkbox enabled in the SIP Profile used by the SIP Trunk to the Alcatel PBX.
- Retrieval of a parked call on the Alcatel PBX from the Cisco UCM is 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-l1.301-27-us-c0s1

Business identification: R11.1

Release:
DELIVERY l1.301
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 objectqqqqqqqqqqqqqqqqqk
x                                  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                                  x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

### 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 domainqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
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
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

### 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

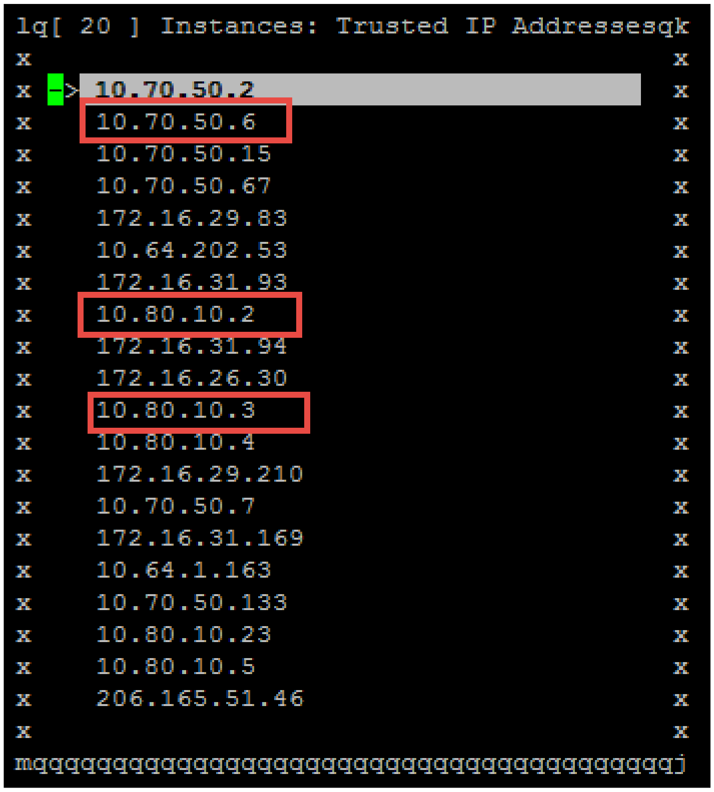
T2 Specification: SIP

Associated Ext SIP gateway: 2

```
lqReview/Modify: Trunk Groupsqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                                   x
x             Node Number (reserved) : 101                                          x
x                     Trunk Group ID : 1                                            x
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-F                                        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
x                                 Implicit Priority                                 x
x                                                                                   x
x                    Activation mode : 0                                            x
x                     Priority Level : 0                                            x
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
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

### 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.




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

**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 Gatewayqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                                      x
x             Node Number (reserved) : 101                                             x
x                Instance (reserved) : 1                                               x
x            SIP External Gateway ID : 2                                               x
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                                                                                      x
x                  Outgoing Password : --------------------                            x
x                            Confirm : --------------------                            x
x                                                                                      x
x                  Incoming username : --------------------------------------------------x
x                                                                                      x
x                  Incoming Password : --------------------                            x
x                            Confirm : --------------------                            x
x                                                                                      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                                                                                      x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

```
x                       Gateway type + Standard type                                   x
x  Re-Trans No. for REGISTER/OPTIONS : 2                                               x
x   P-Asserted-ID in Calling Number + True                                             x
x      Trusted P-Asserted-ID header + True                                             x
x    Diversion Info to provide via  + Diversion                                        x
x Proxy identification on IP address + False                                           x
x                Outbound calls only + False                                           x
x         SDP relay on Ext. Call Fwd + Default                                         x
x          SDP Transparency Override + False                                           x
x RFC 5009 supported / Outbound call + Not Supported                                   x
x           Nonce caching activation + NO                                              x
x                 FAX Procedure Type + T38 only                                        x
x  DNS SRV/Call retry on busy server : 0                                               x
x        Unattended Transfer for RSI + NO                                              x
x          Redirection functionality + NO                                              x
x                  Attended Transfer + NO                                              x
x                  Send BYE on REFER + YES                                             x
x        Support UTF8 characters set + NO                                              x
x        CSTA User-to-User supported + NO                                              x
x                Trusted From header + True                                            x
x      Support Re-invite without SDP + False                                           x
x          Type of codec negotiation + Default                                         x
x                                                                                      x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

**Configure SIP External Gateway for CUCM Publisher**

SIP Remote domain: 10.80.10.2 (CUCM_PUBLISHER)

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

Node Number: 101, this is used in this example

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

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

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

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

Dynamic Payload type for DTMF: 101

Outbound Calls 100 REL + Supported

Incoming Calls 100 REL + Not Requested

P-Asserted-ID in Calling Number + True

Trusted P-Asserted-ID header + True

Diversion Info to provide via + Diversion

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

Ignore inactive/Black hole + True, this is used in this example

```
lqReview/Modify: SIP Ext Gatewayqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                                    x
x           Node Number (reserved) : 101                                             x
x              Instance (reserved) : 1                                               x
x          SIP External Gateway ID : 4                                               x
x                                                                                    x
x                     Gateway Name : CUCM_PUB                                        x
x                SIP Remote domain : 10.80.10.2                                      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                                                                                    x
x                Outgoing Password : --------------------                            x
x                          Confirm : --------------------                            x
x                                                                                    x
x                Incoming username : ---------------------------------------------------- x
x                                                                                    x
x                Incoming Password : --------------------                            x
x                          Confirm : --------------------                            x
x                                                                                    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 + False                                          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                                                                                    x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

```
lqReview/Modify: SIP Ext Gatewayqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                                  x
x                                                                                  x
x                  Incoming Password : --------------------                        x
x                            Confirm : --------------------                        x
x                                                                                  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 + False                                       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             Outbound Calls 100 REL + Supported                                   x
x             Incoming Calls 100 REL + Not Requested                               x
x                       Gateway type + Standard type                               x
x  Re-Trans No. for REGISTER/OPTIONS : 2                                           x
x    P-Asserted-ID in Calling Number + True                                        x
x       Trusted P-Asserted-ID header + True                                        x
x     Diversion Info to provide via  + Diversion                                   x
x Proxy identification on IP address + False                                       x
x                Outbound calls only + False                                       x
x         SDP relay on Ext. Call Fwd + Default                                     x
x          SDP Transparency Override + False                                       x
x RFC 5009 supported / Outbound call + Not Supported                               x
x           Nonce caching activation + NO                                          x
x                 FAX Procedure Type + T38 only                                    x
x  DNS SRV/Call retry on busy server : 0                                           x
x         Unattended Transfer for RSI + NO                                         x
x          Redirection functionality + NO                                          x
x                  Attended Transfer + NO                                          x
x                  Send BYE on REFER + YES                                         x
x         Support UTF8 characters set + NO                                         x
x         CSTA User-to-User supported + NO                                         x
x                Trusted From header + True                                        x
x       Support Re-invite without SDP + False                                      x
x           Type of codec negotiation + Default                                    x
x                                                                                  x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

### Configure Network Routing Table

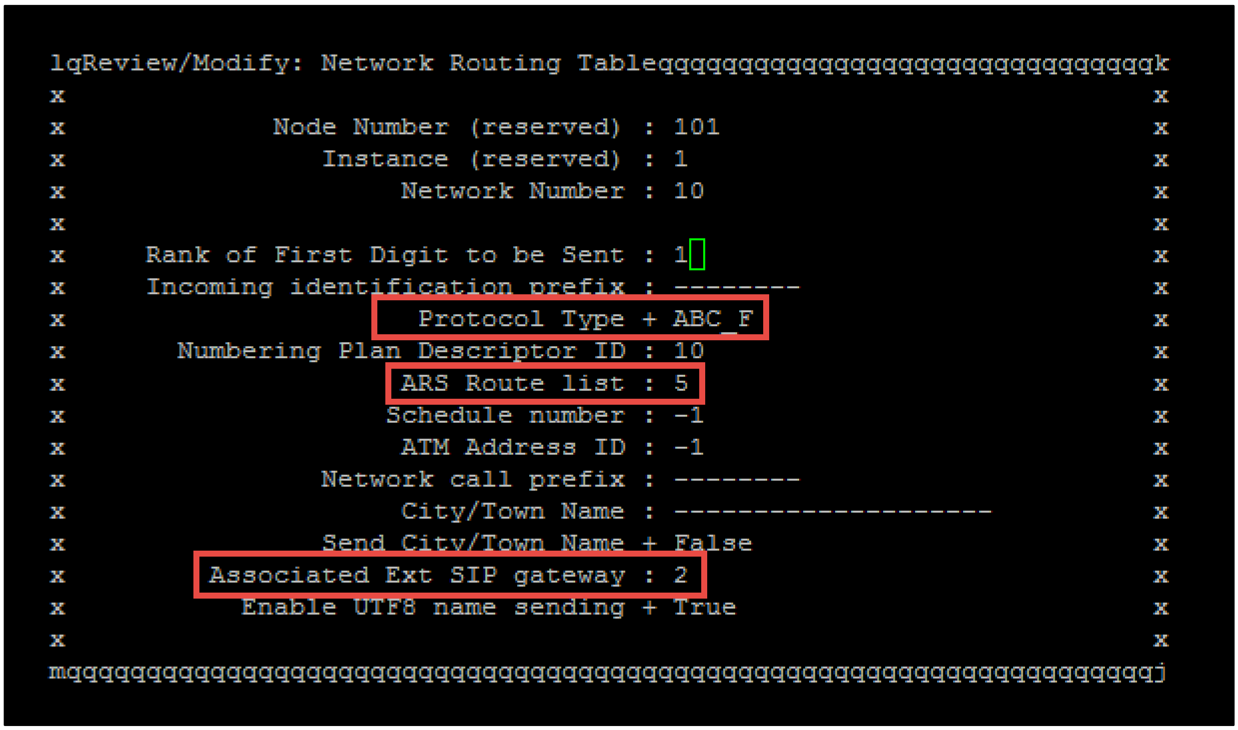
**Navigation:** Translator → Network Routing Table

Node Number: 101, this is used in this example

Protocol Type: ABC-F

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

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




### Configure Prefix Plan – Pointing to Cisco Phones.

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

Node Number: 101, this is used in this example

Prefix Meaning + Routing No.

Network Number: 10, this is used in this example

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

Number of Digits: 4

Note: Cisco extension are started with 5 as prefix digit.

```
lqReview/Modify: Prefix Planqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                          x
x            Node Number (reserved) : 101                                  x
x               Instance (reserved) : 1                                    x
x                            Number : 5                                    x
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                                                                          x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

**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 Planqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                          x
x            Node Number (reserved) : 101                                  x
x               Instance (reserved) : 1                                    x
x                            Number : 7                                    x
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                                                                          x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

**Configure Privacy**

Privacy for particular user

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

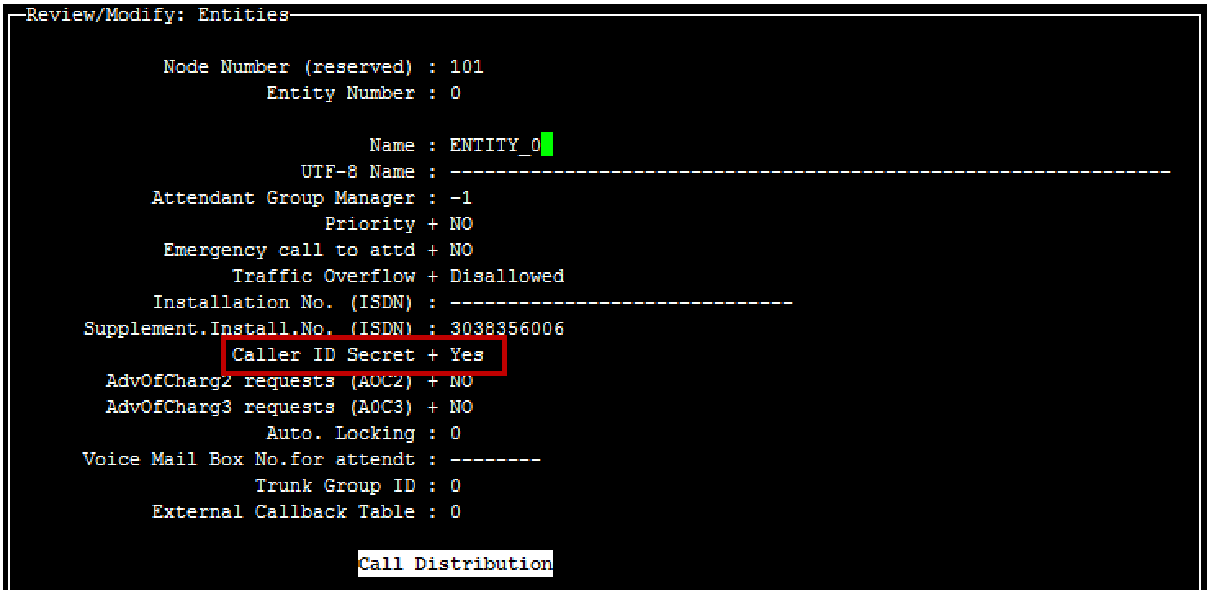
Set **Consistent with Identity:** False i.e., Secret Identity Enabled

```
lqReview/Modify: Dynamic State Userqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                  x
x             Node Number (reserved) : 101                         x
x                   Directory Number : 4004                        x
x                Instance (reserved) : 1                           x
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                                                                  x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

Enable Privacy in Entity Privacy for particular user

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

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




Enable Privacy in COS

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

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

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

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

```
lqReview/Modify: Phone Features COSqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                         x
x            Node Number (reserved) : 101                                 x
x               Instance (reserved) : 1                                   x
x                Phone Features COS : 1                                   x
x                                                                         x
x                    Phone COS Name : -------------------                 x
x                                                                         x
x                                  Rights                                 x
x                                                                         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-l 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                                                                         x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

**Configure Call Park**

a. Authorization : To enable the rights on the user to be able to use the call park/retreive feature
**Navigation**: Classes of Service→ Phone Features COS→ Review/Modify → External Services → Park Call/Retrieve: 1

```
                        External Services

               Direct trunk seizure : 0
              Business account code : 0
                 Redial last number : 1
            Night service answering : 0
              DTMF Frequencies test : 0
                 Park Call/Retrieve : 1
Waiting call Consultation / Access : 0
          Rotary End-to-End Signal. : 0
            DTMF End-to-End Signal. : 1
                     Malicious call : 0
                        Common Hold : 0
                   Priority of Call : 0
                    Secret/Identity : 1
                Alphanumeric Paging : 0
                        Manual Hold : 0

                            Suffixes
```

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

```
lqReview/Modify: Prefix Planqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
x                                                                        x
x             Node Number (reserved) : 101                               x
x                Instance (reserved) : 1                                 x
x                             Number : *75                               x
x                                                                        x
x                     Prefix Meaning + External Features                 x
x                  External Features + Park Call/Retrieve                x
x                                                                        x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

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 domainqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqk
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 + 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                                                                                  x
mqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqqj
```

**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: tekvizionoxe

SIP Authentication: 4003

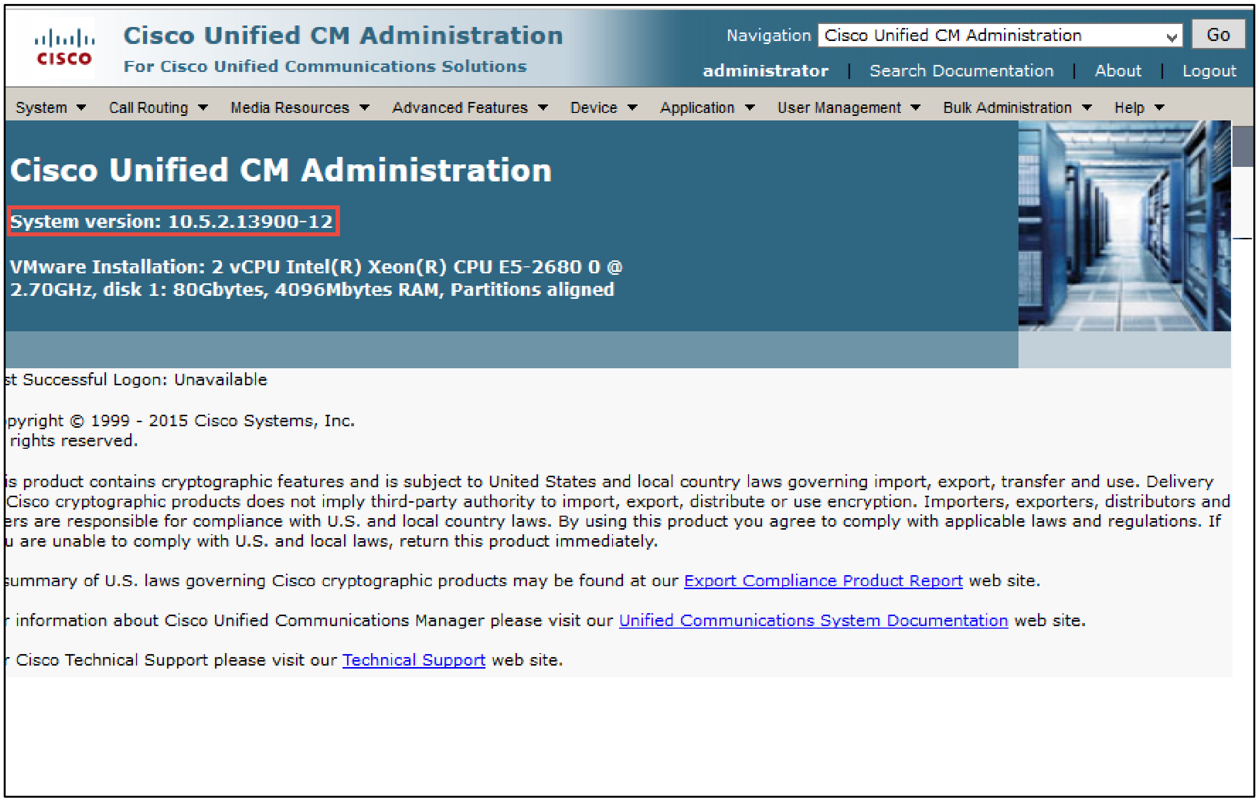
SIP Password: ******

Confirm: ******

```
                         Explicit Priority

                 Activation mode : 0
                  Priority Level : 0

  Pre-emptable Primary Inc. Line + NO
Pre-emptable Secondary Inc. Line + NO
           Priority Presentation + NO
                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 : ******
                         Confirm : ******

      Called Associated DECT set : --------
      Dial by name and text msg. + NO
                 Text msg number : 8
```

# Configuring the Cisco Unified Communications Manager

Cisco Unified Communications Manager Software Version



## Cisco Unified Communications Manager SIP Trunk Security Profile

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

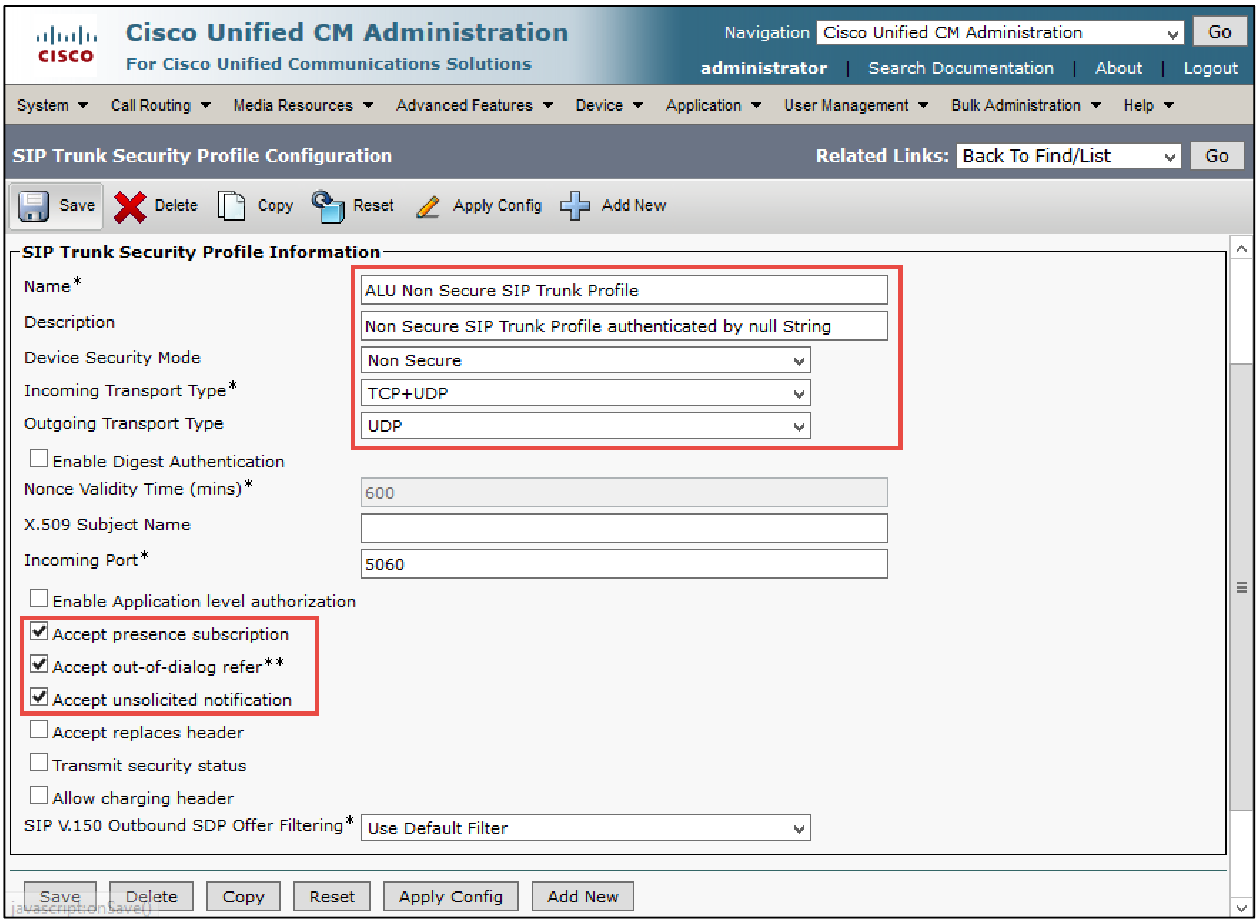
Set Name*= Non Secure SIP Trunk Profile.  This is used for this example.

Set Description = Non Secure SIP Trunk Profile authenticated by null String

Check Accept out of dialog refer

Check Accept unsolicited notification

All other values are default.




Note: Alcatel sends refer-to with replaces header for transfer scenarios, Cisco sends 403 forbidden but there were no issues in the call flow.

## Cisco Unified Communications Manager SIP Trunk Security Profile for Unity Connection

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

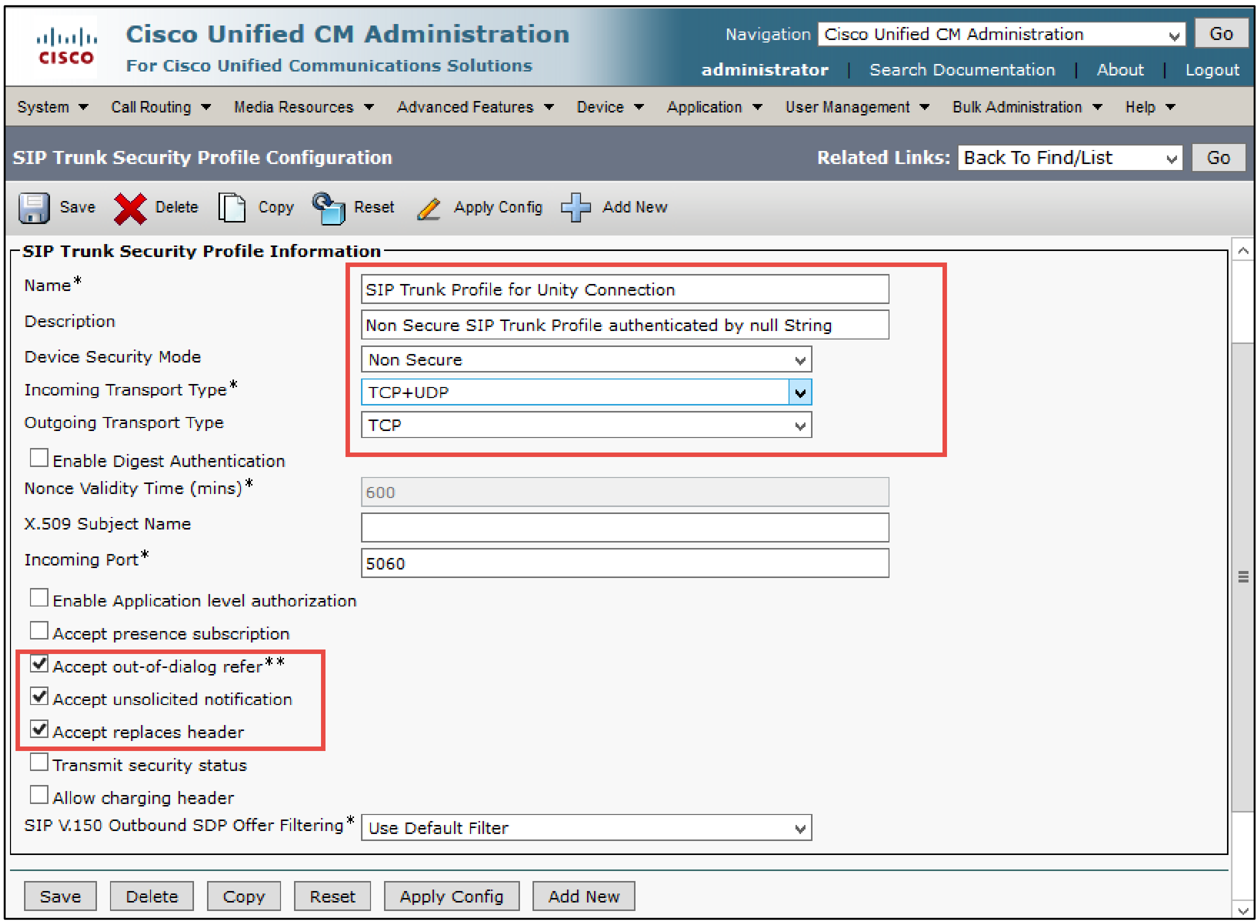
Set Description = this text is used to identify this SIP Trunk Security Profile.

Check Accept presence subscription

Check Accept out of dialog refer**

Check Accept unsolicited notification

All other values are default.




## Cisco Unified Communications Manager SIP Profile

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

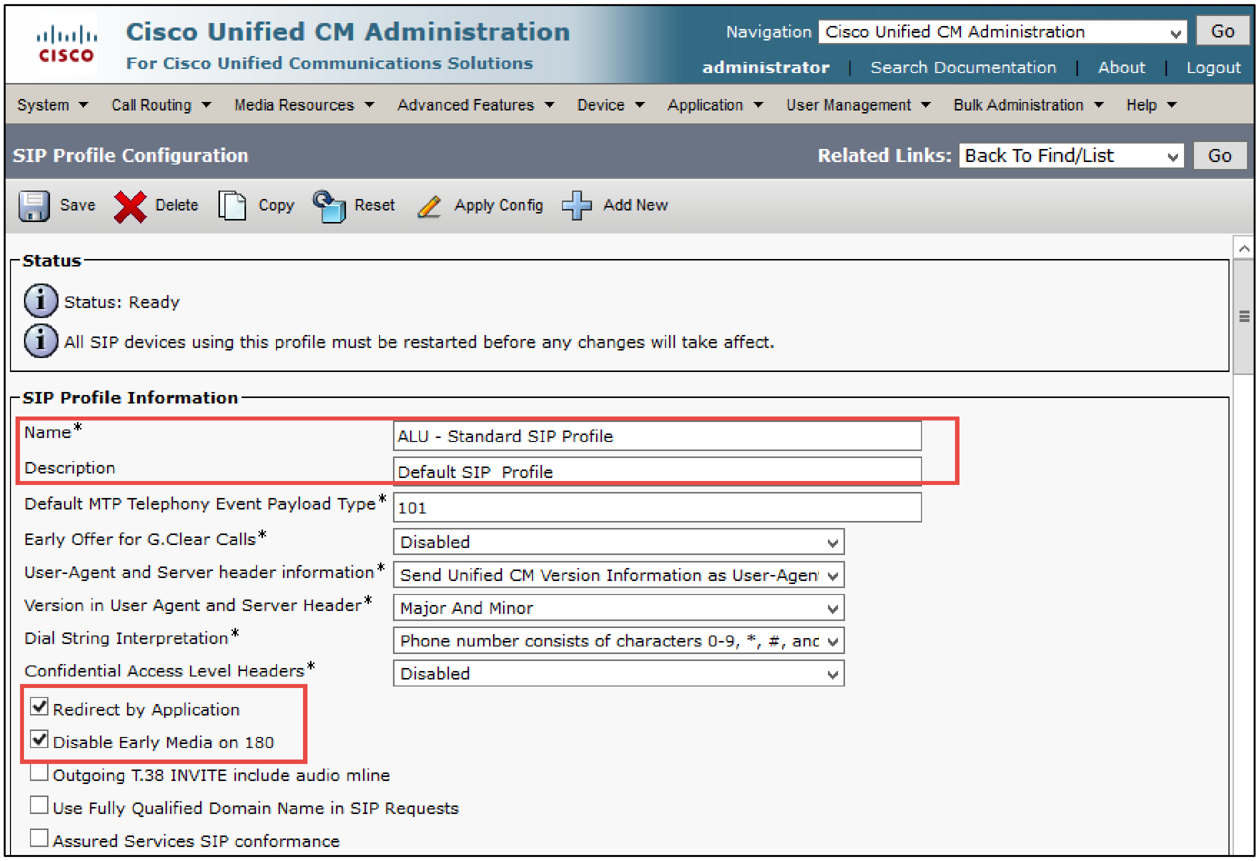
Set Name*= ALU - Standard SIP Profile.  This is used for this example.

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

Check Disable Early Media on 180

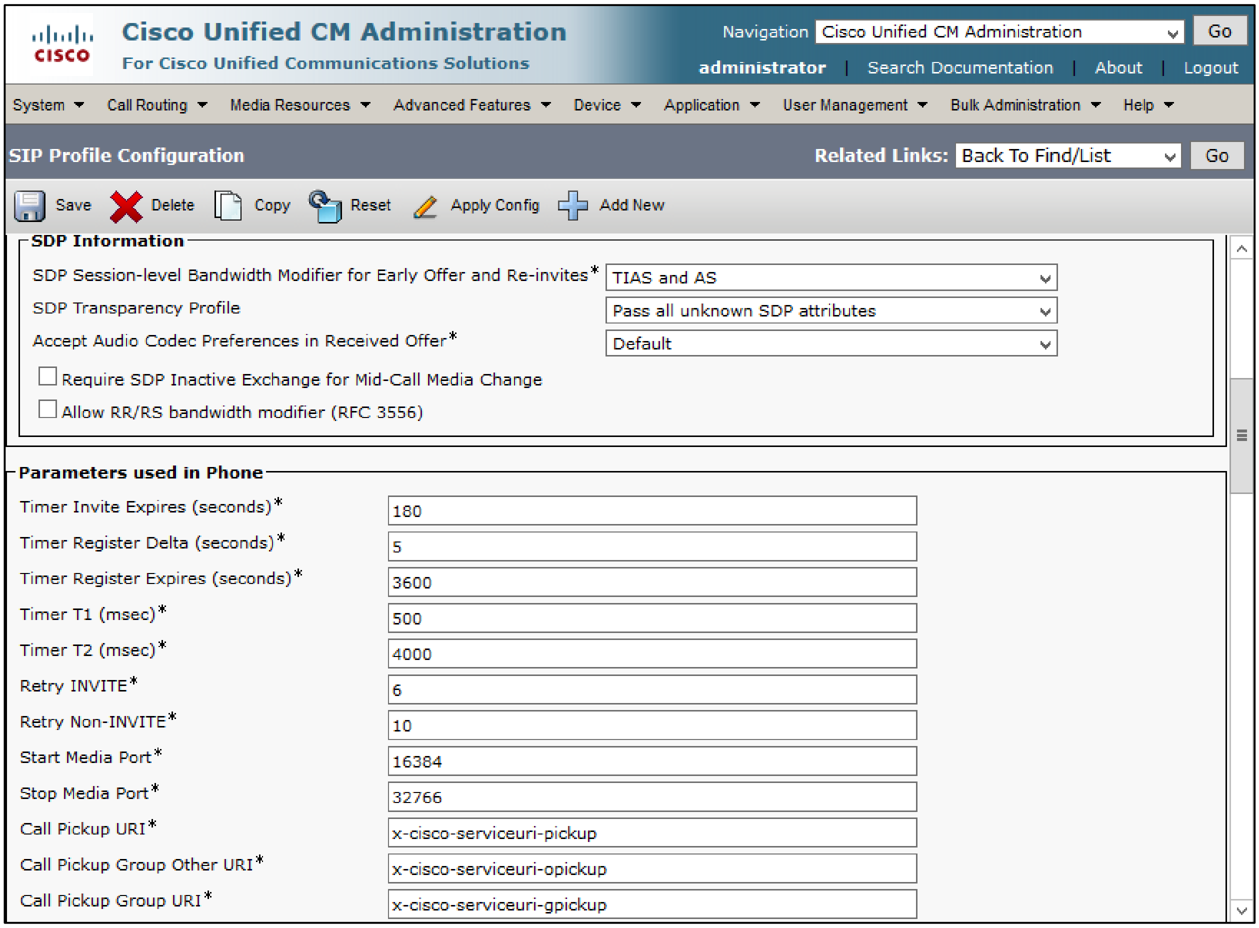
Check Redirect by Application

All other values are default.



**Cisco Unified Communications Manager SIP Profile (Continued)**

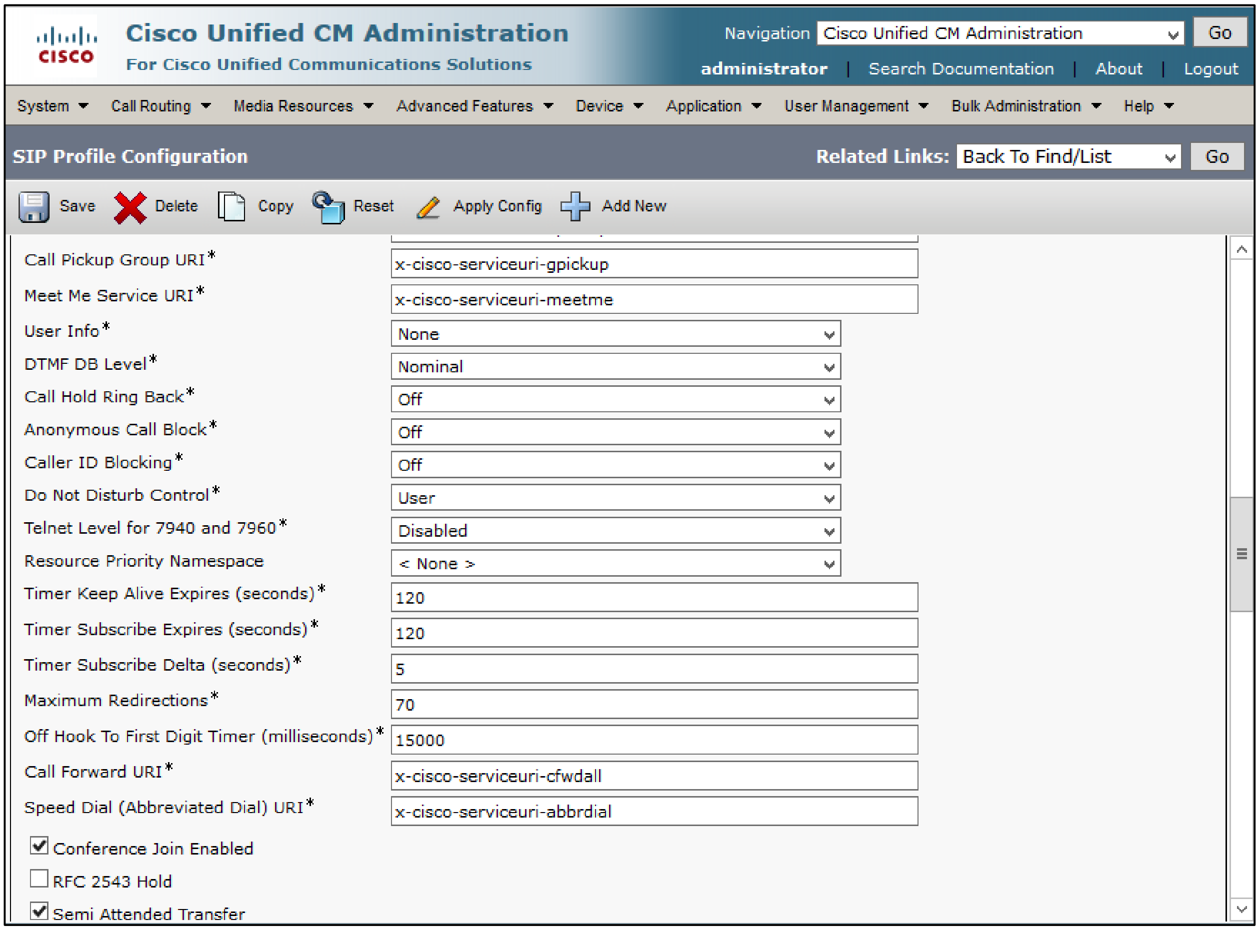
These values are default.



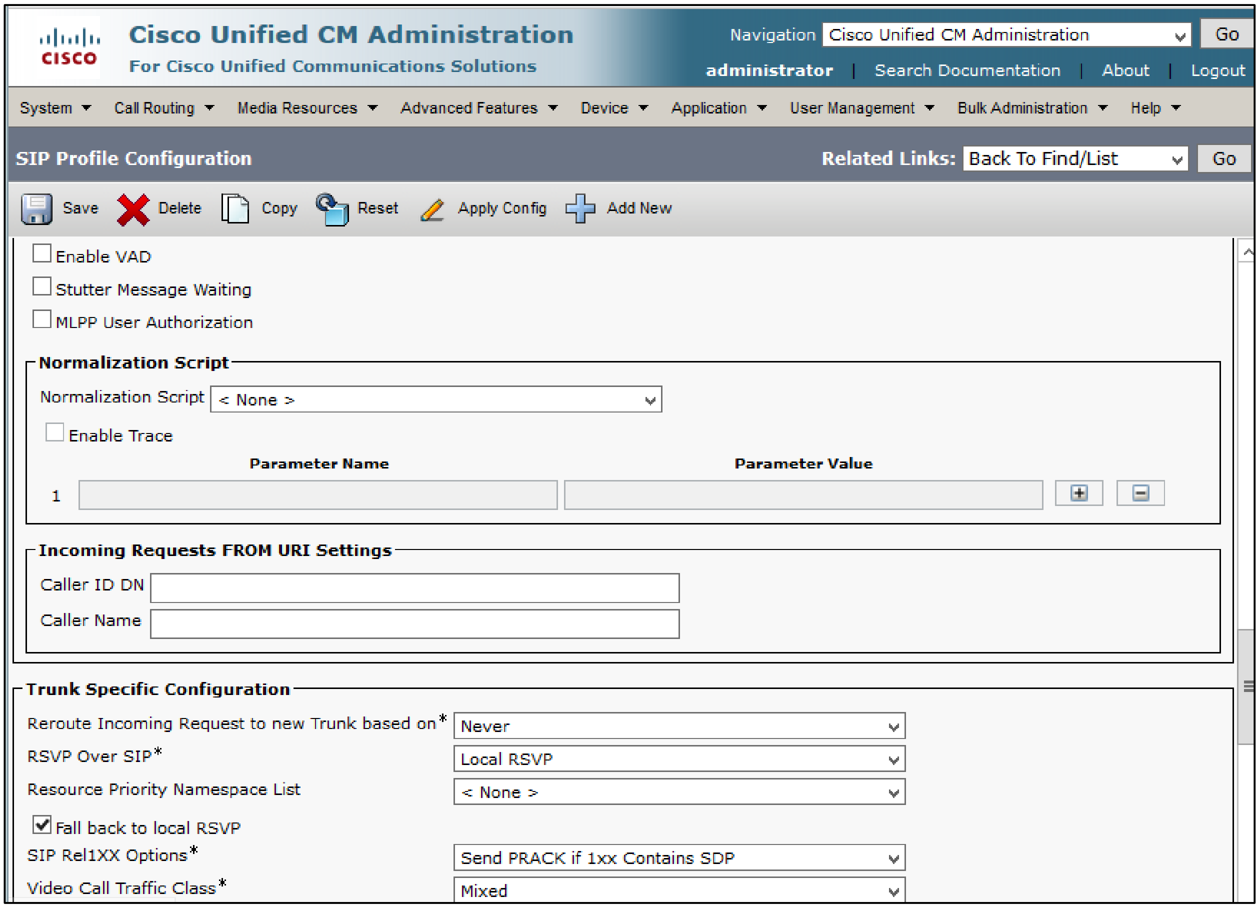

Set SIP Rel1XX Options* = Send PRACK if 1xx Contains SDP

All other values are default.

## Cisco Unified Communications Manager SIP Profile (Continued)



## Cisco Unified Communications Manager SIP Profile (Continued)



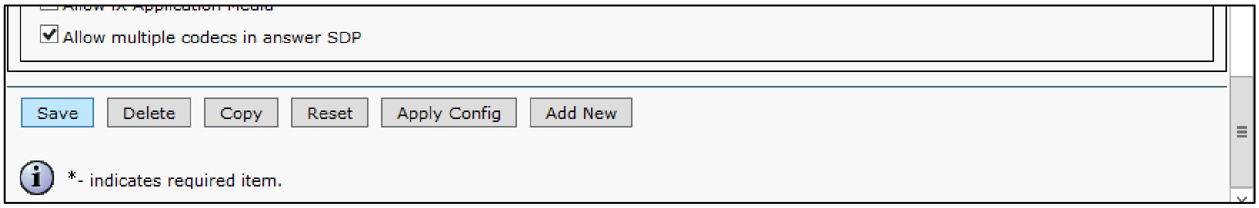
**Cisco Unified Communications Manager SIP Profile (Continued)**

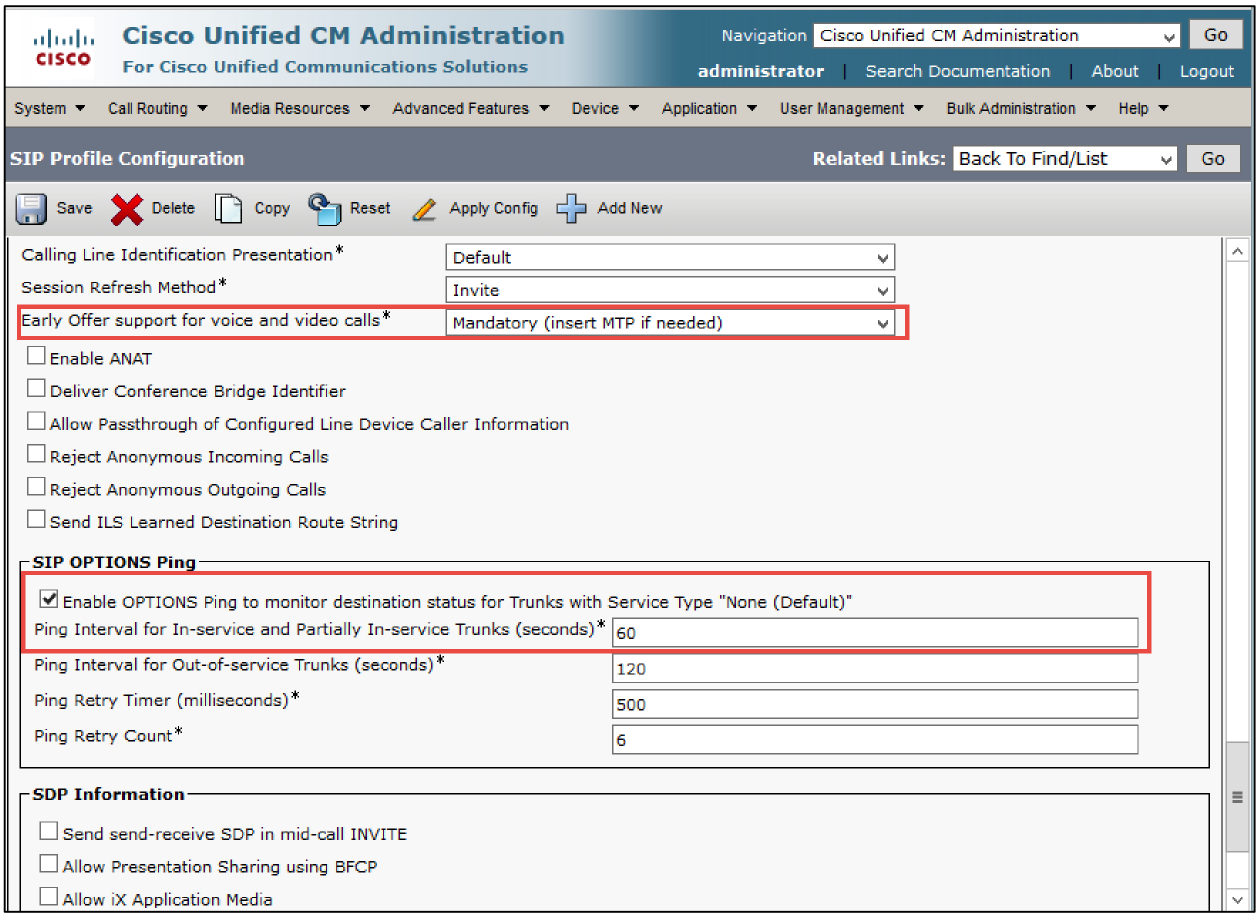
Early Offer support for voice and video calls: Mandatory (insert MTP if needed)

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

Check Send send-receive SDP in mid-call INVITE

All other values are default.



## Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration

**Navigation:** Device → Trunk

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

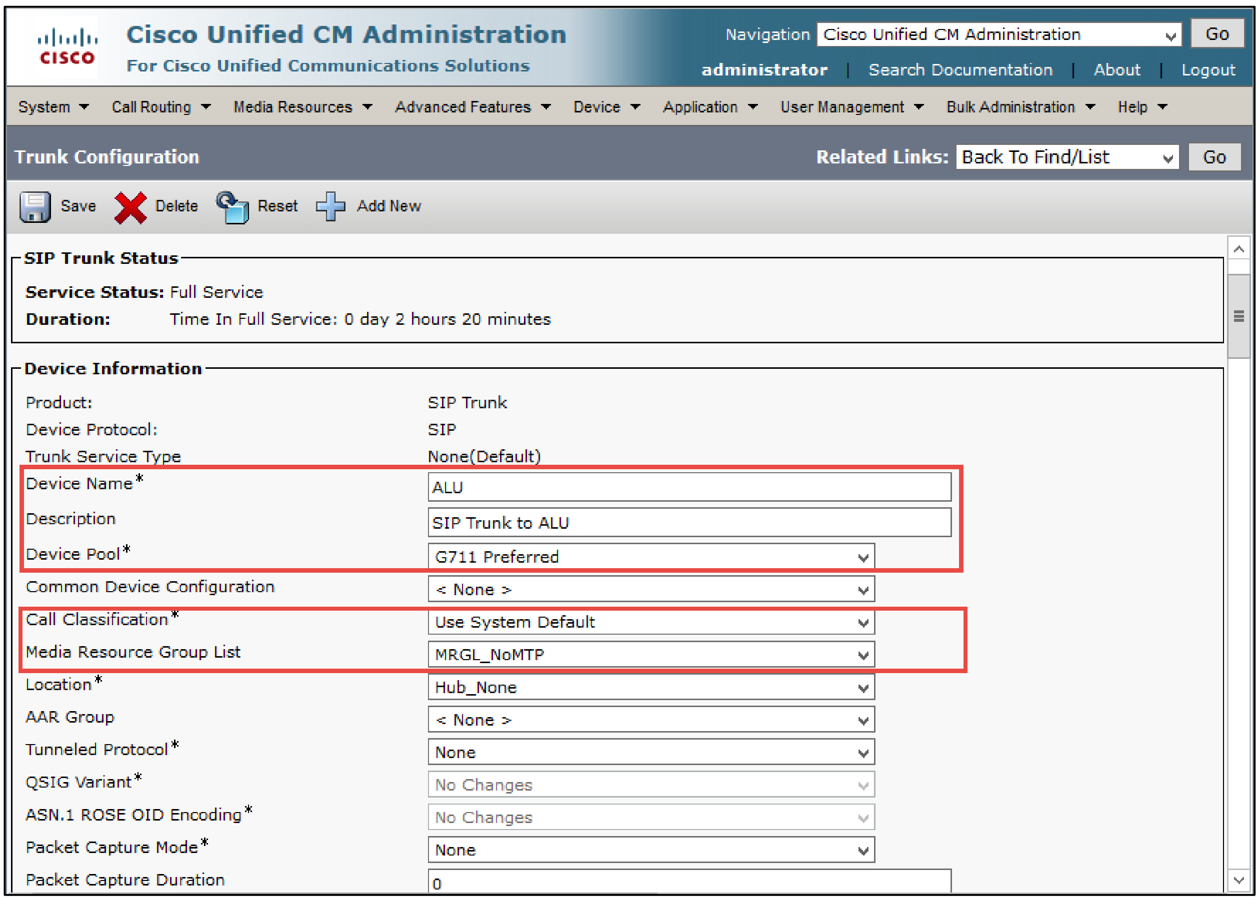
Set Description = this text is used to identify this Trunk Group.

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

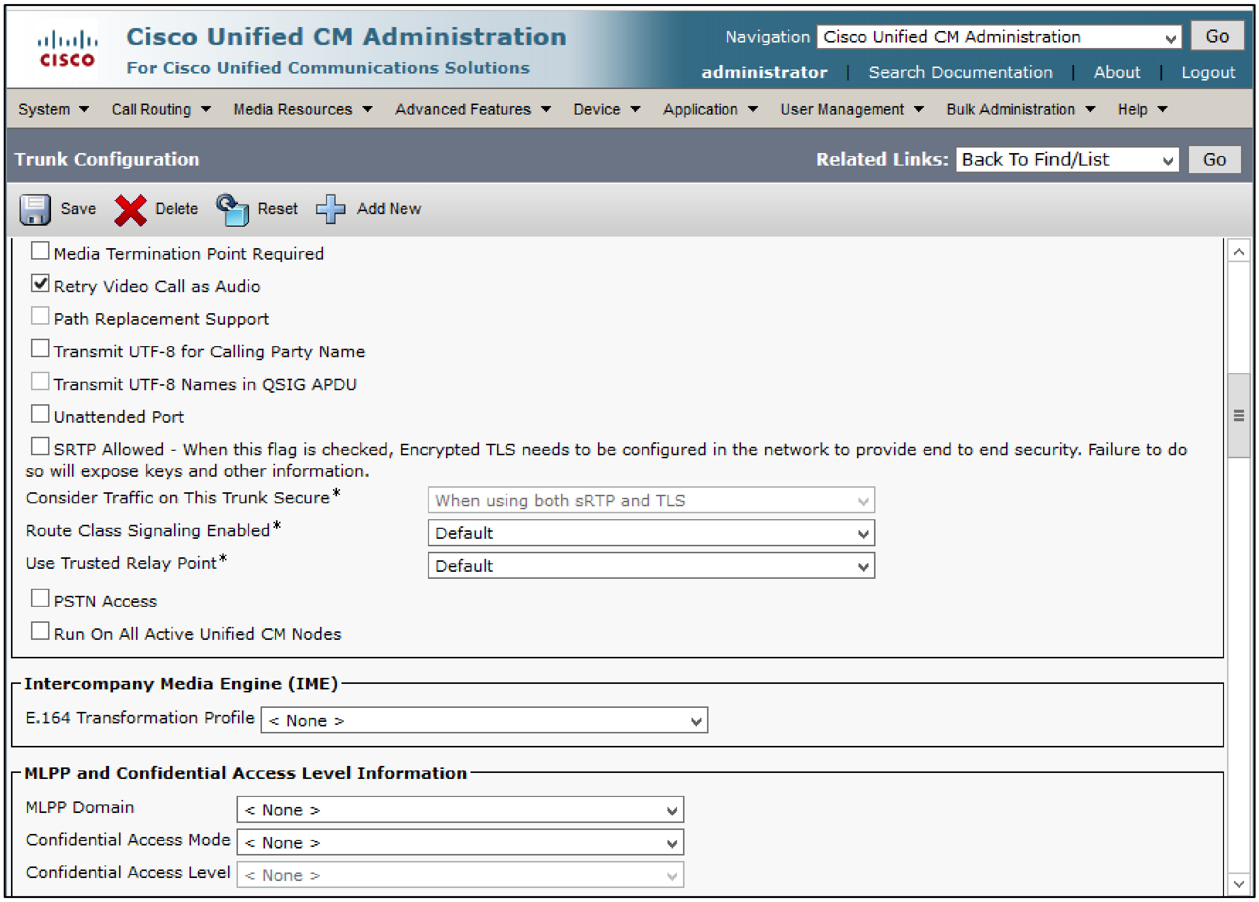
Set Call Classification*= Use System Default. This is used for this example

Set Media Resource Group List = MRGL_NoMTP. This is used for this example

All other values are default.

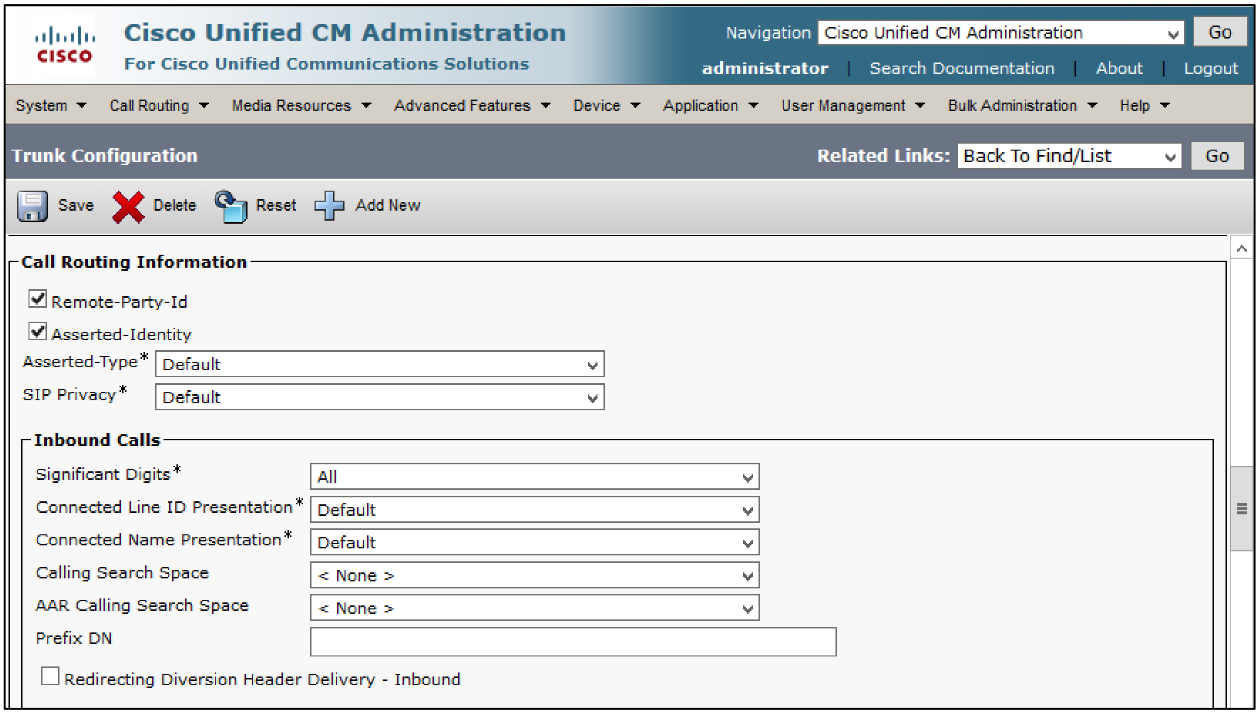


**Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)**

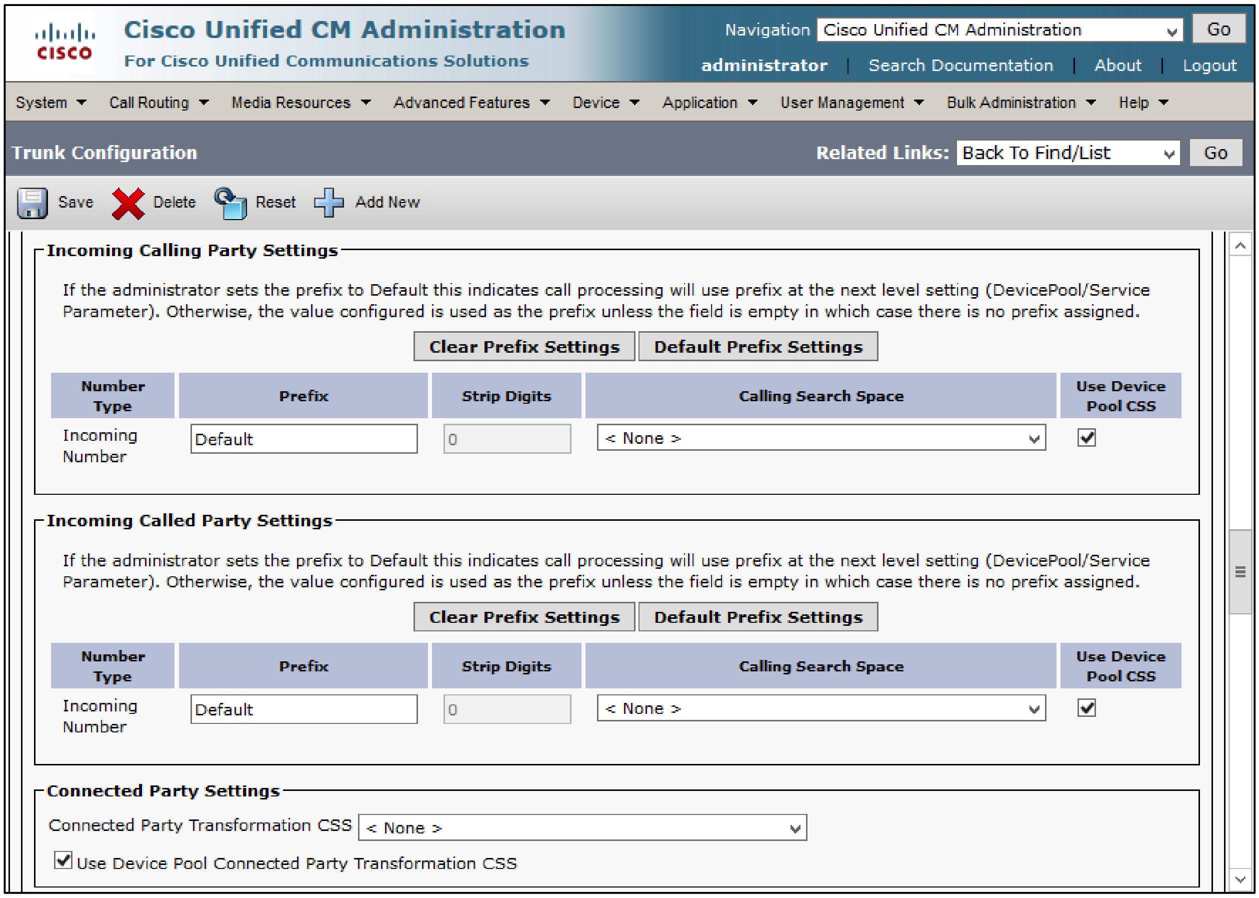



All other values are default.

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



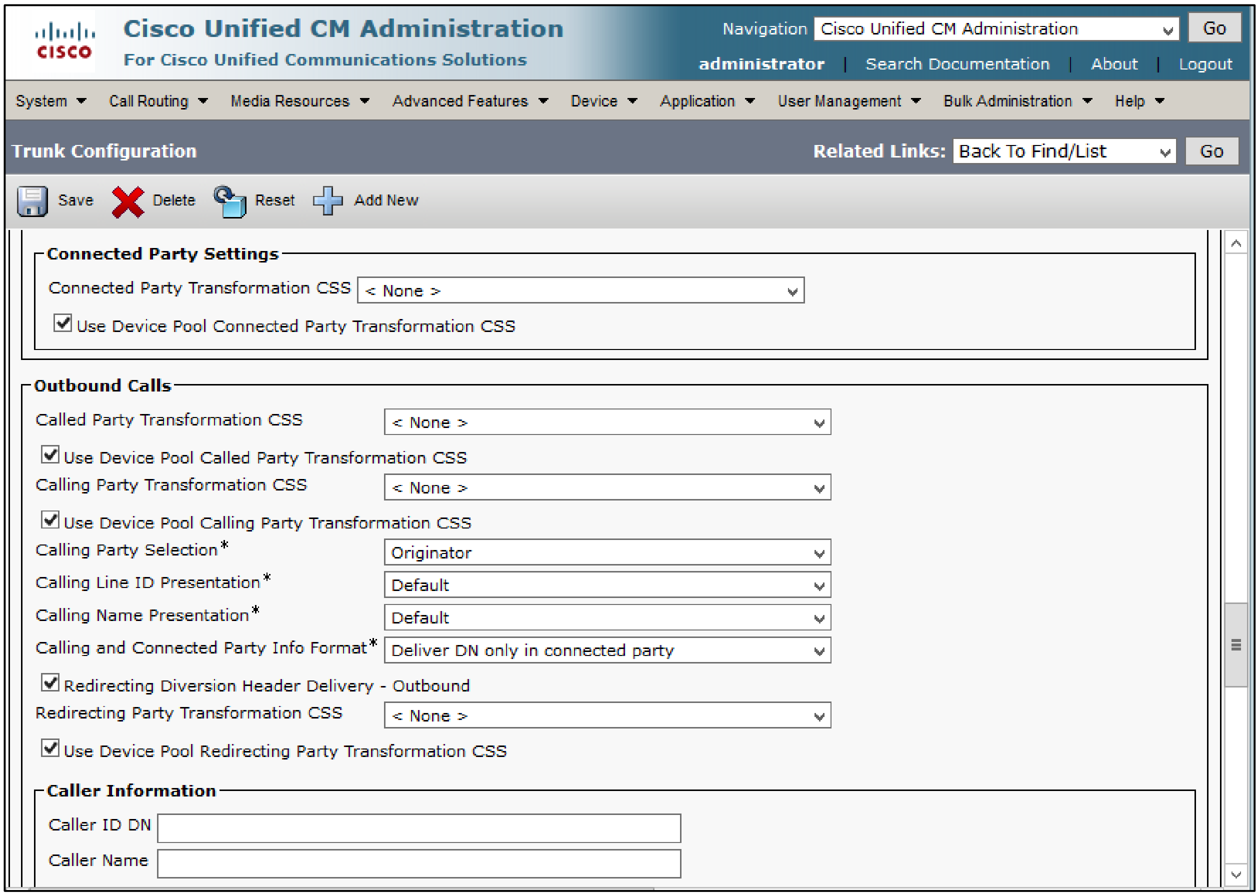
**Cisco Unified Communications Manager SIP Trunk to Alcatel Configuration (Continued)**




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

Check Redirecting Diversion Header Delivery - Outbound

All other values are default.




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

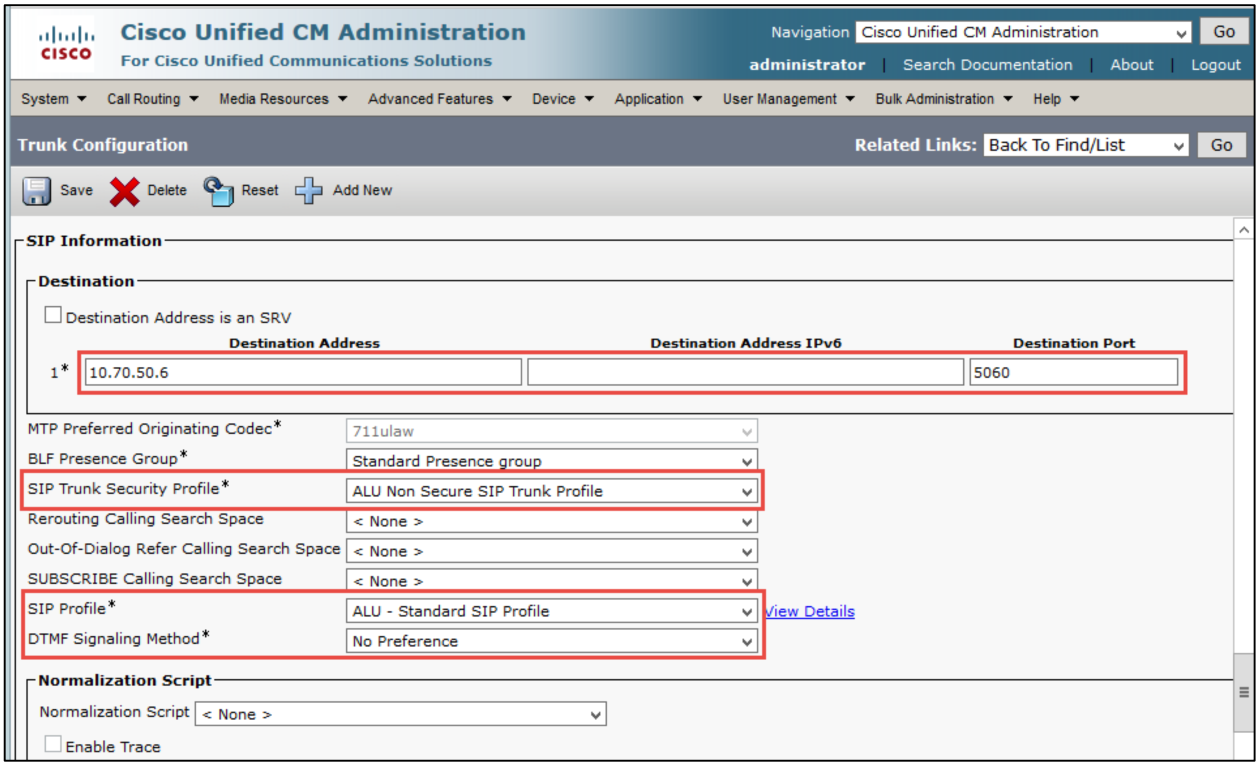
Set Destination Address = 10.70.50.6. This is used in this example.

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

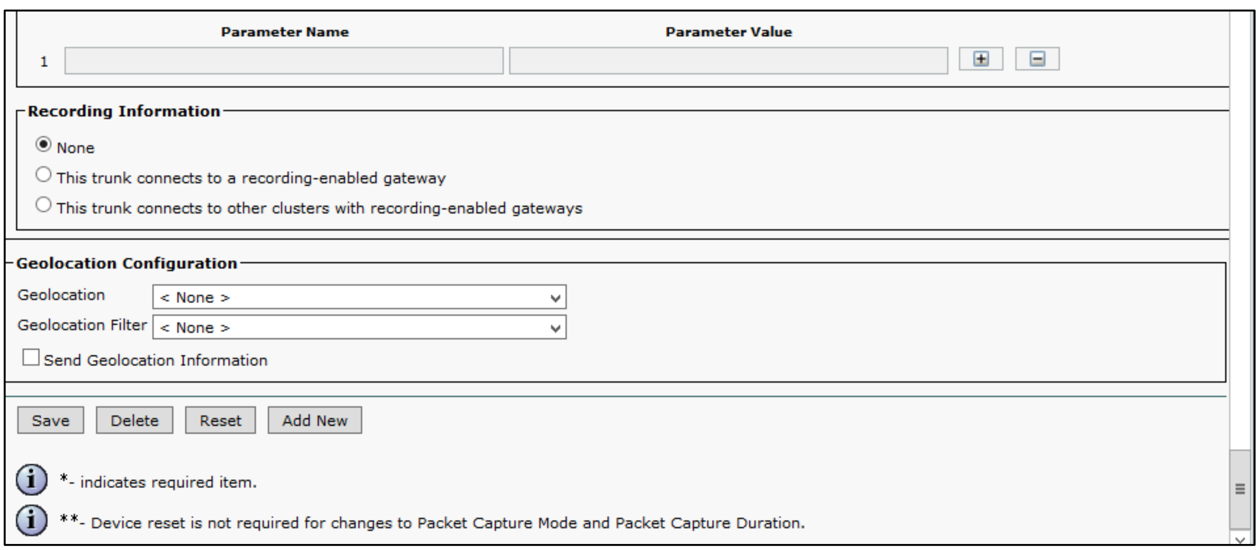
Set SIP Profile*= ALU – Standard SIP Profile

Set DTMF Signaling Method*= No Preference

All other values are default.




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



## Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration

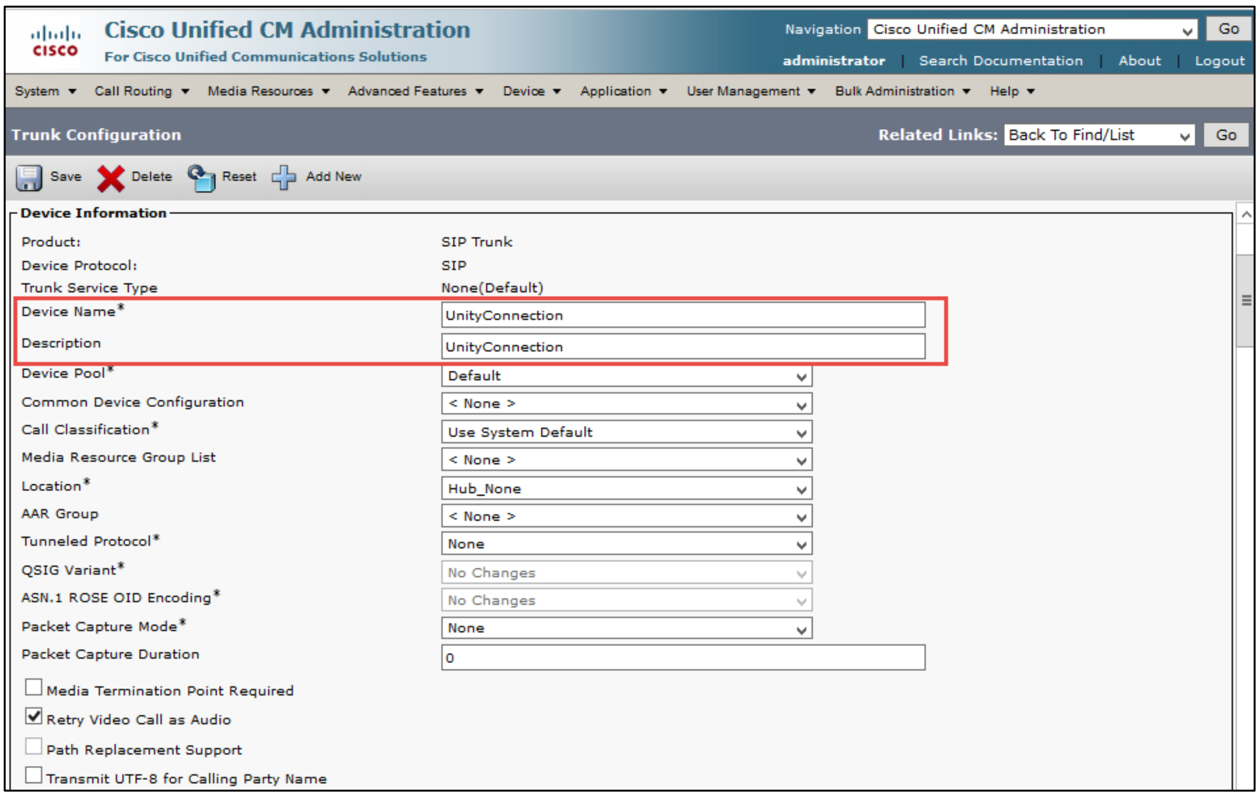
**Navigation:** Device → Trunk

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

Set Description = this text is used to identify this Trunk Group.

Set Device Pool* = Default

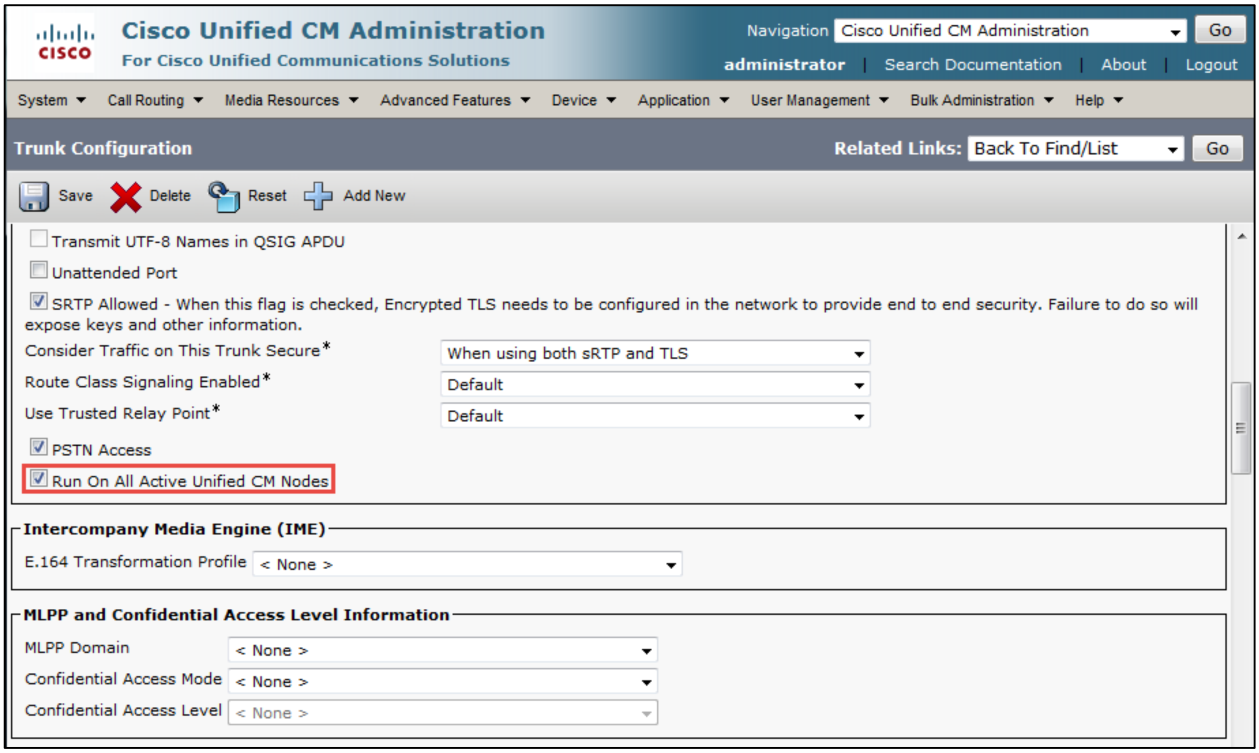
All other values are default.



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

Check Run On All Active Unified CM Nodes

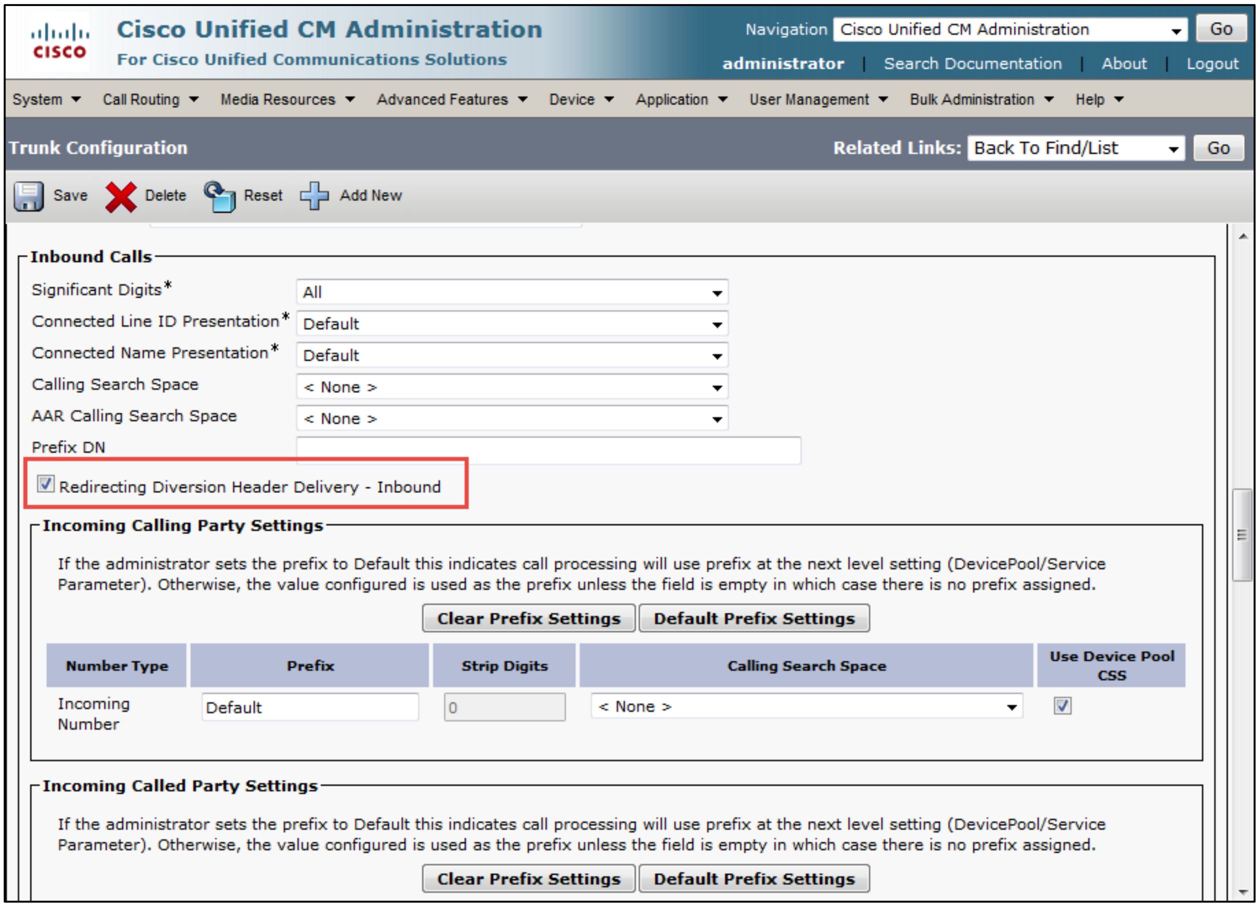
All other values are default.



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

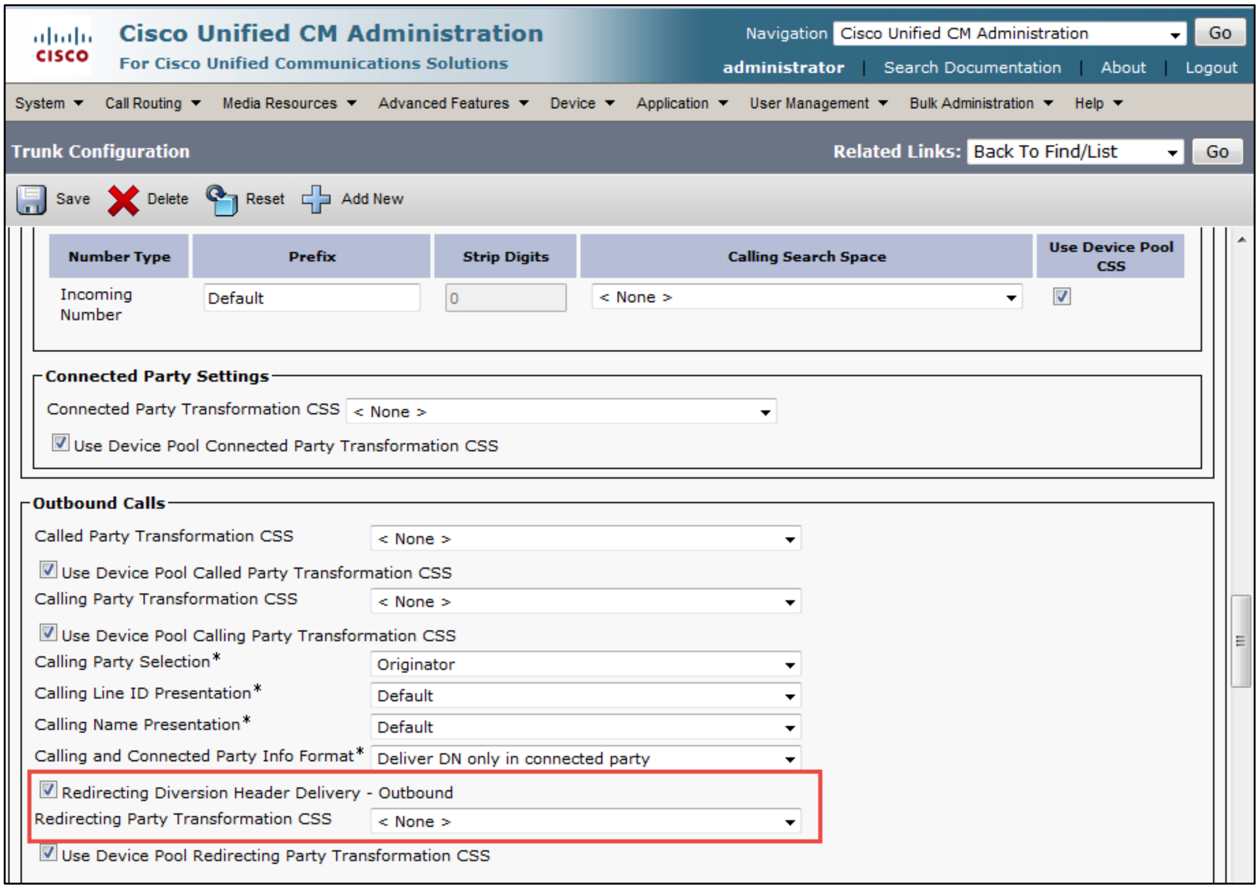
Check Redirecting Diversion Header Delivery - Inbound

All other values are default.




Check Redirecting Diversion Header Delivery – Outbound

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



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

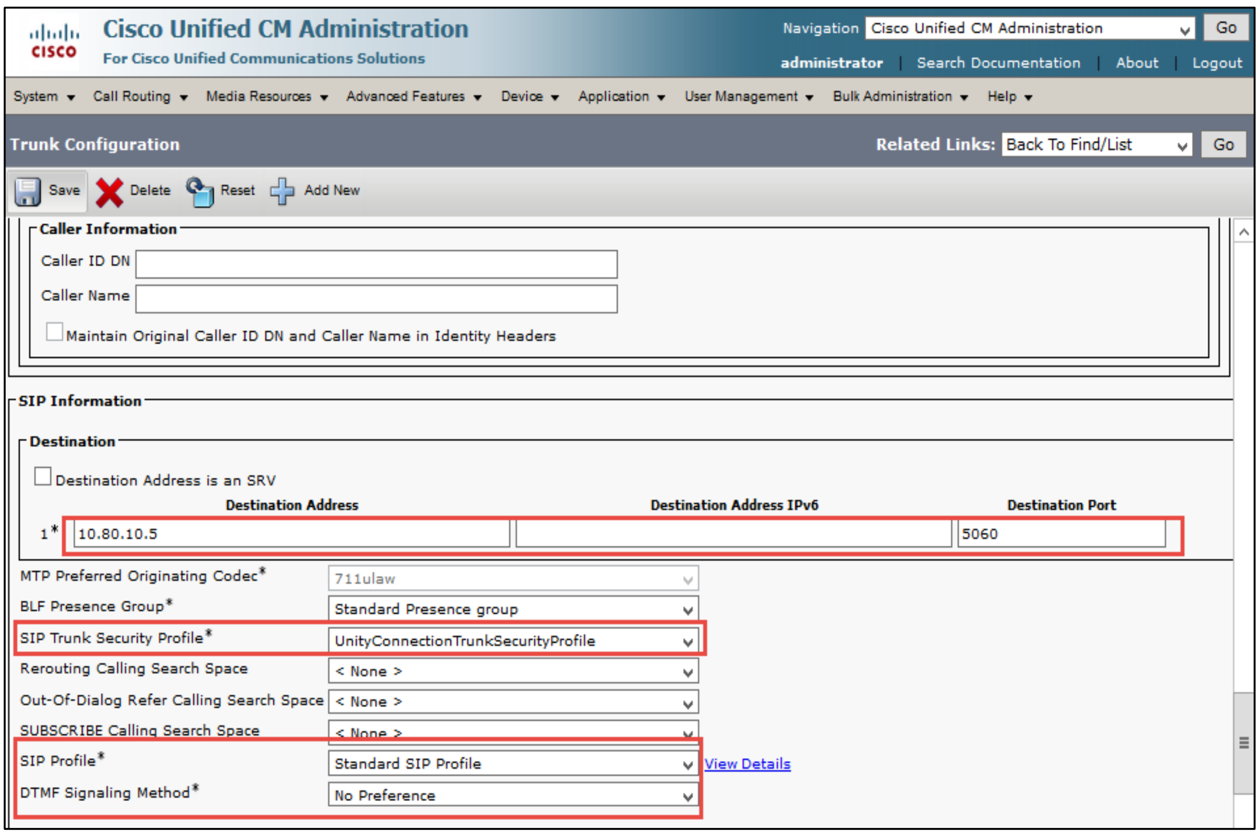
Set Destination Address = 10.80.10.5. This is used in this example.

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

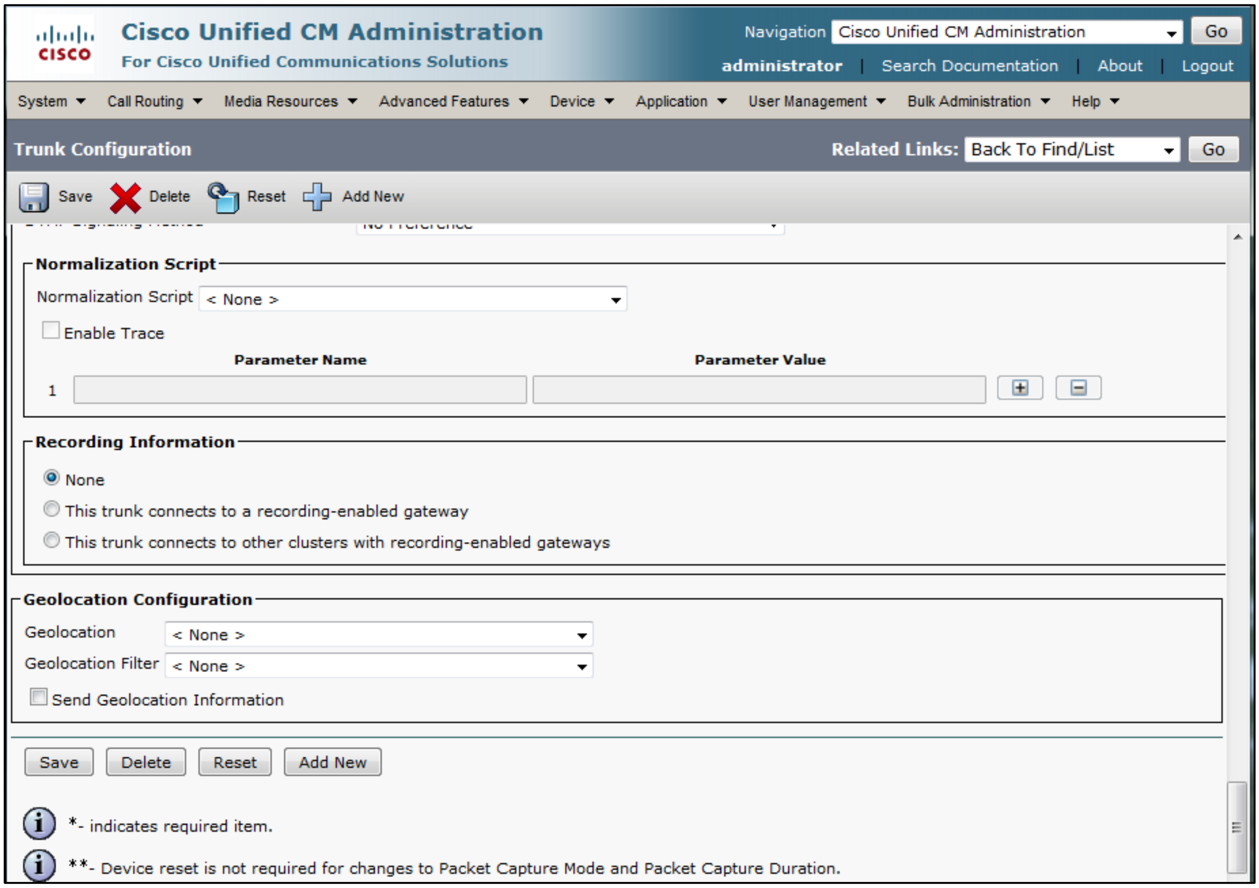
Set SIP Profile*= Standard SIP Profile

DTMF Signaling Method *= No Preference

All other values are default.



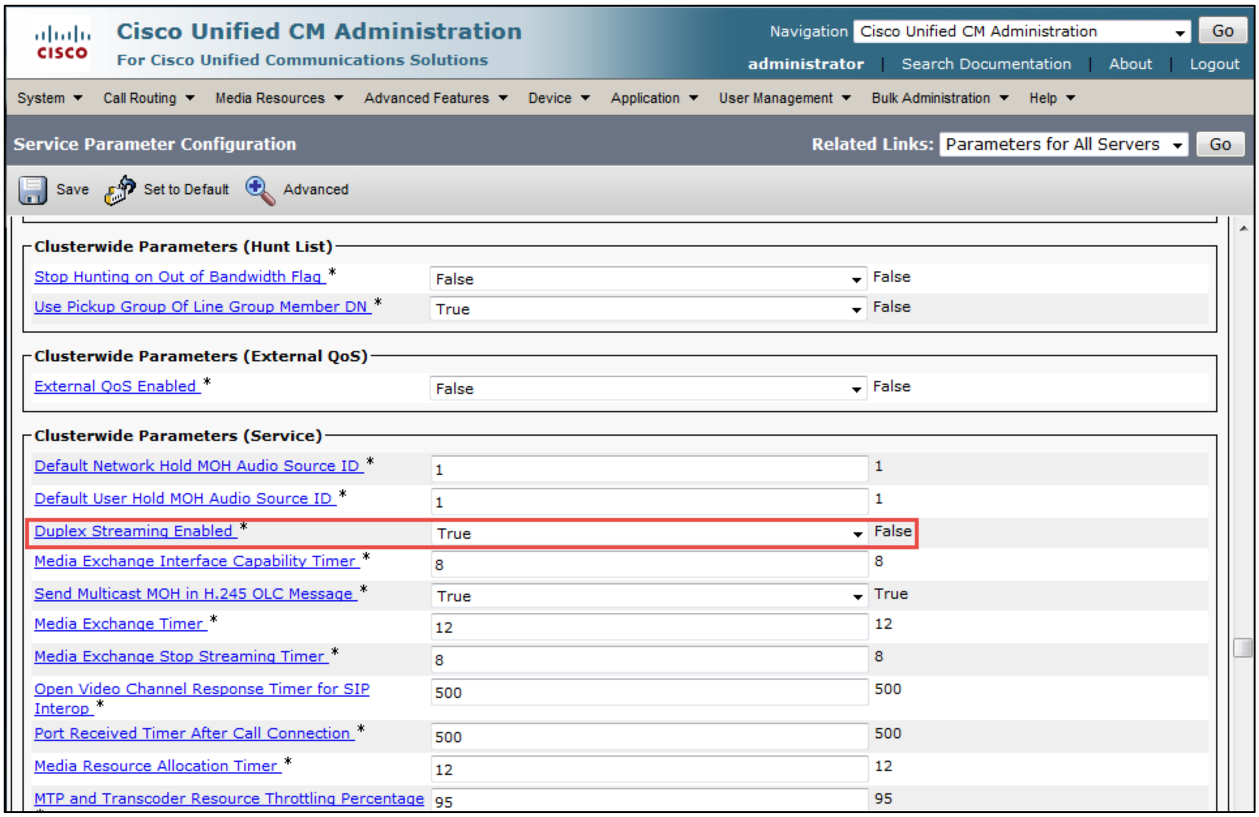
**Cisco Unified Communications Manager SIP Trunk to Cisco Unity Connection Configuration (Continued)**



## Cisco Unified Communications Manager Service Parameter

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

Set Duplex Streaming Enabled* = True.



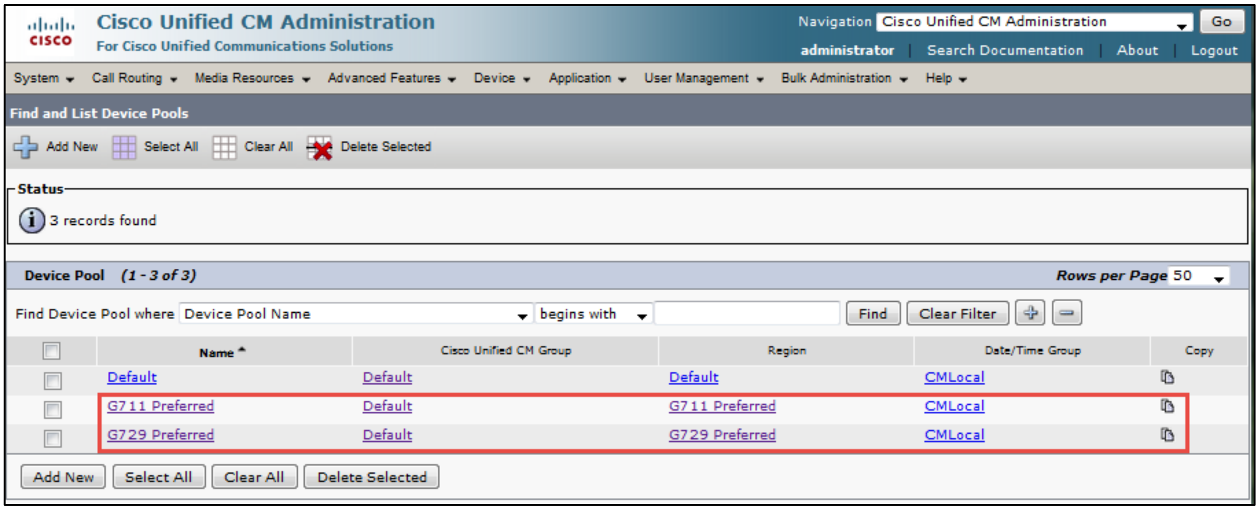

***Note:** Cisco Unified Communications Manager Service Parameter "Duplex Streaming Enabled" should be set to "True" in order for MoH and ring back to work properly during call transfers/conferences initiated by Cisco stations to Alcatel IP endpoints.

## Cisco Unified Communications Manager Device Pool Configuration

**Navigation Path:** System → Device Pool

G711 Preferred and G729 Preferred created in this example.

All other values are default.




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

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

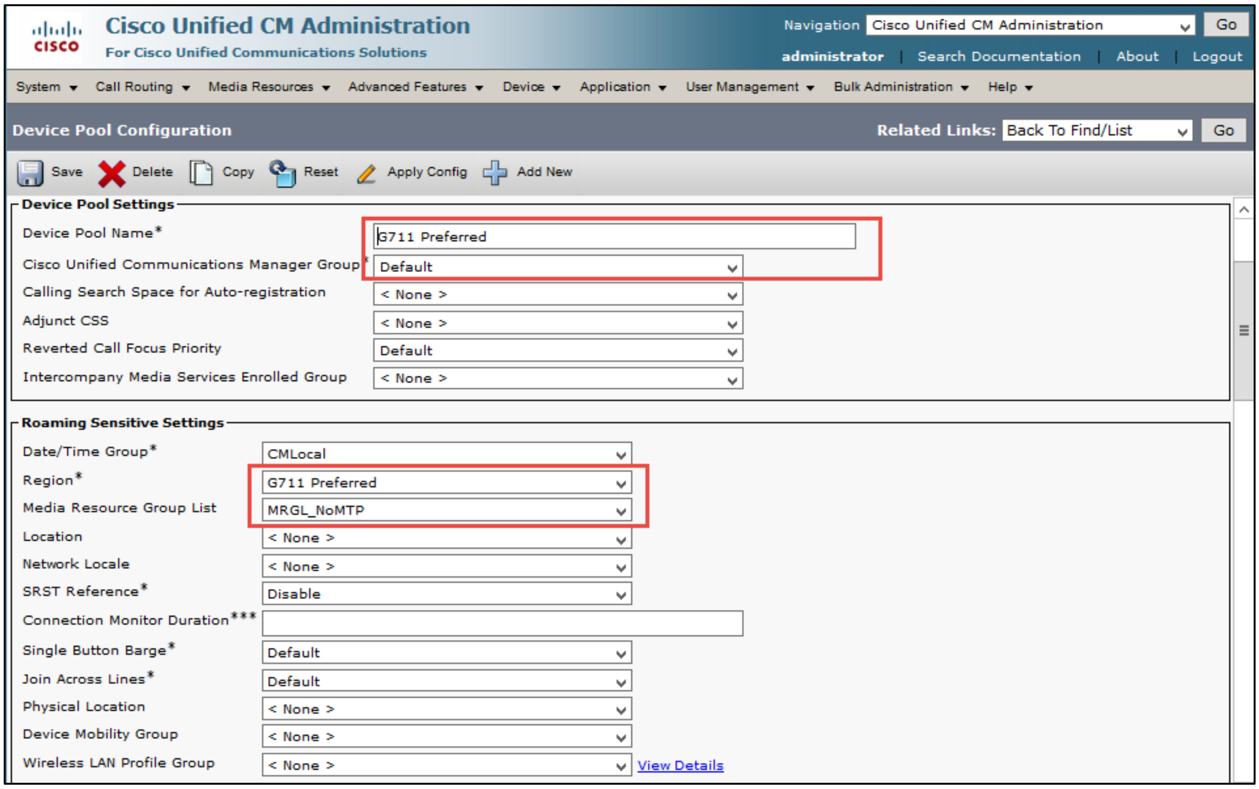
Set Cisco Unified Communications Manager Group*= Default

Set Date/Time Group* = CMLocal

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

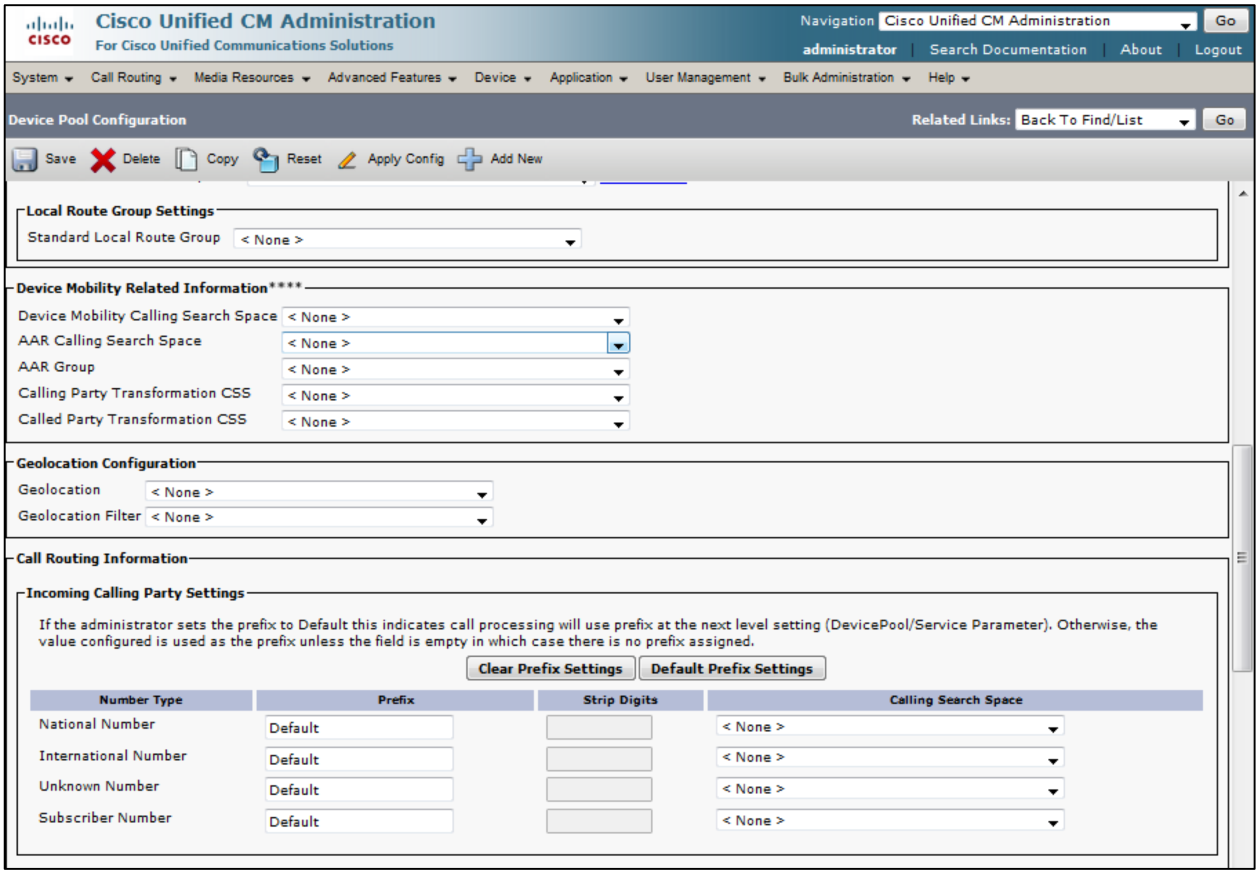
Set Media Resource Group List =MRGL_noMTP. This is used in this example.

All other values are default.



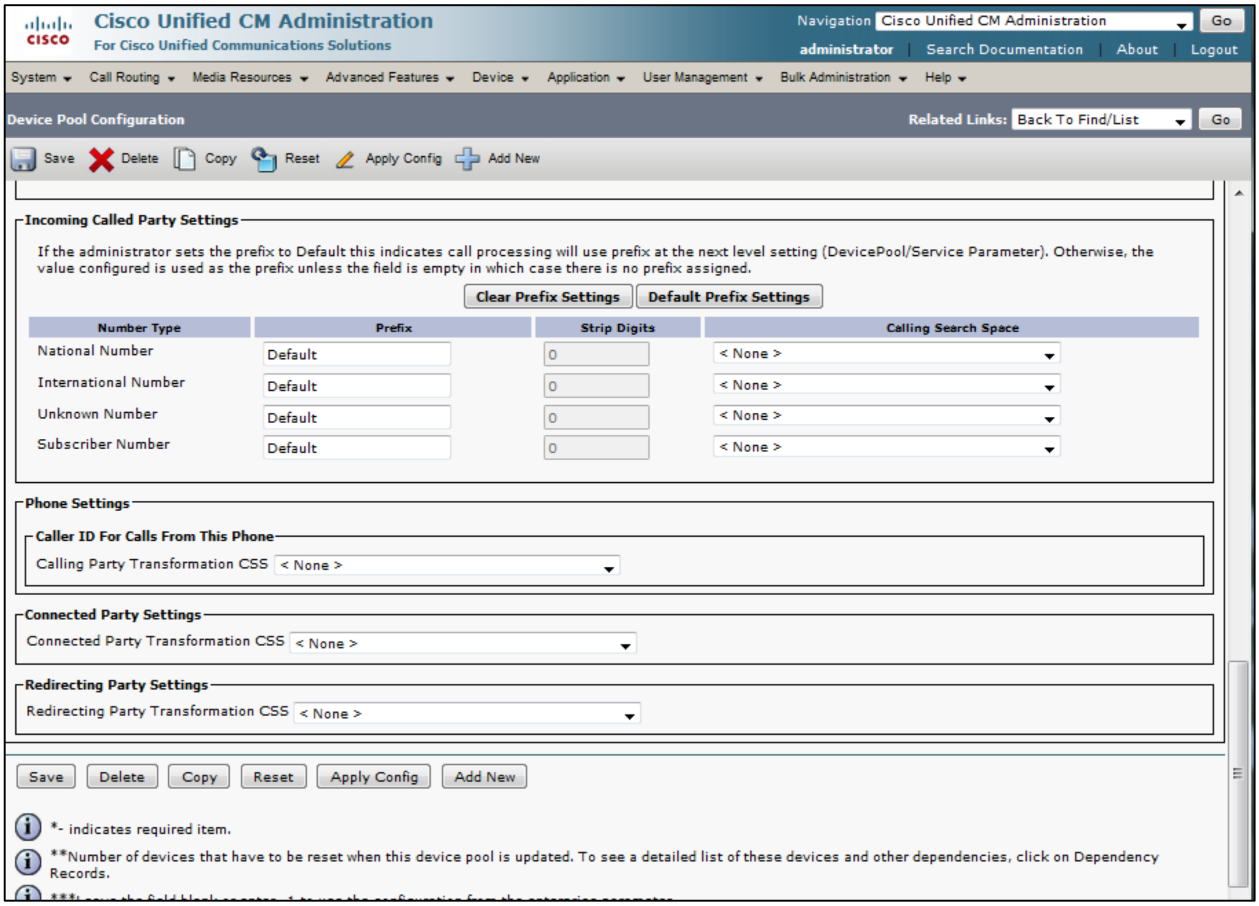

**Cisco Unified Communications Manager Device Pool Configuration (Continued)**

All values are default.




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

All values are default.




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

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

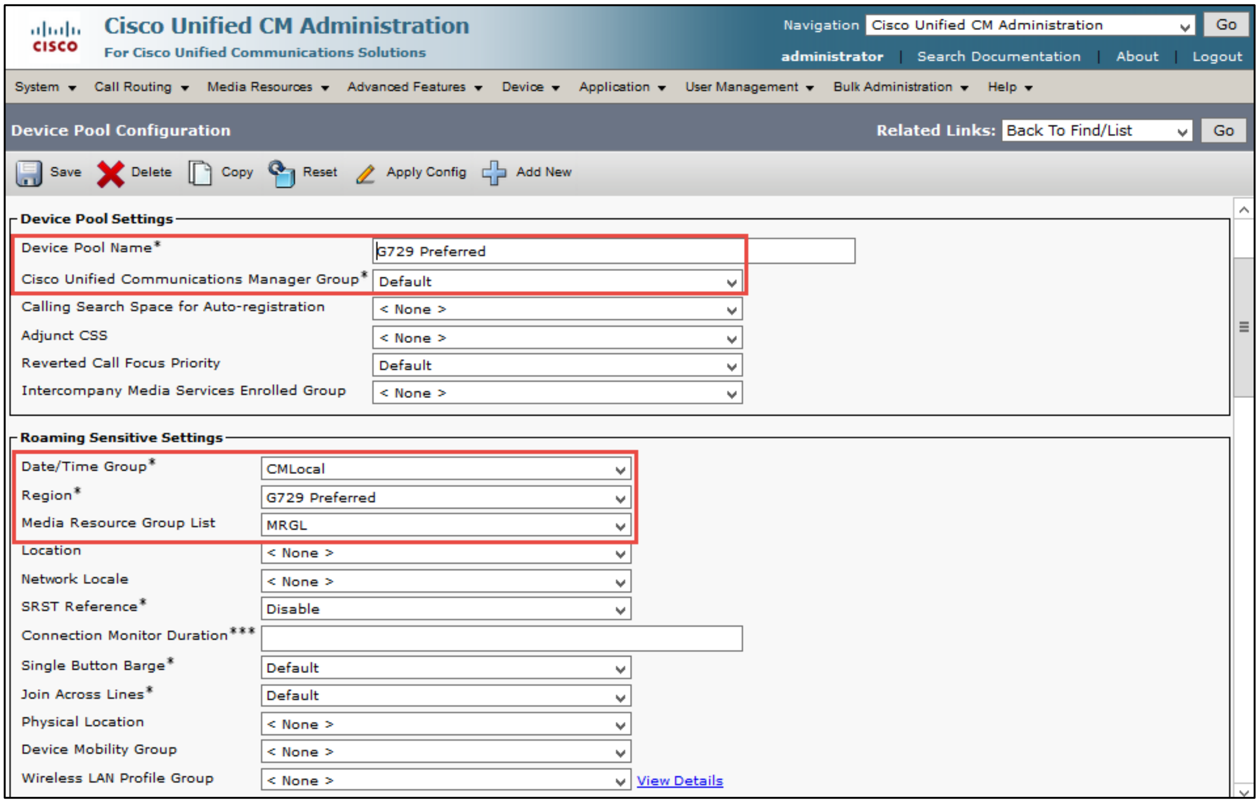
Set Cisco Unified Communications Manager Group*= Default

Set Date/Time Group* = CMLocal

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

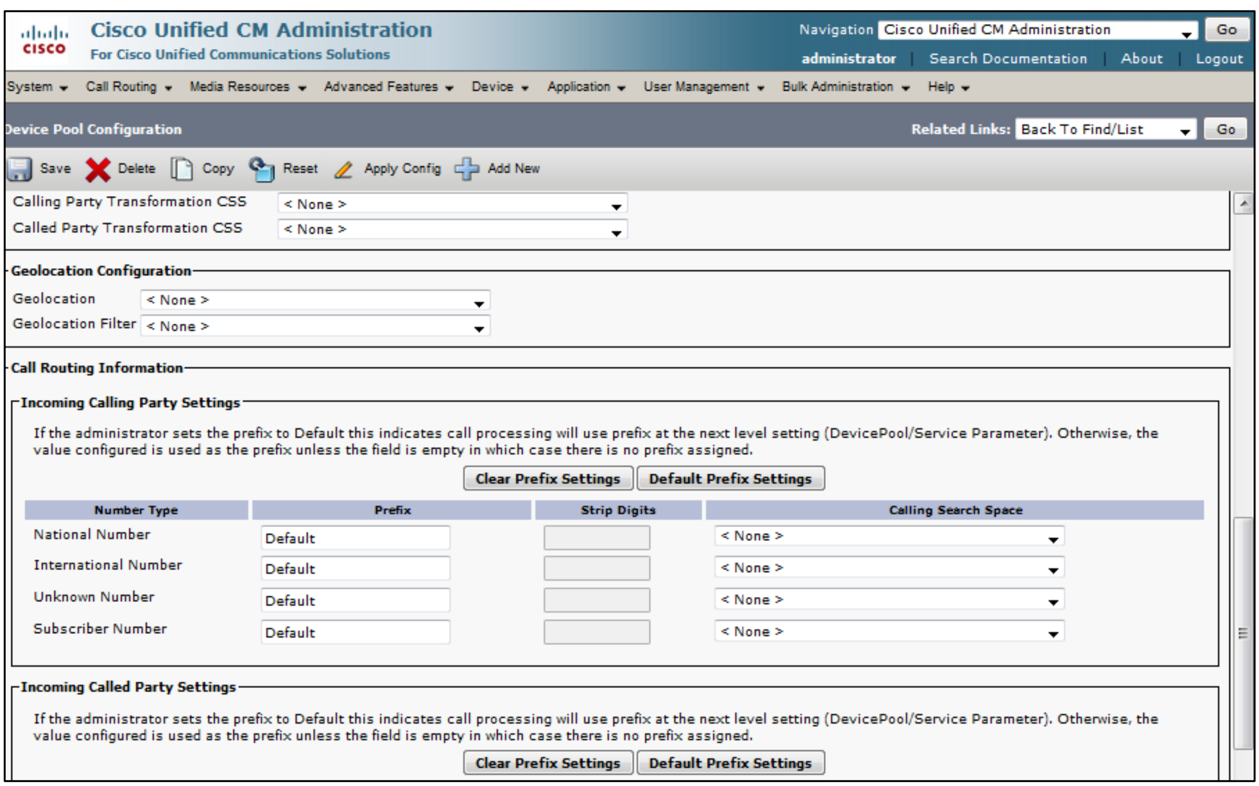
Set Media Resource Group List =MRGL. This is used in this example.

All other values are default



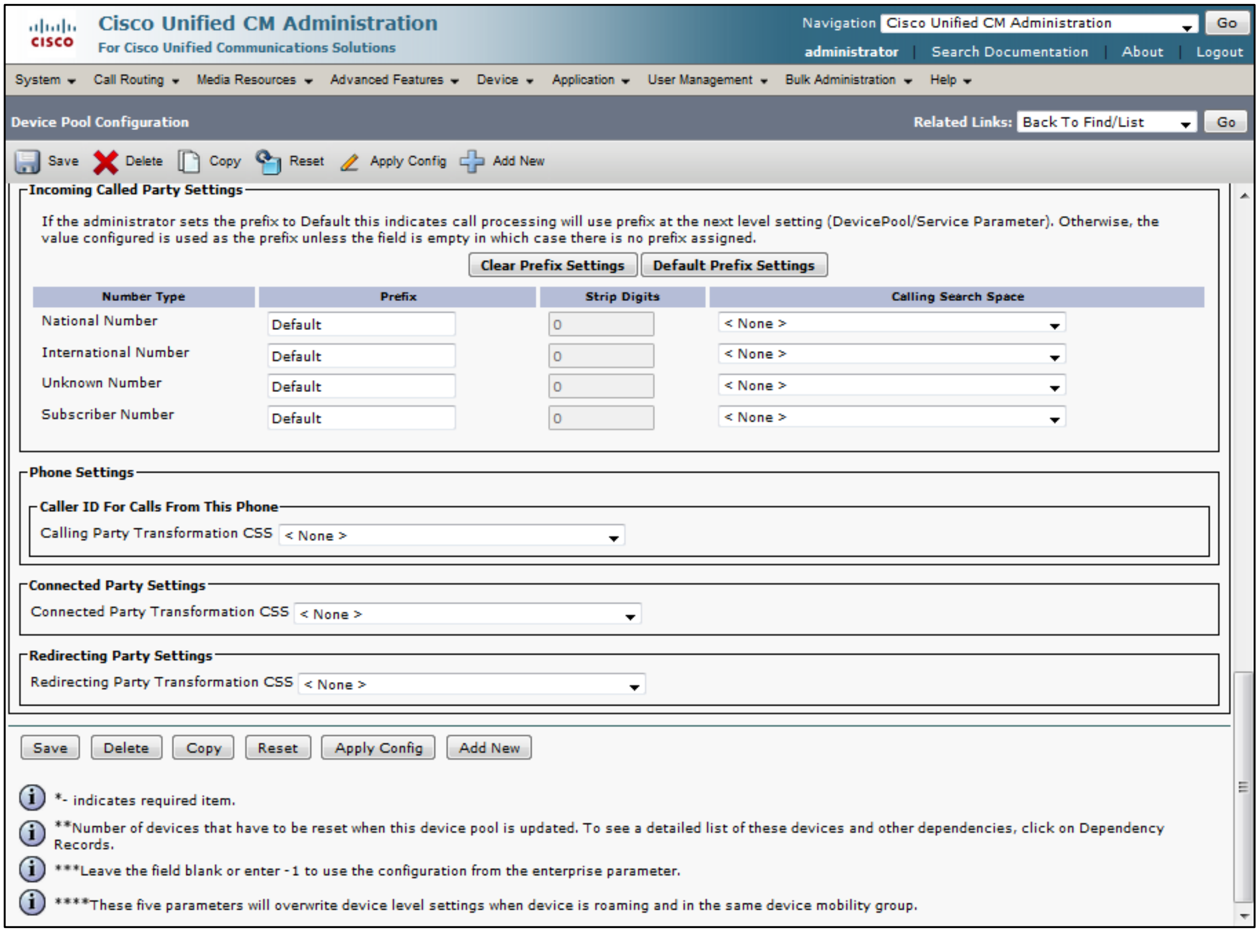

## Cisco Unified Communications Manager Device Pool Configuration (Continued)

All values are default.



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

All values are default.

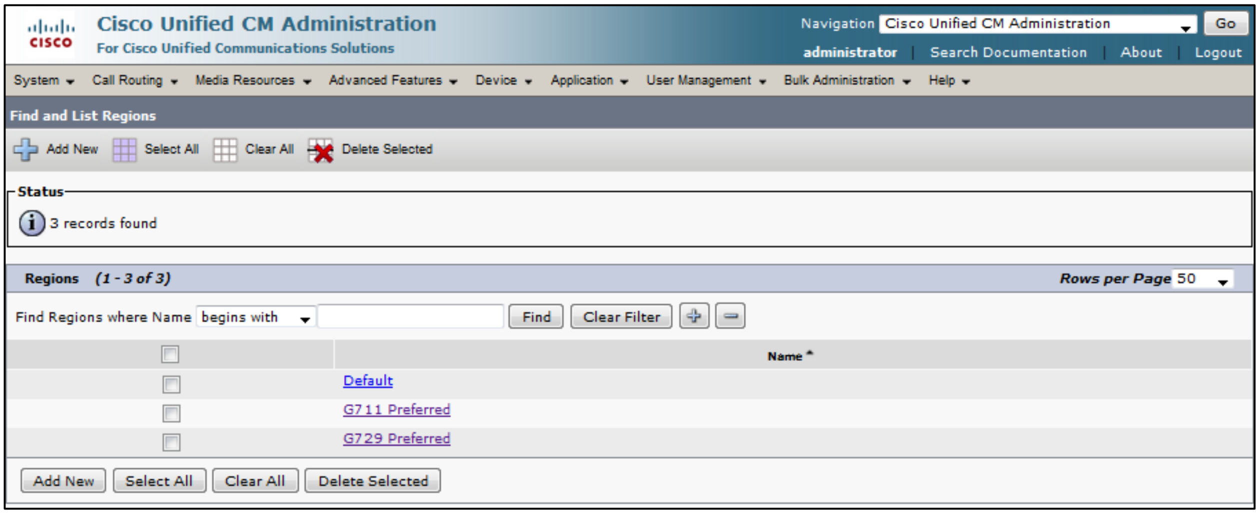



## Cisco Unified Communications Manager Region Configuration

**Navigation Path:** System → Region Information → Region

G711 Preferred and G729 Preferred created in this example.

All other values are default.




**Cisco Unified Communications Manager Region Configuration (Continued)**

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

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

Set Audio Codec Preference List= G711 Preferred

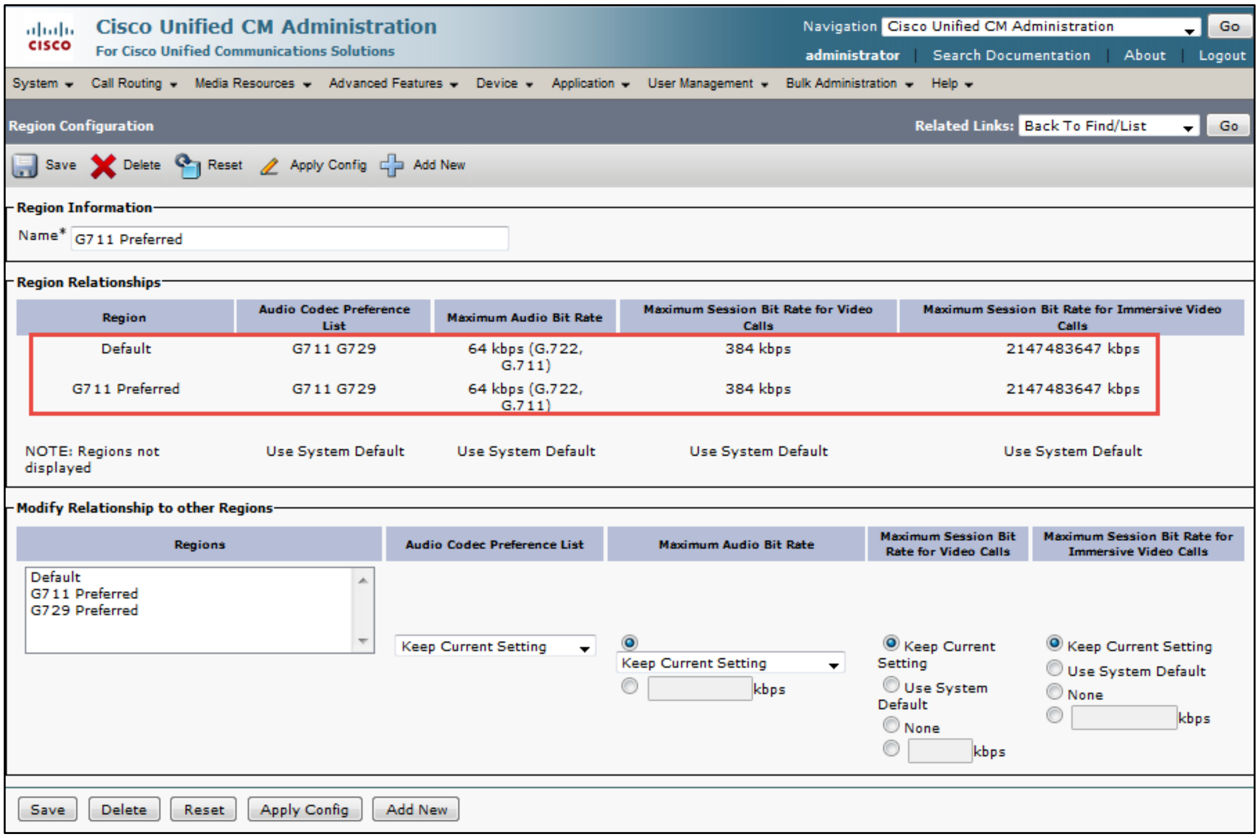
Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example

Set Region=Default. This is used in this example

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

Set Maximum Audio Bit Rate= 64 Kbps (G722, G7.11). This is used in this example

All other values are default




**Cisco Unified Communications Manager Region Configuration (Continued)**

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

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

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

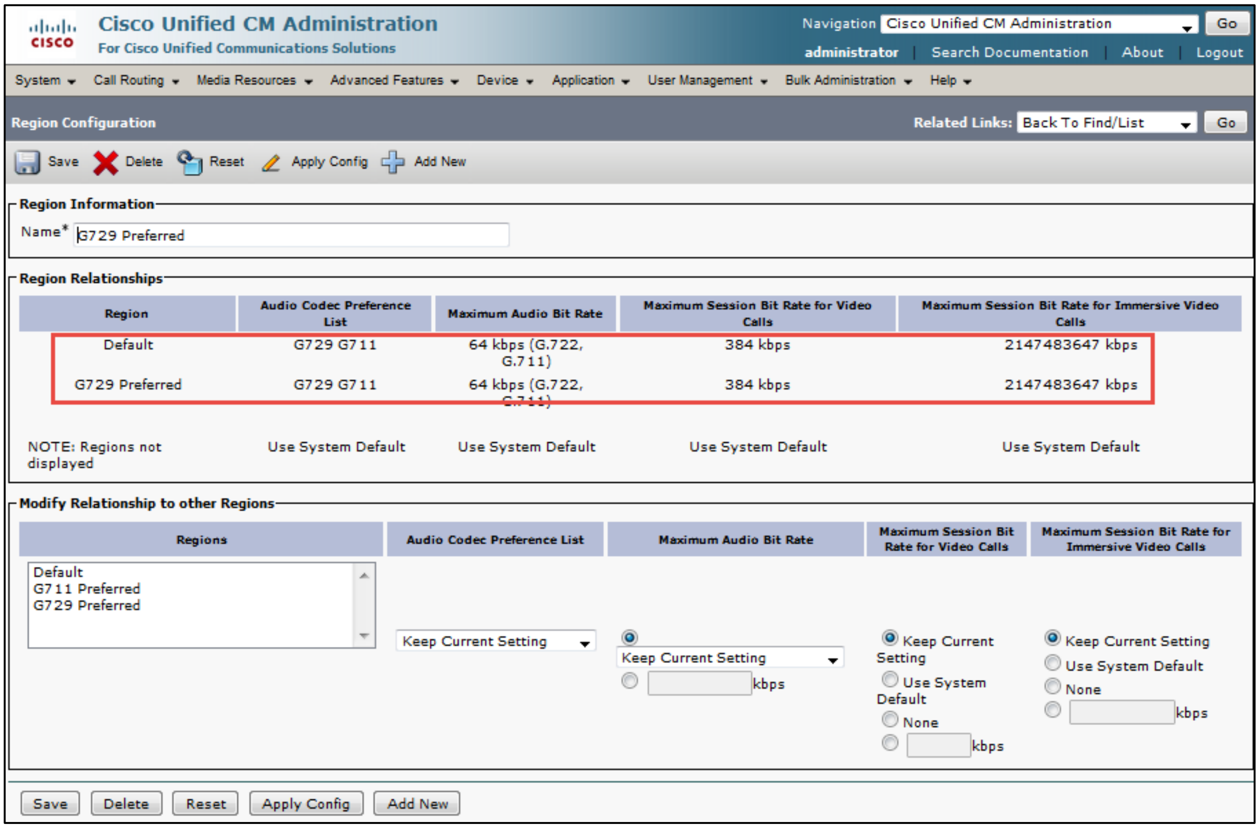
Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example.

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

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

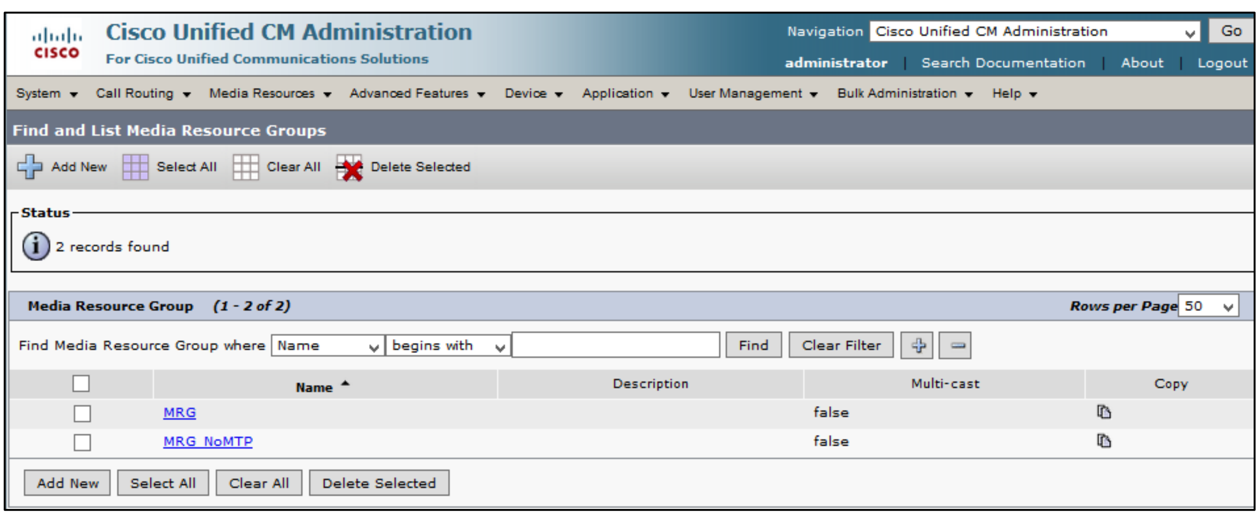
Set Maximum Audio Bit Rate= 64 Kbps (G7.22, G7.11). This is used in this example

All other values are default.




## Cisco Unified Communications Manager Media Resource Group

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



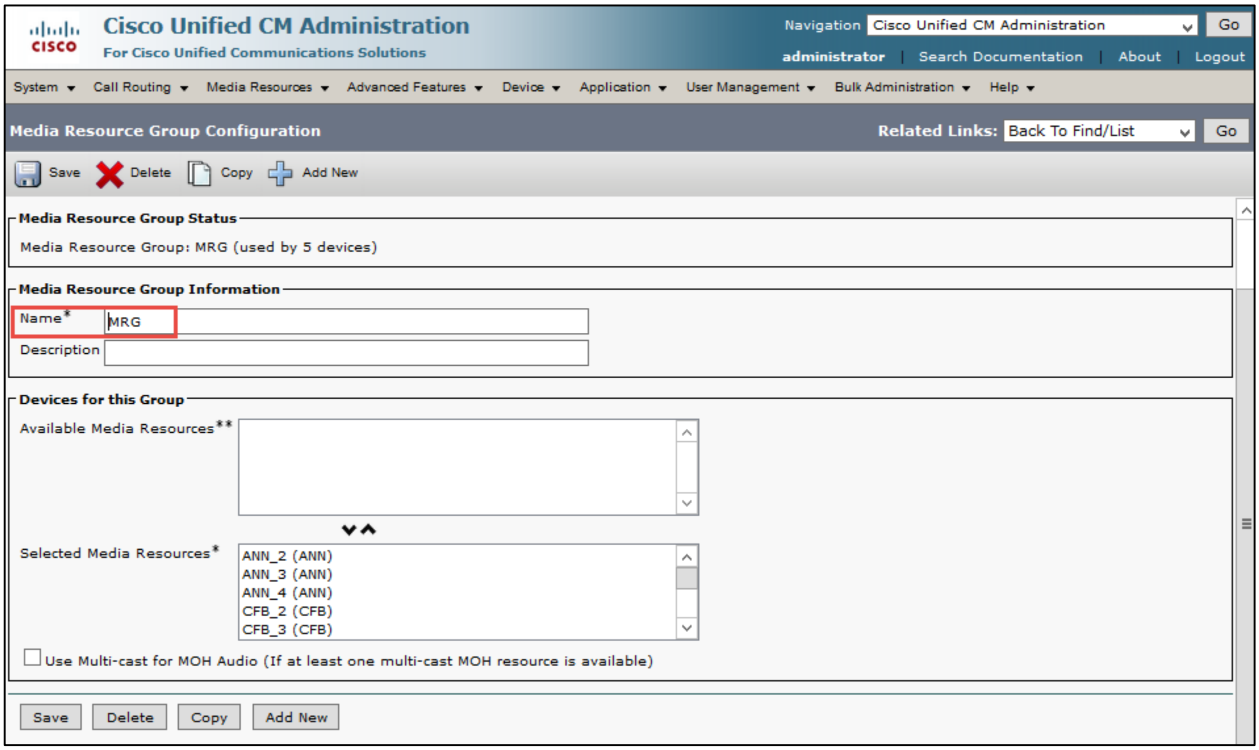
### Media Resource Group MRG

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

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

Set all resources in the Selected Media Resources* Box.

All other values are default.




### Resource Group for MRG_NoMTP

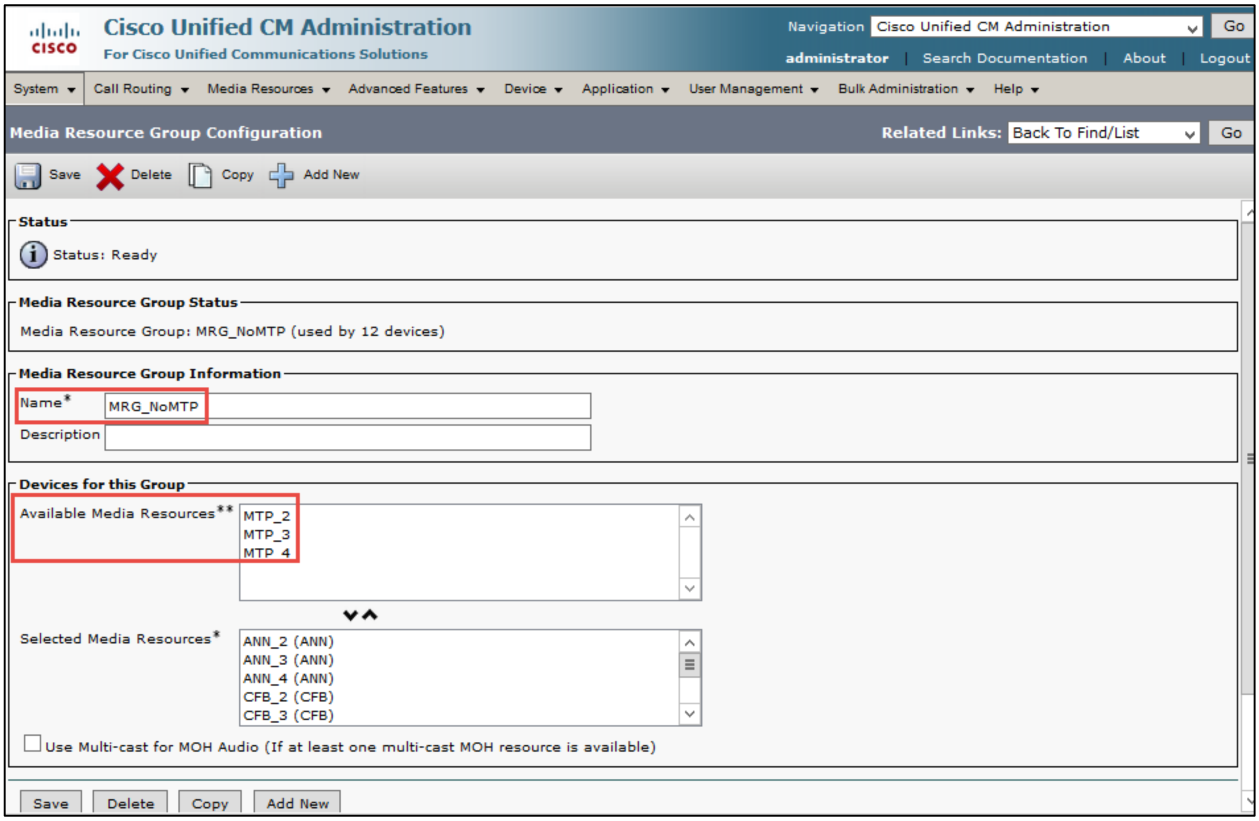
Set Name*= MRG_NoMTP. This is used for this example.

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

Set Available Media Resources = MTP_2, MTP_3 and MTP_4

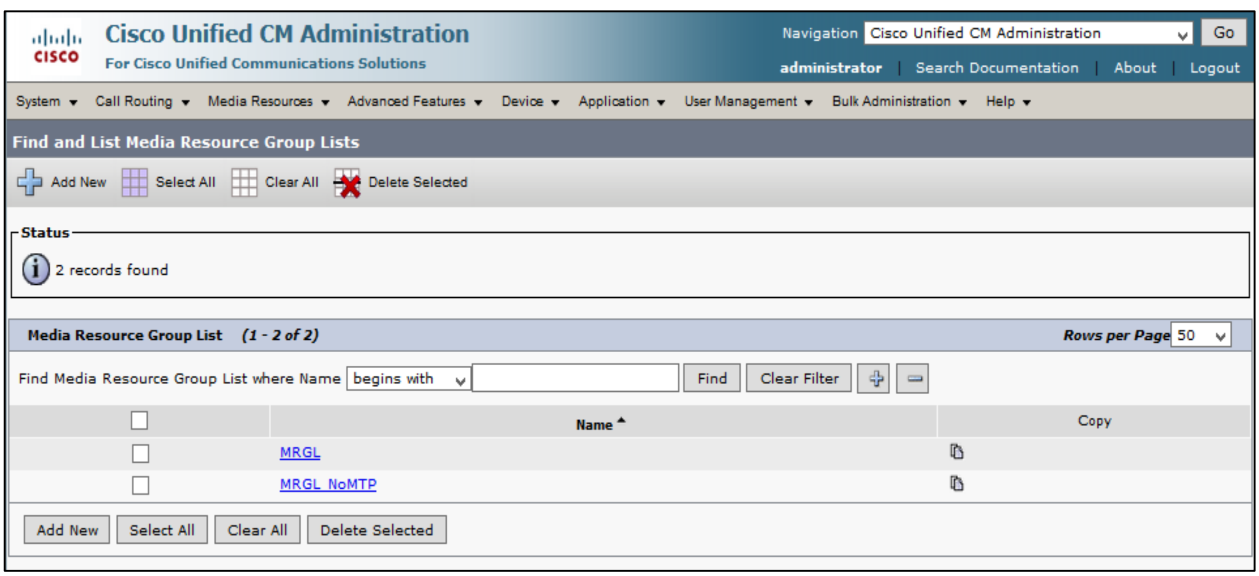
Set other resources in the Selected Media Resources*

All other values are default.




## Cisco Unified Communications Manager Media Resource Group List

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

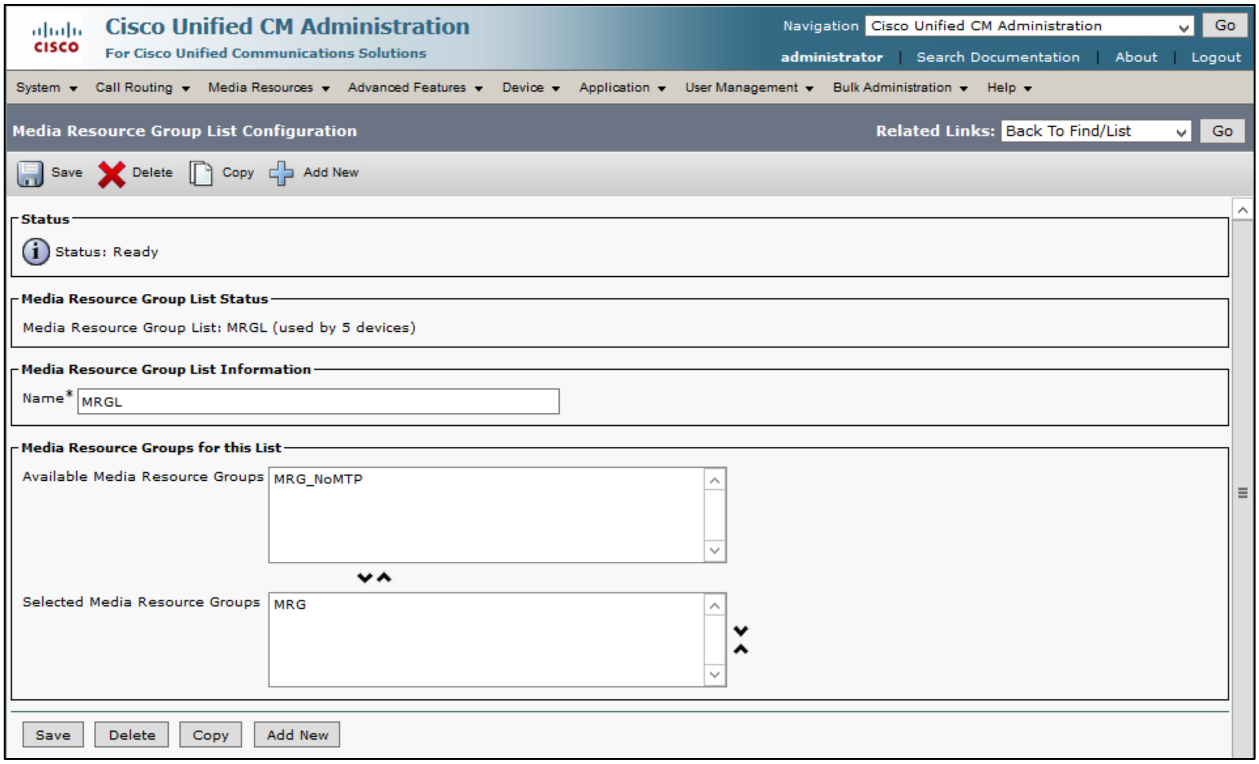


Set Name*= MRGL. This is used for this example.

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

Set Available Media Resources = MRGL

Set Selected Media Resource Groups= MRG

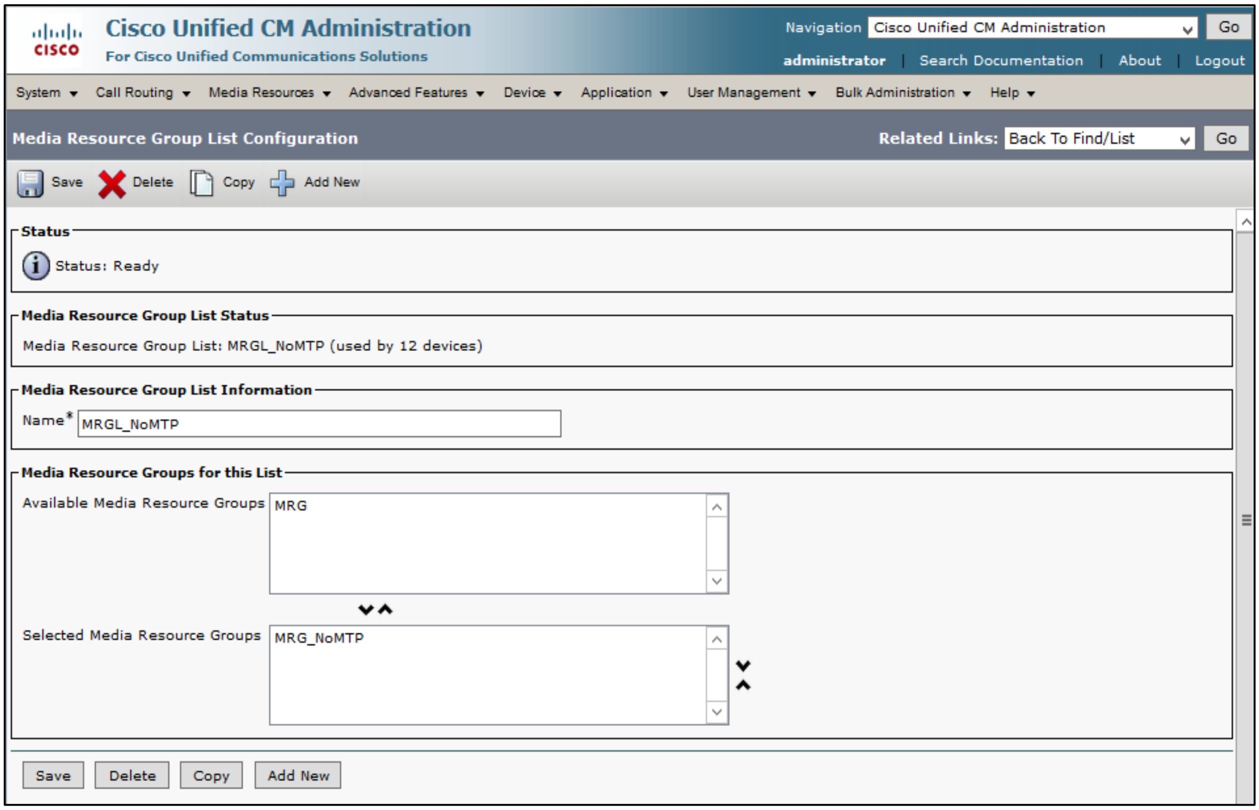



Set Name*= MRGL_noMTP. This is used for this example

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

Set Available Media Resources MRG

Set Selected Media Resource Groups= MRGL_NoMTP




Note: MRG_NoMTP Media resource group was used to test the scenarios that did not require MTP.

## Cisco Unified Communications Manager Route Pattern to Alcatel

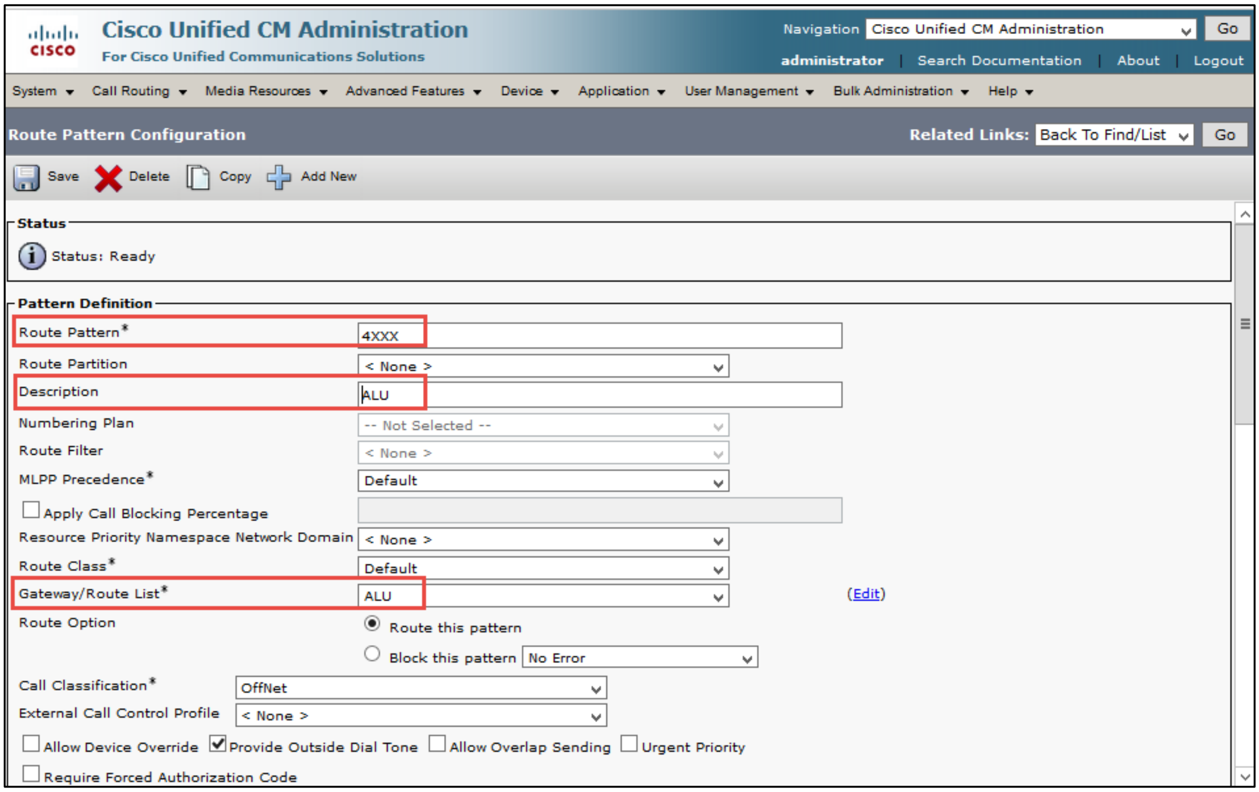
Set Route Pattern* =4XXX. This is used to route to the Alcatel PBX in this example

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

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

Uncheck Provide Outside Dial Tone

All other values are default.




### Route Pattern Configuration for 4xxx (Continued)

Set Calling Party Transform Mask = XXXX

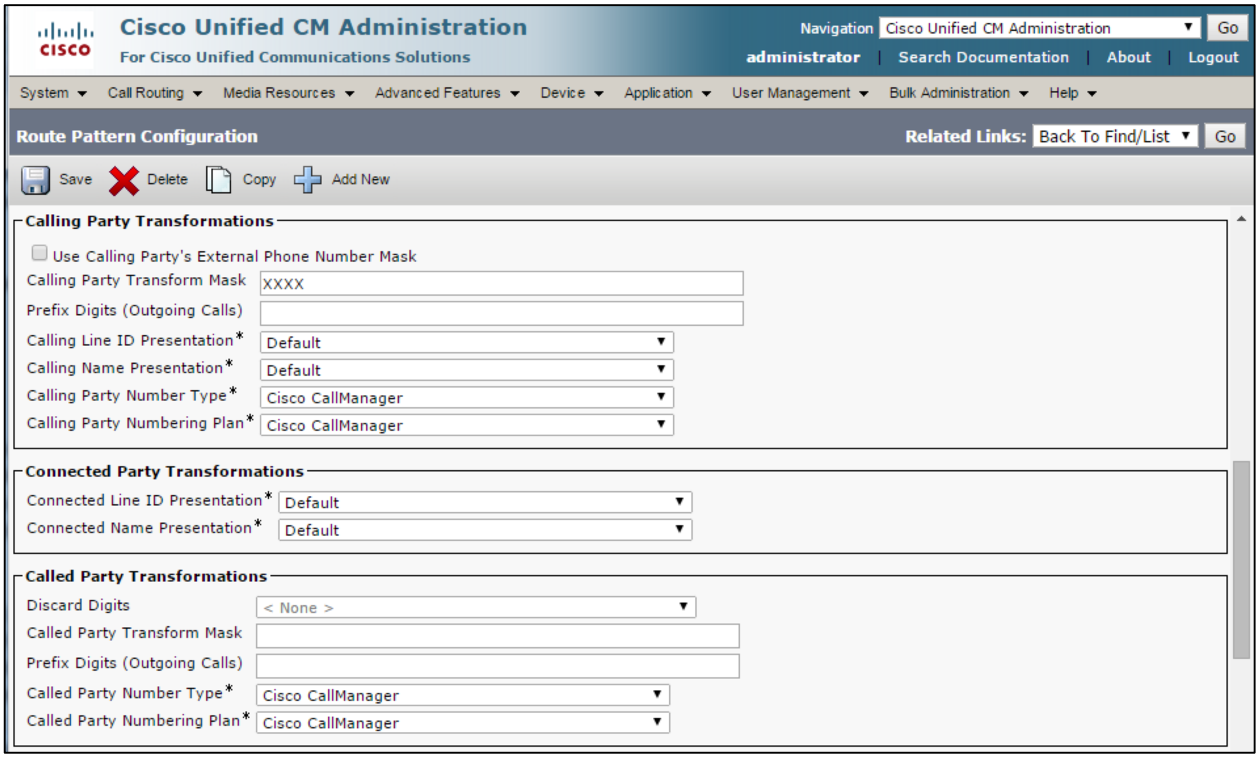
Set Calling Line ID Presentation= Allowed

Set Calling Name Presentation= Allowed

Set Connected Line ID Presentation*= Default

Set Calling Name Presentation* = Default

All other values are default.




## Cisco Unified Communications Manager SIP Phone Device Level Configuration

**Navigation Path:** Device→ Phone

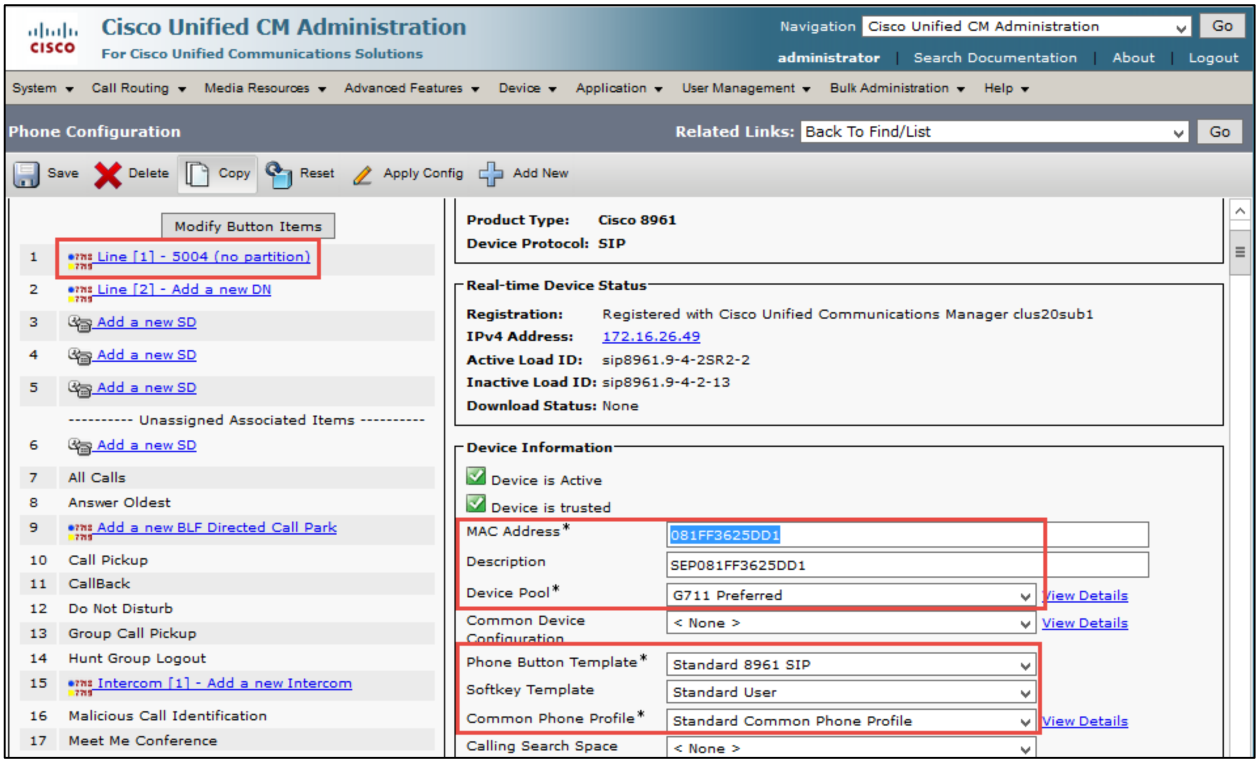
Set MAC Address* = 081FF3625Dxx. This is used in this example

Set Description = this text is used to identify this Phone

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

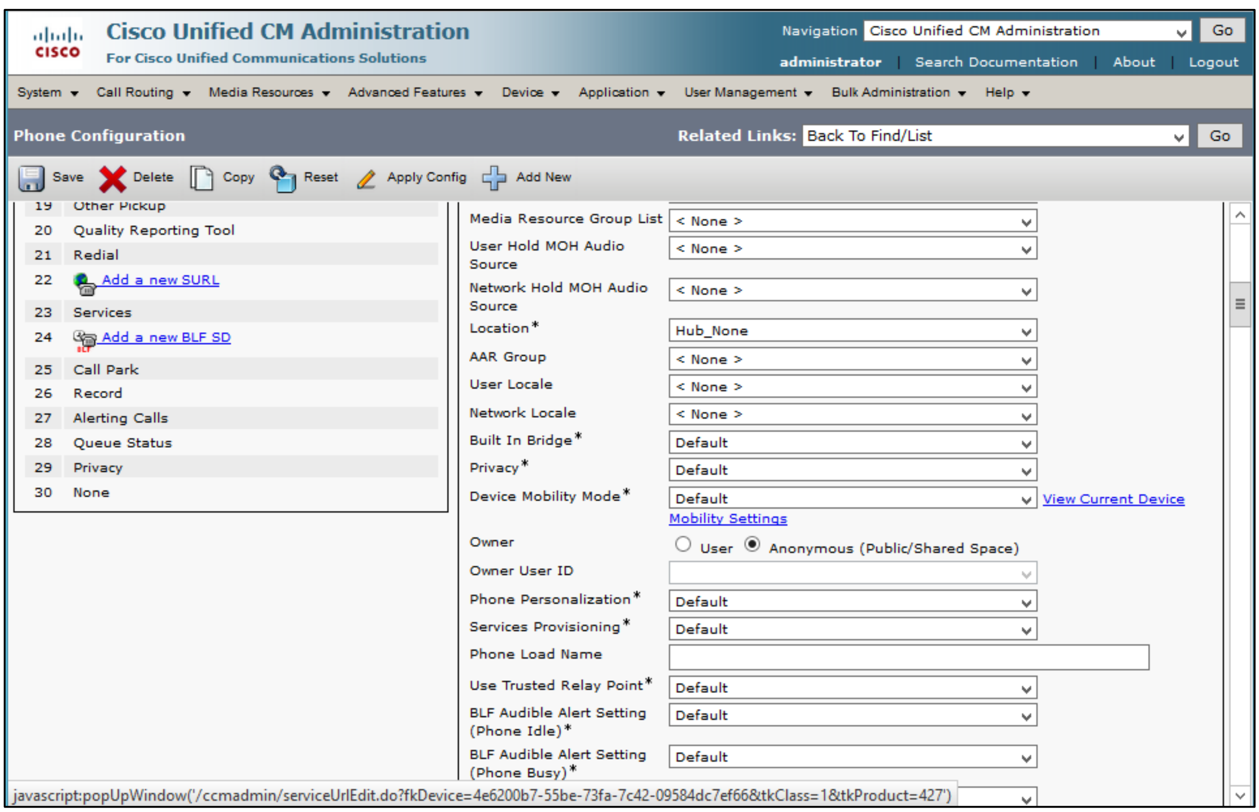
Set Phone Button Template*= Standard 8961 SIP. This is used in this example

Common Phone Profile *= Standard Common Phone Profile

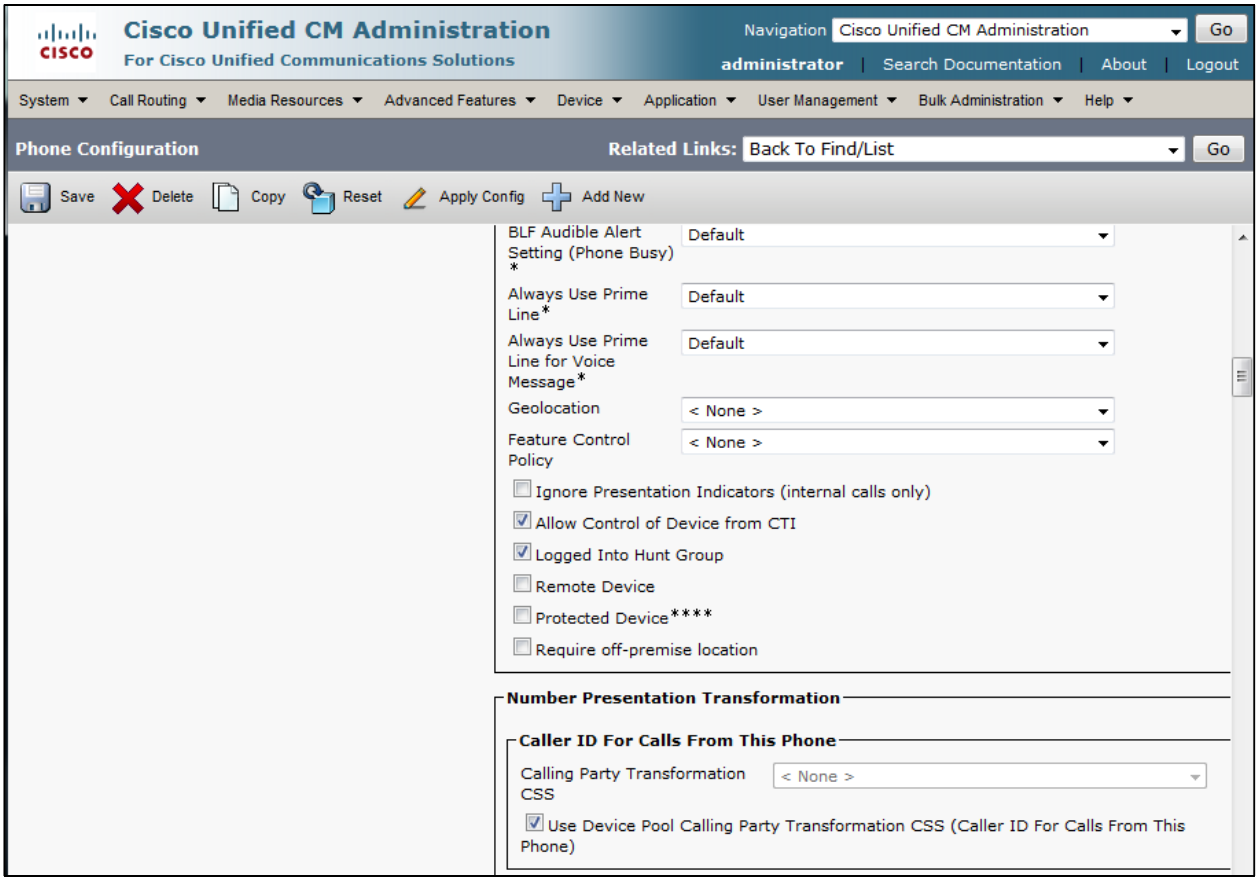



**Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)**

Set Media Resource Group List = None. This is used in this example.All other values are default.



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

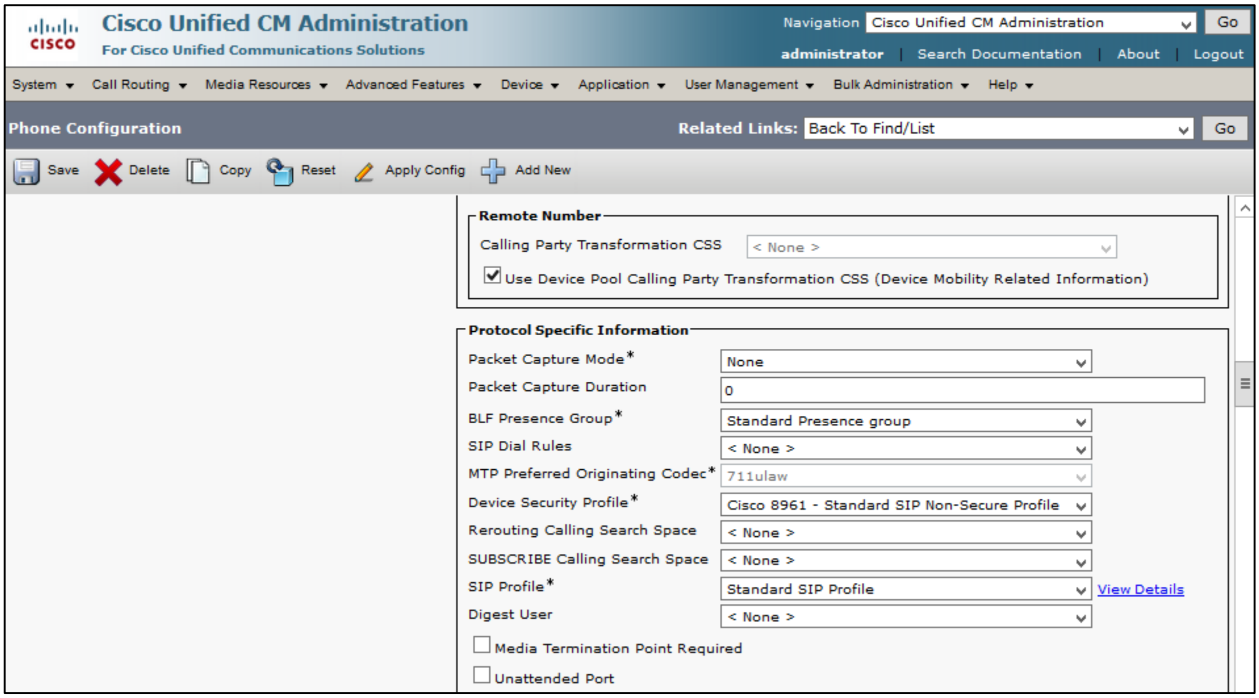



**Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)**

Set Device Security Profile* = Cisco 8961- Standard SIP Non-Secure Profile. This is used in this example.

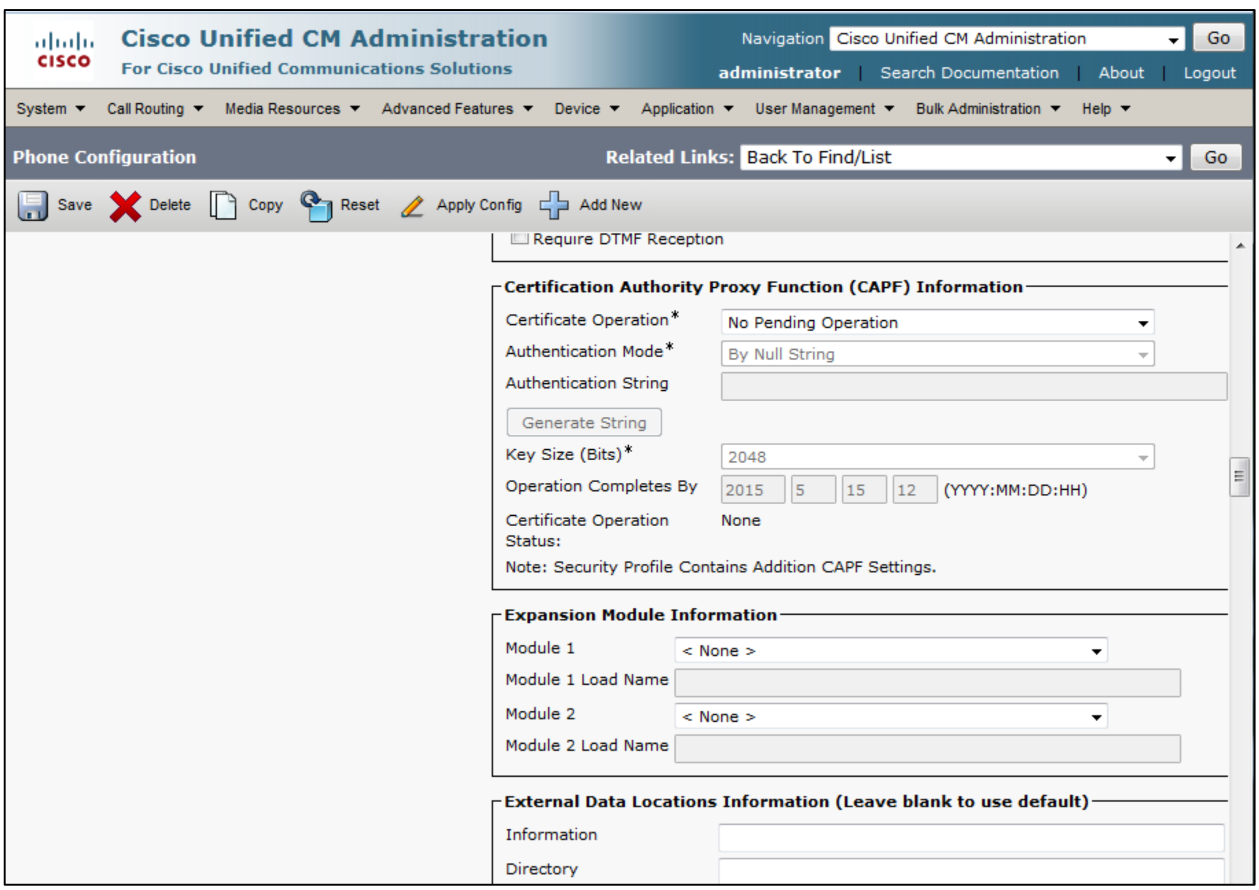
Set SIP Profile*= Standard SIP Profile. This is used in this example.

All other values are default.



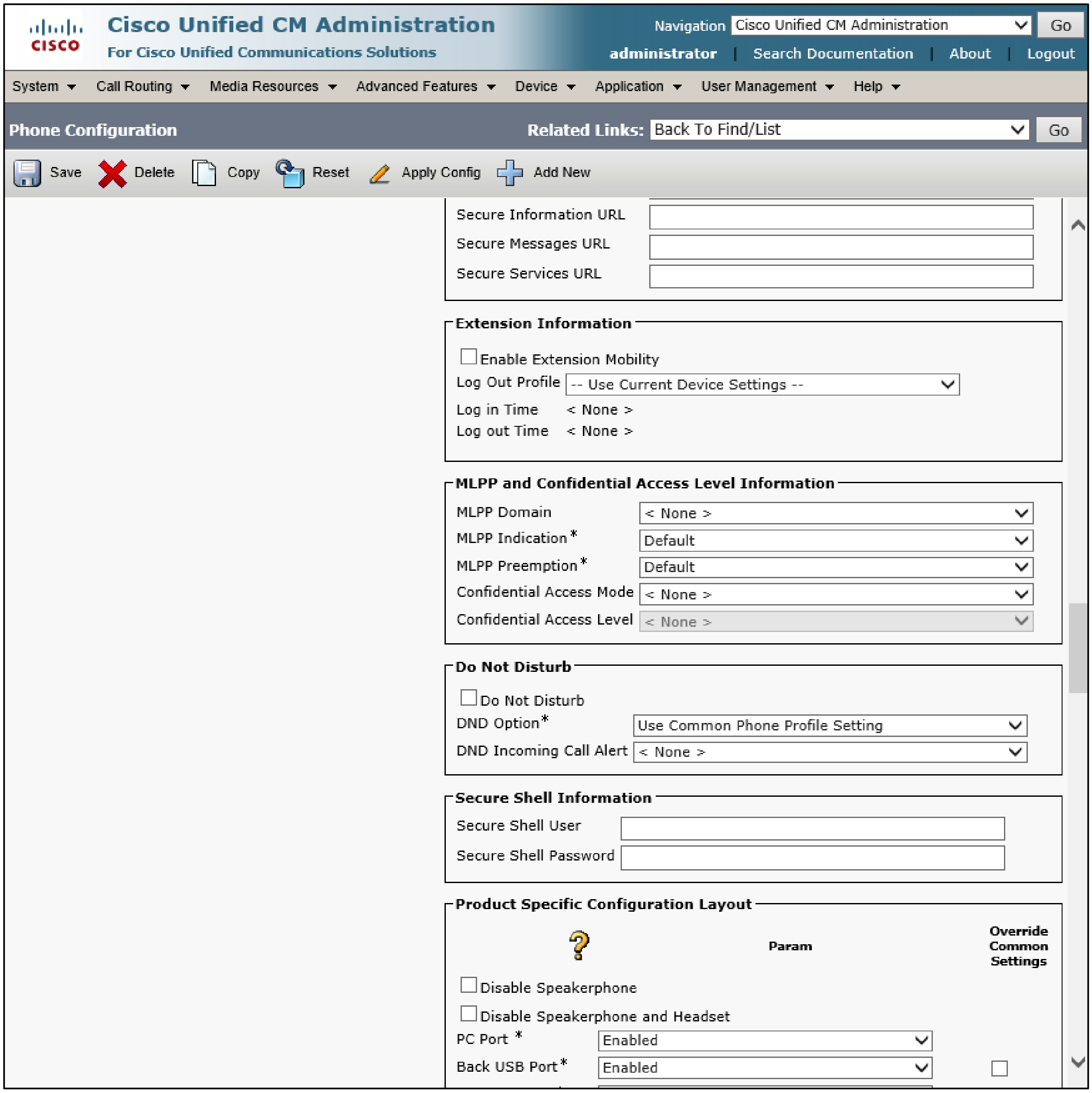

## Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All other values are default.



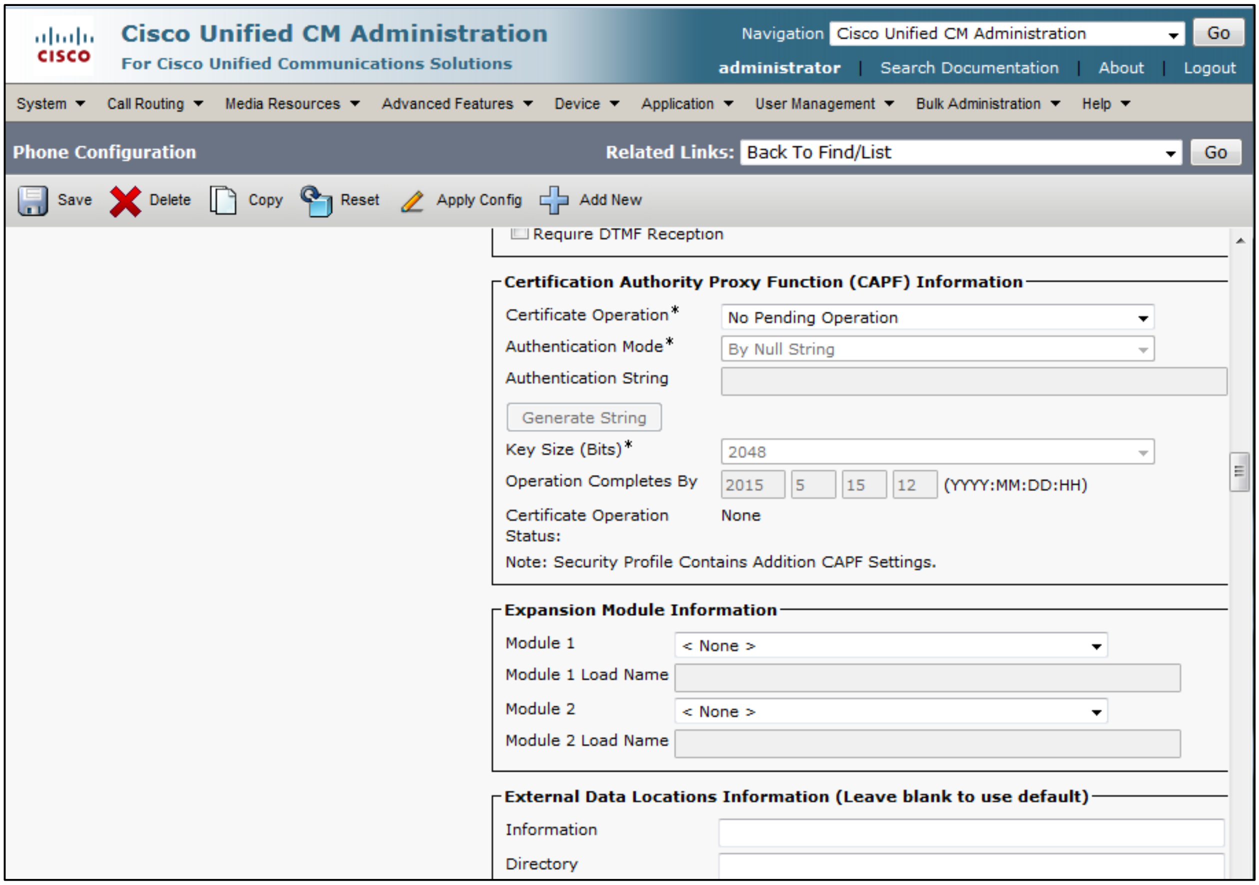
### Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

All values are default.



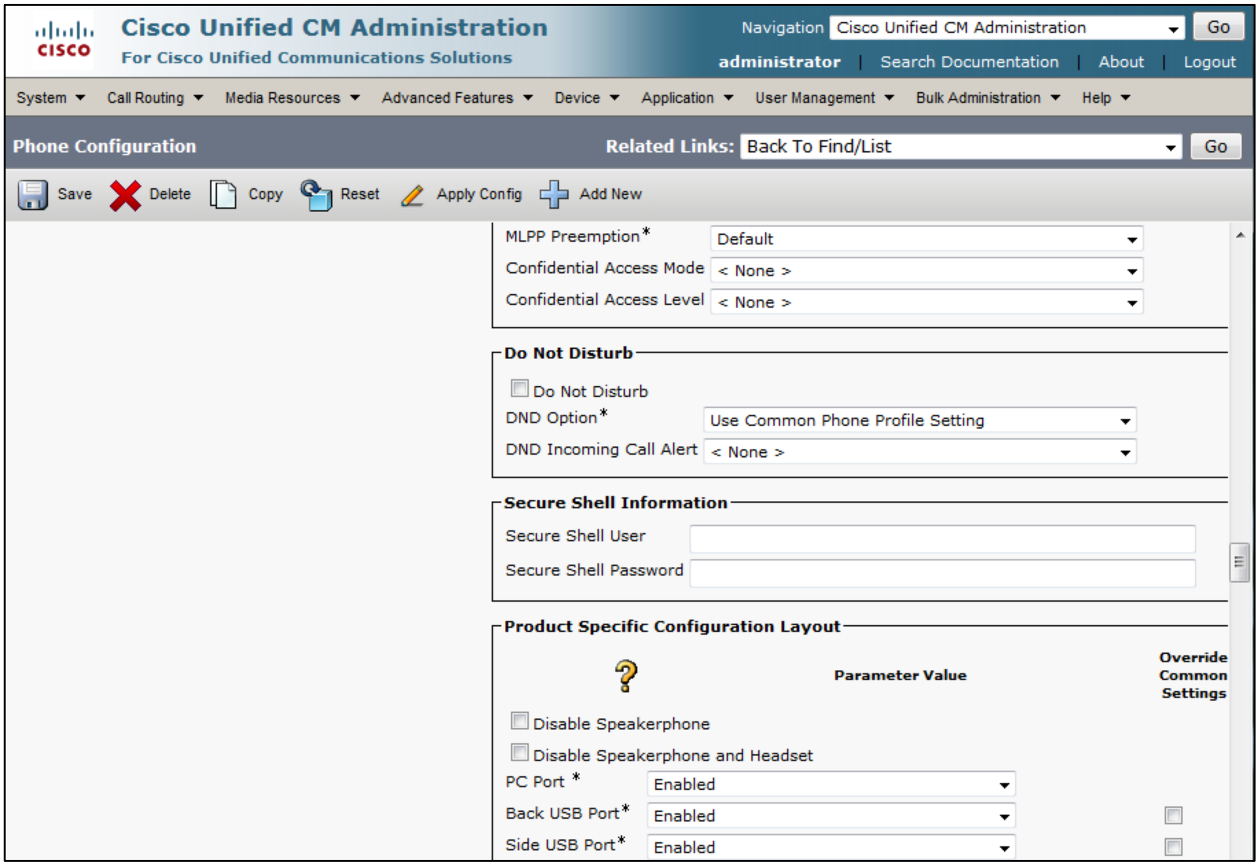
**Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)**

All other values are default.



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

All other values are default.

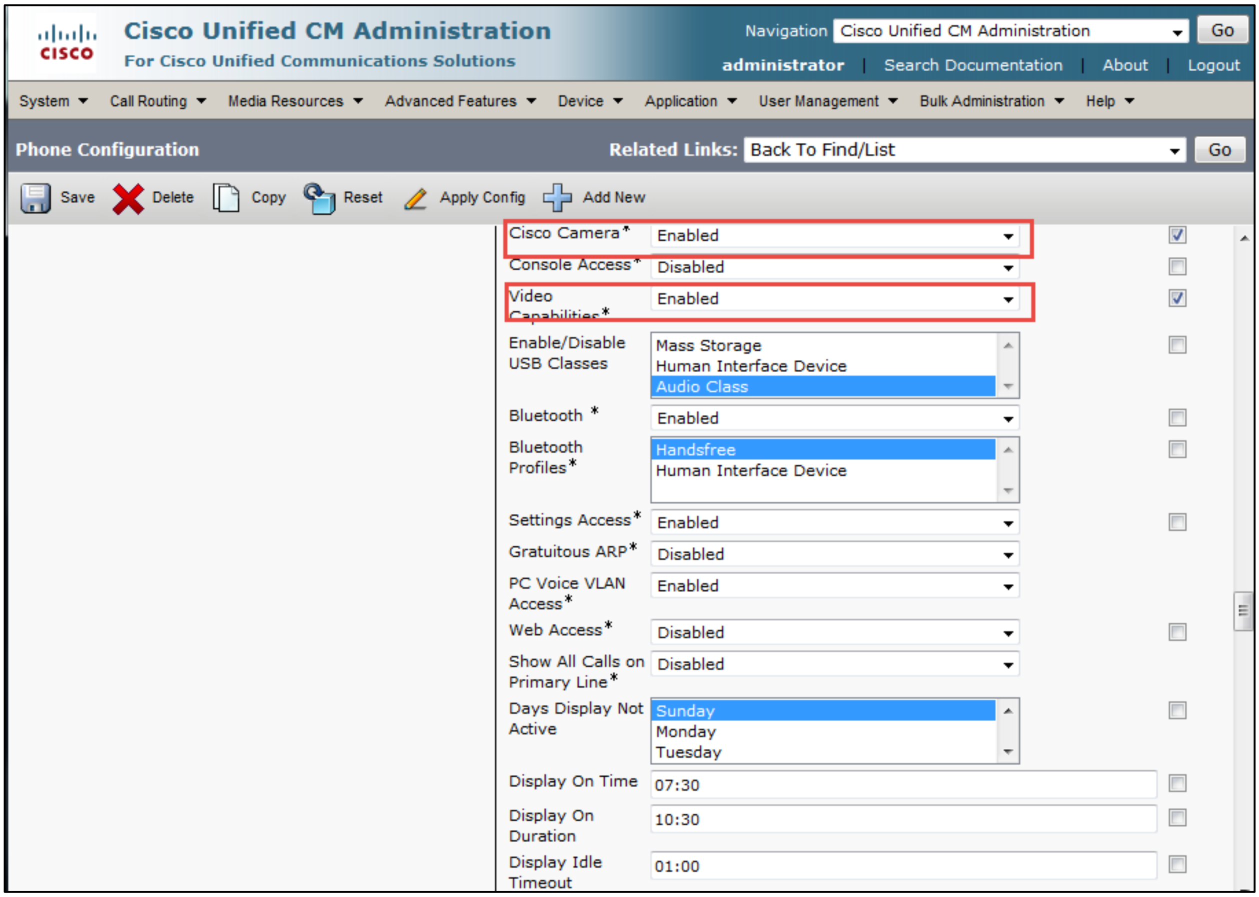



## Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)

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

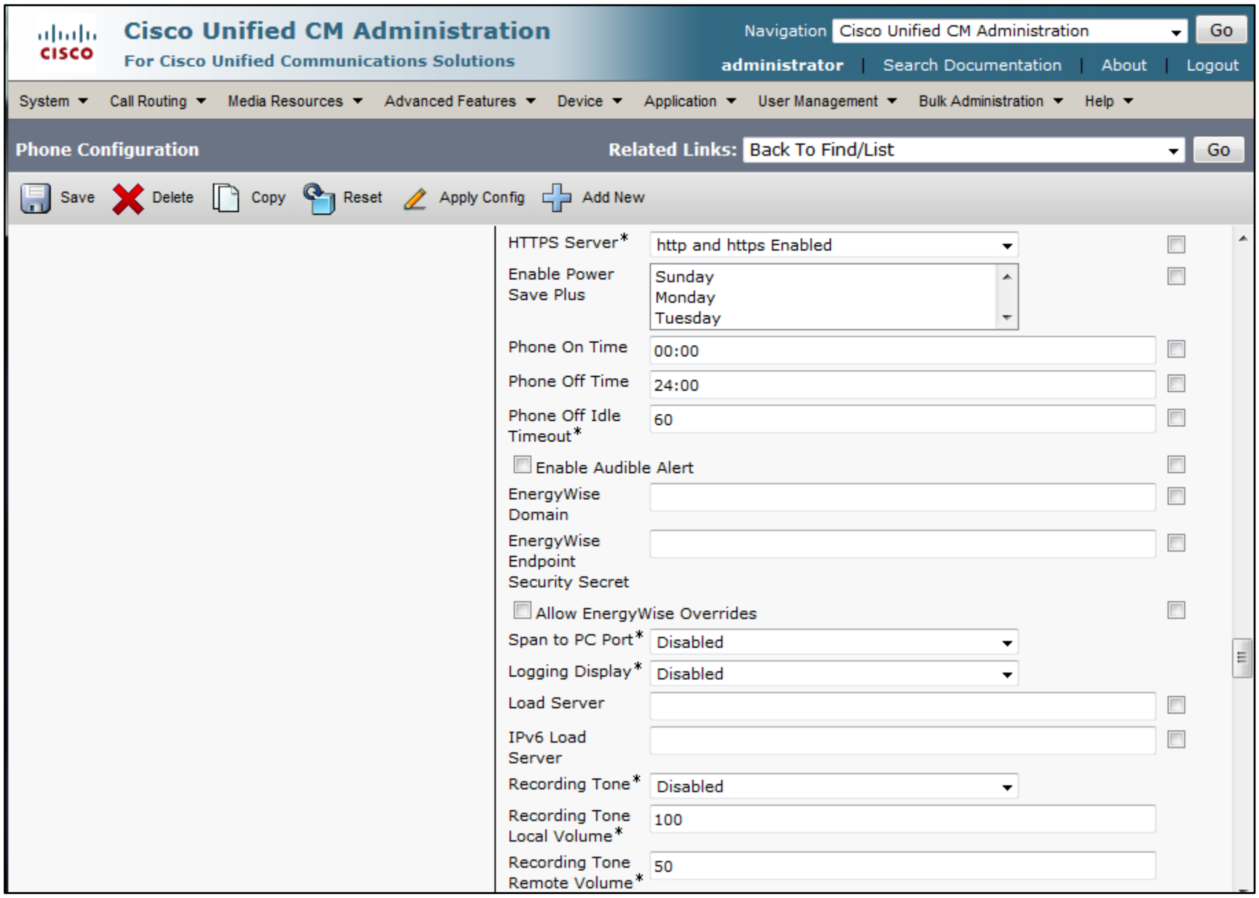
Set Video Capablities* = Enabled. This is used in this example.

All values are default.




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

All values are default.


Cisco Unified CM Administration
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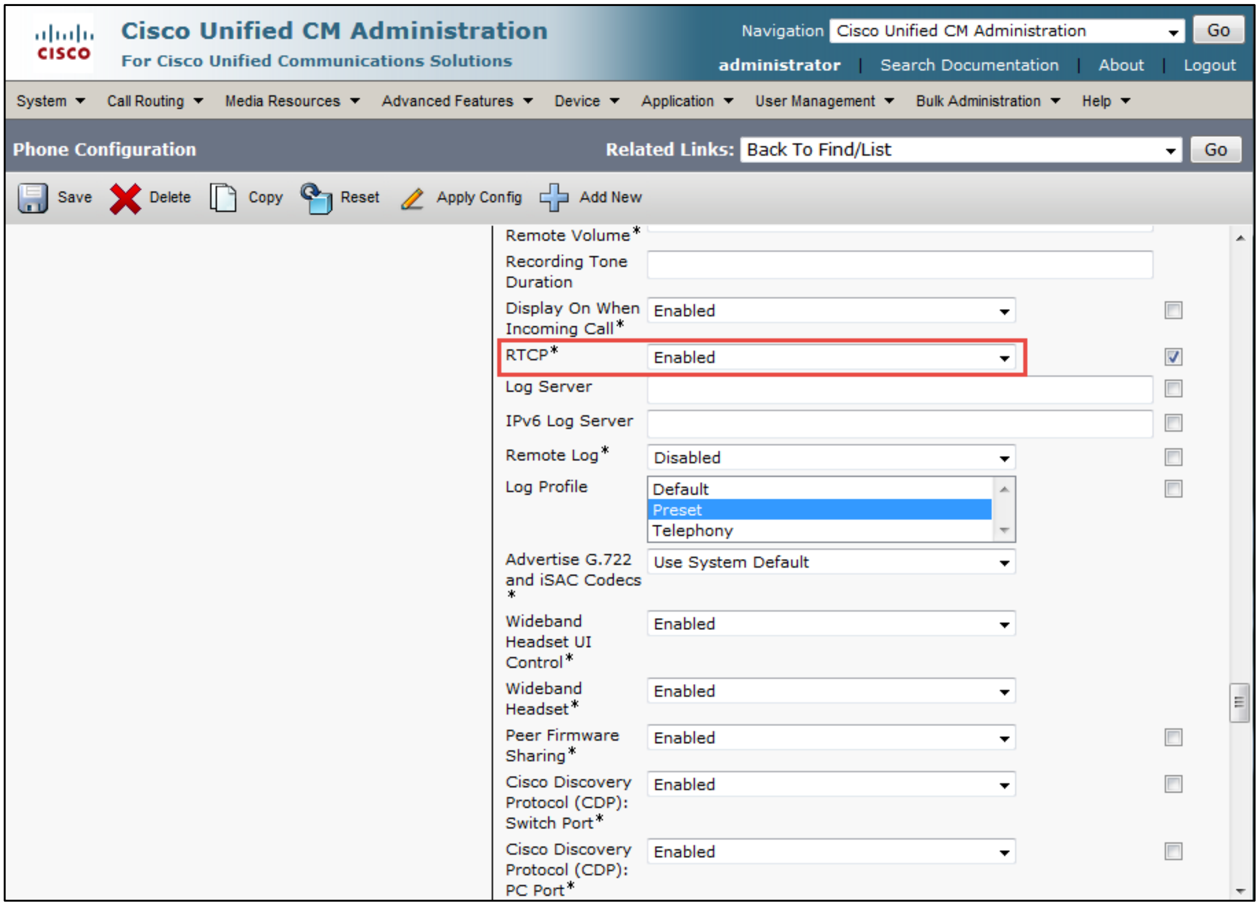
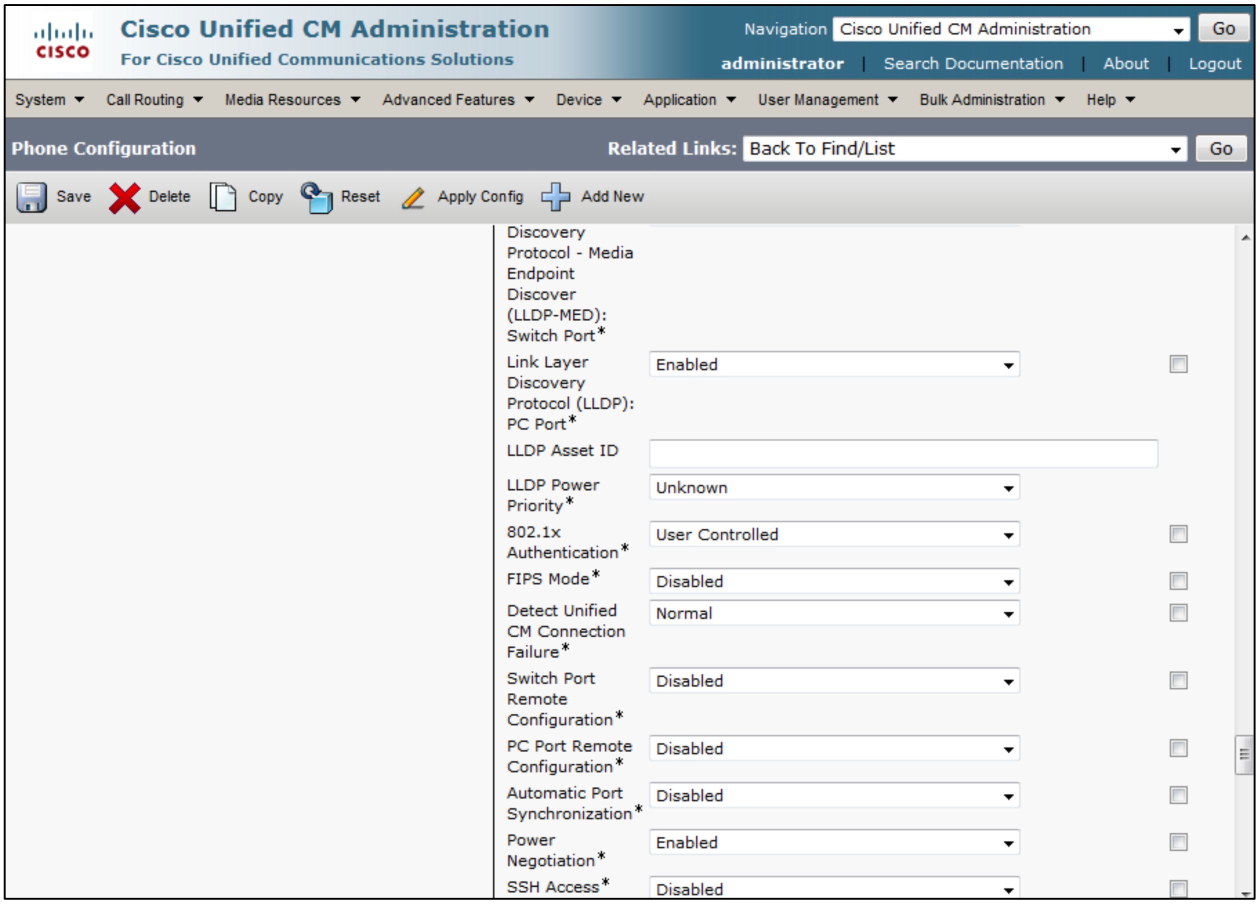
administrator | Search Documentation | About | Logout
System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help
Phone Configuration
Related Links: Back To Find/List Go
Save Delete Copy Reset Apply Config Add New
HTTPS Server* http and https Enabled
Enable Power Save Plus
Sunday
Monday
Tuesday
Phone On Time 00:00
Phone Off Time 24:00
Phone Off Idle Timeout* 60
Enable Audible Alert
EnergyWise Domain
EnergyWise Endpoint Security Secret
Allow EnergyWise Overrides
Span to PC Port* Disabled
Logging Display* Disabled
Load Server
IPv6 Load Server
Recording Tone* Disabled
Recording Tone Local Volume* 100
Recording Tone Remote Volume* 50

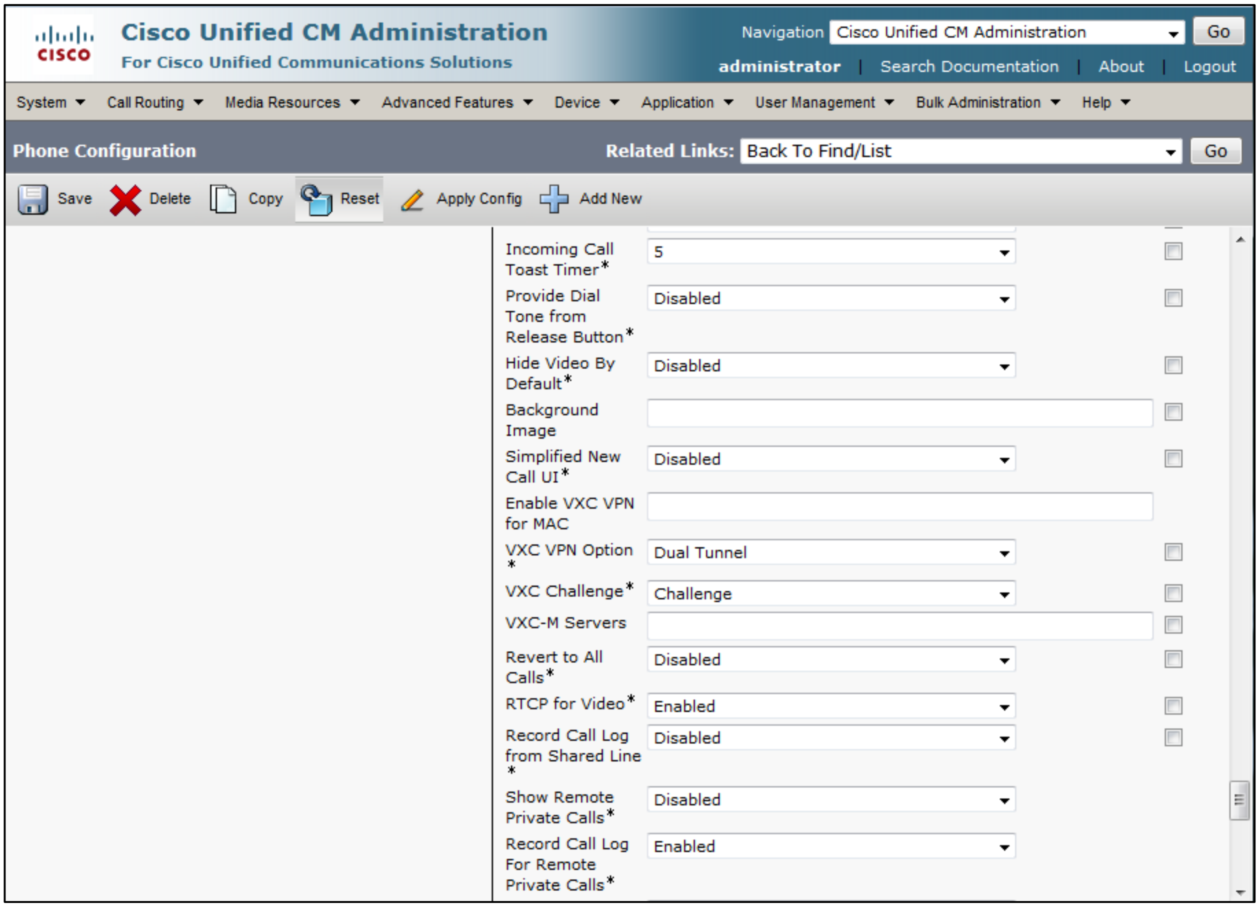
**Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)**



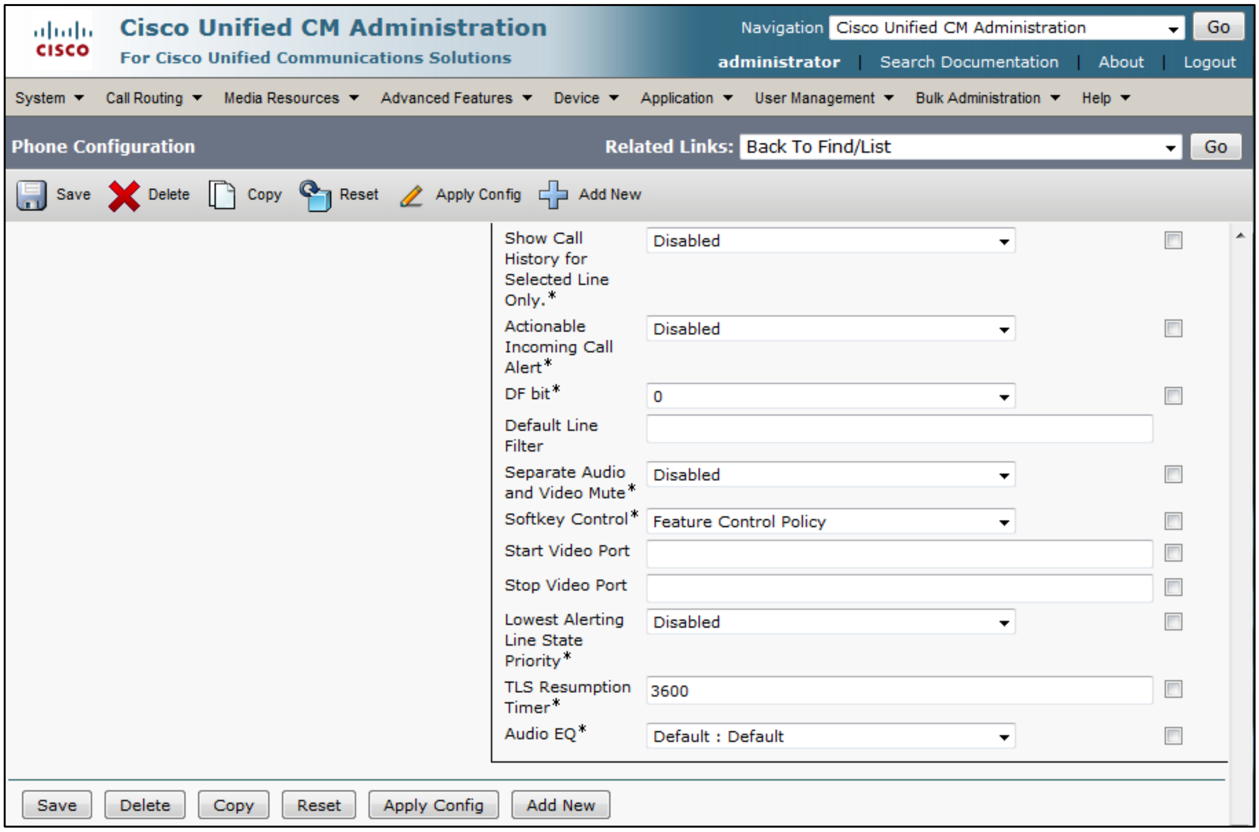
## Cisco Unified Communications Manager SIP Phone Device Level Configuration (Continued)



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



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



## Cisco Unified Communications Manager SCCP Phone Device Level Configuration

**Navigation Path:** Device→ Phone

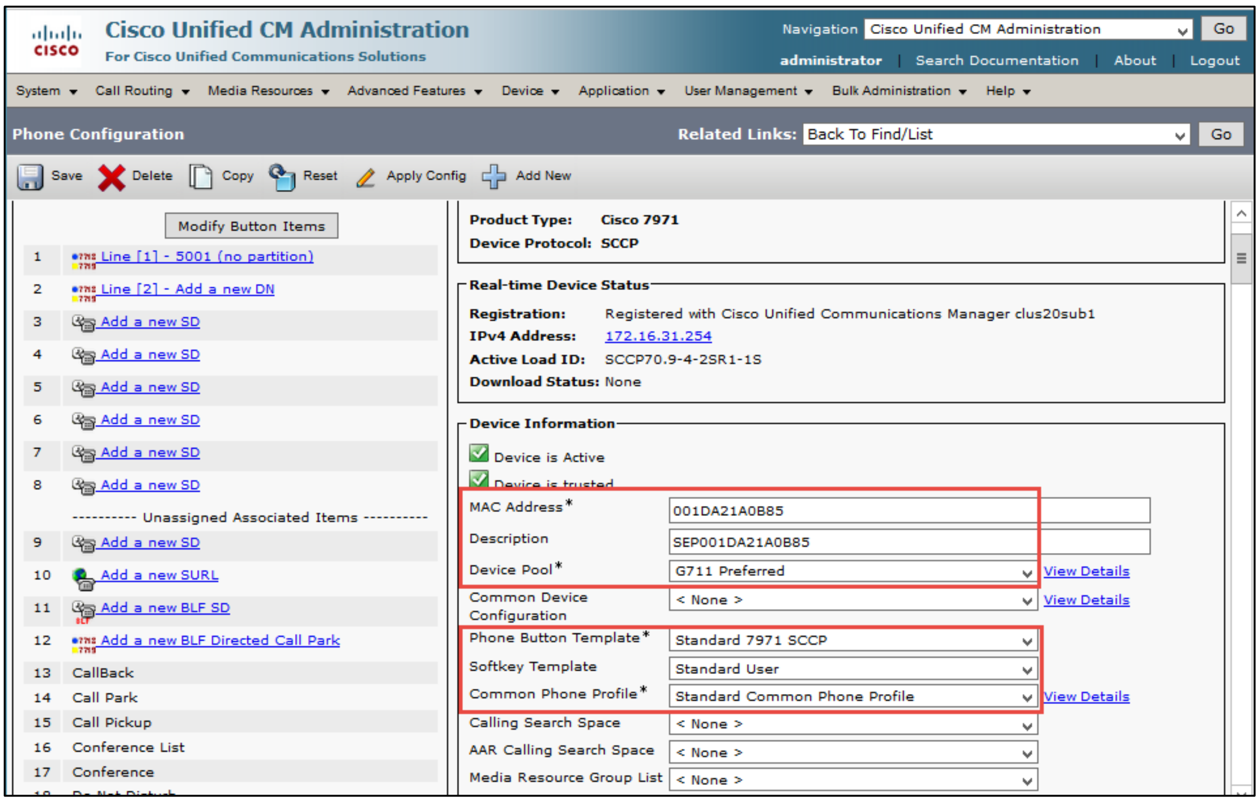
Set MAC Address* = 001DA21A0Bxx. This is used in this example.

Set Description = this text is used to identify this Phone

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

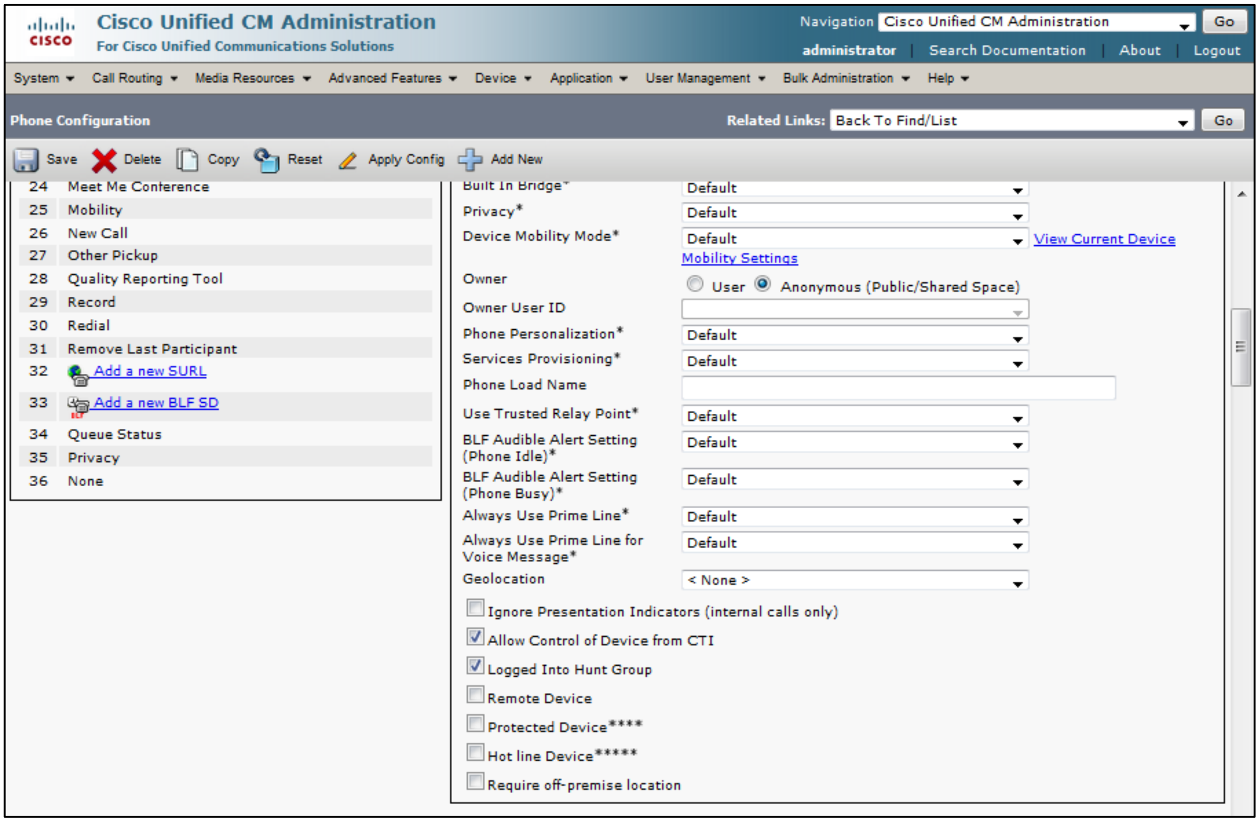
Set Phone Button Template*= Standard 7971 SCCP. This is used in this example

All other values are default.



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

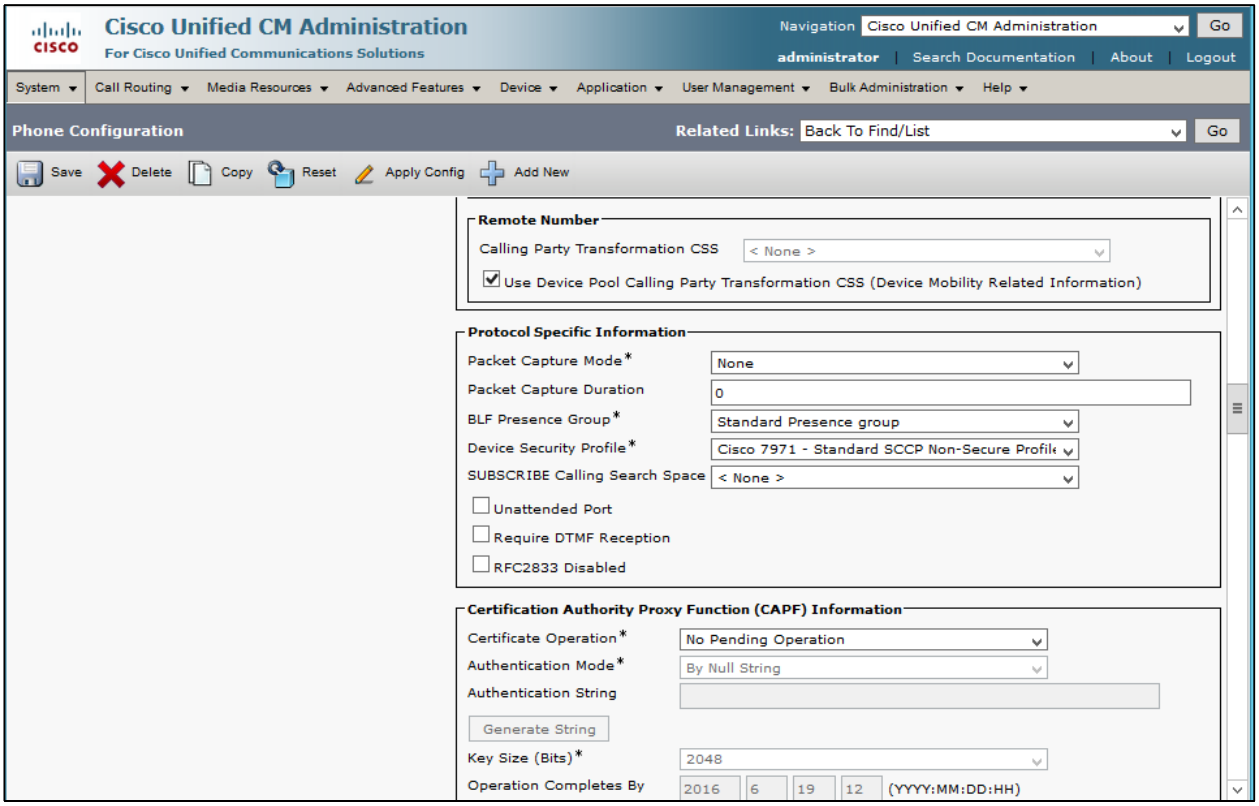
All other values are default.



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

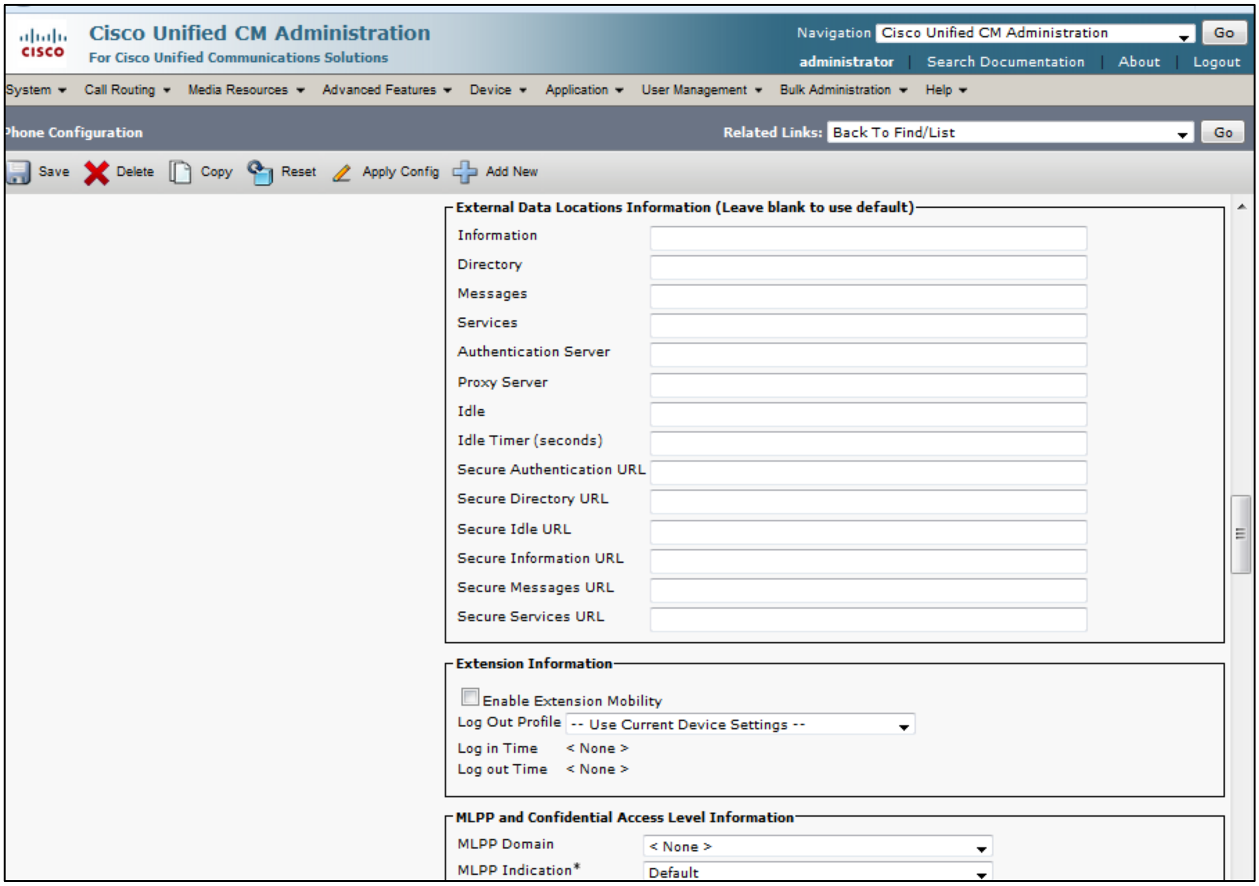
Set Device Security Profile* = Cisco 7971 – Standard SCCP Non-Secure Profile. This is used in this example

All other values are default.



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

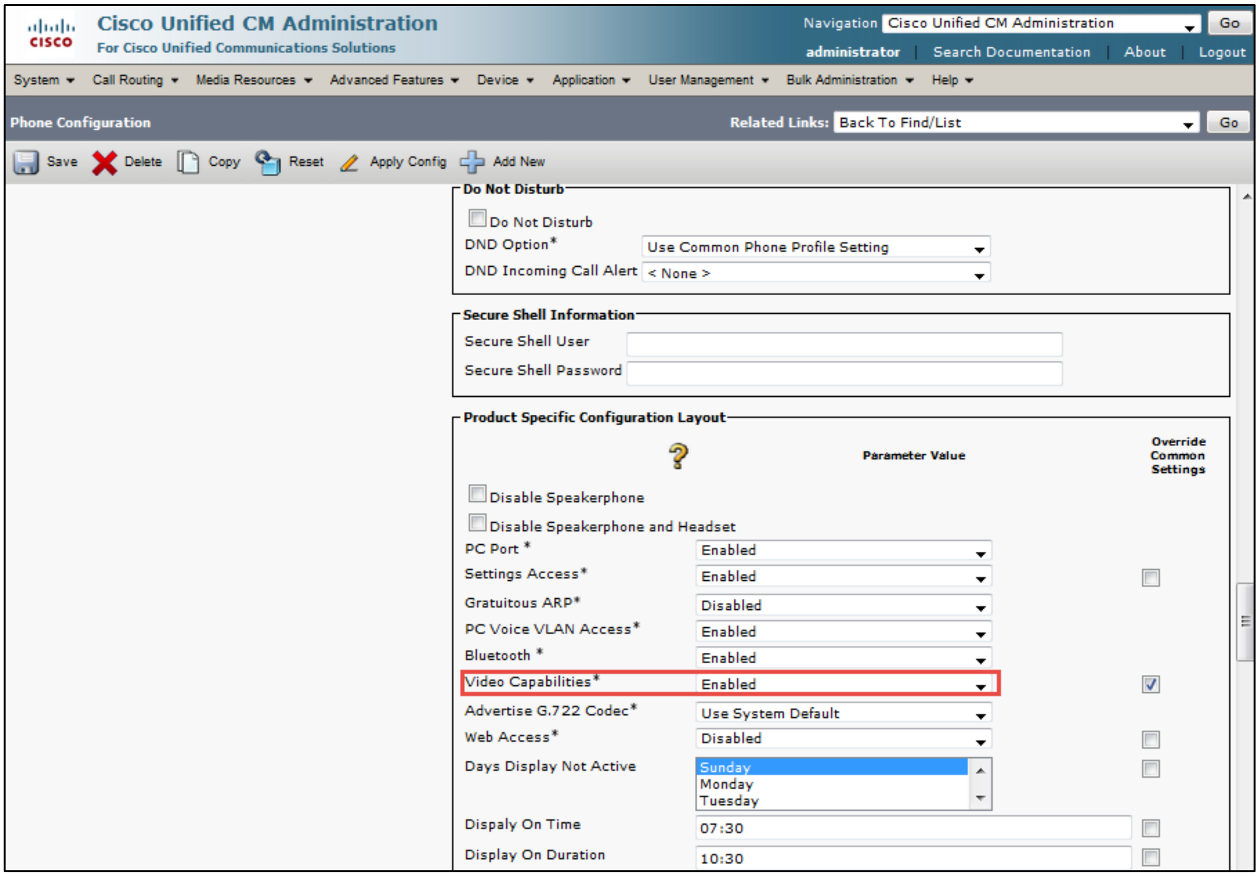
These values are default.



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

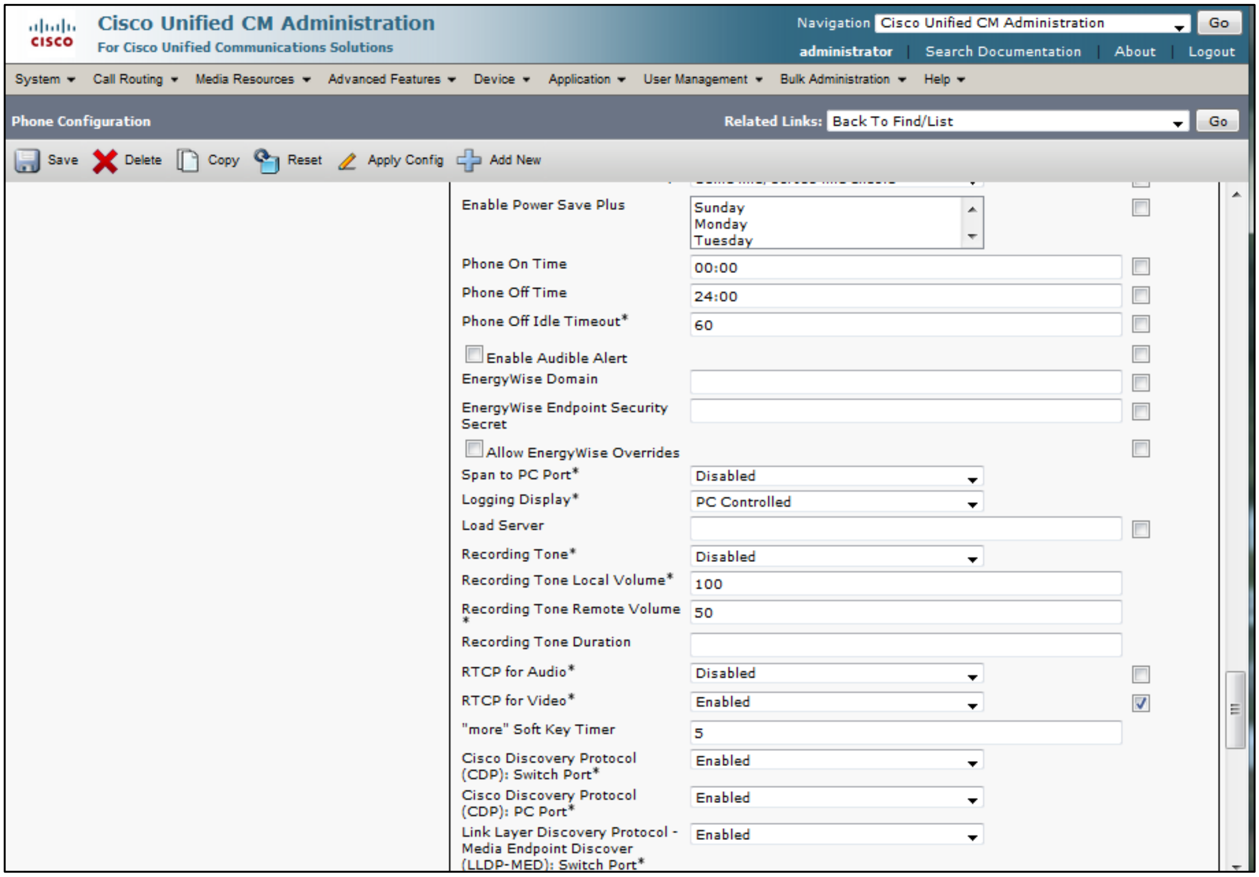
Set Video Capabilities* = Enabled

These values are default.



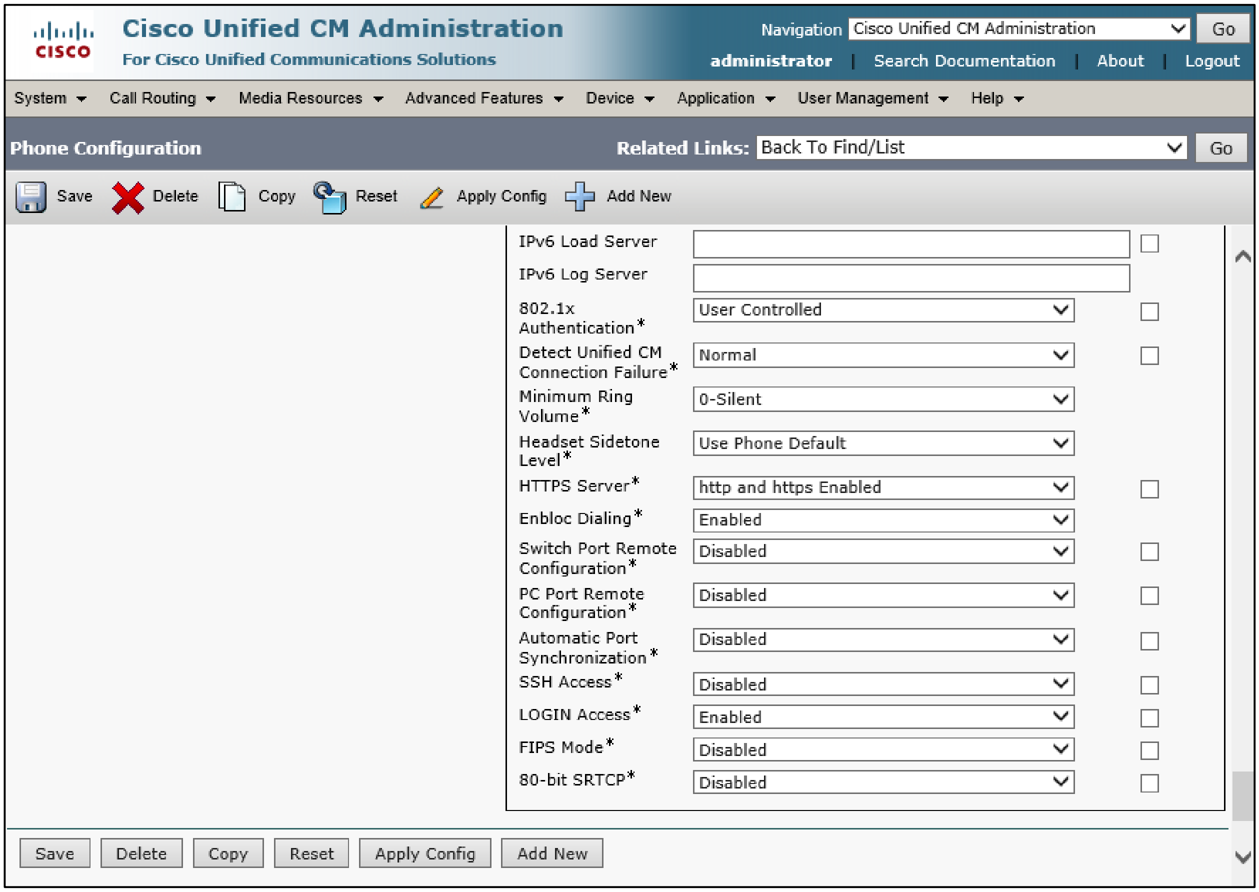
## Cisco Unified Communications Manager SCCP Phone Device Level Configuration (Continued)

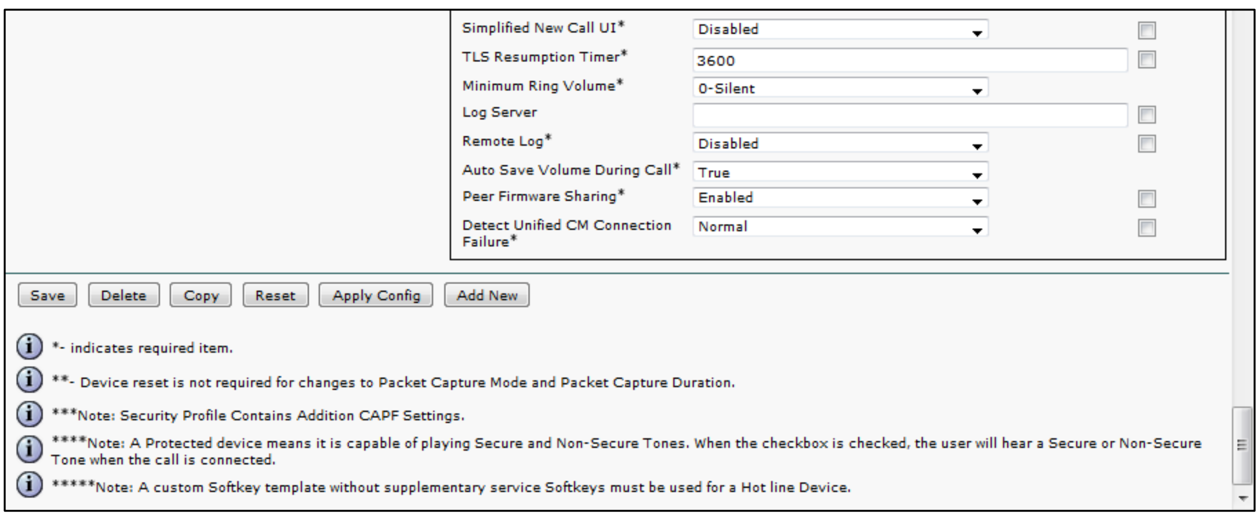
These values are default.



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

These values are default.

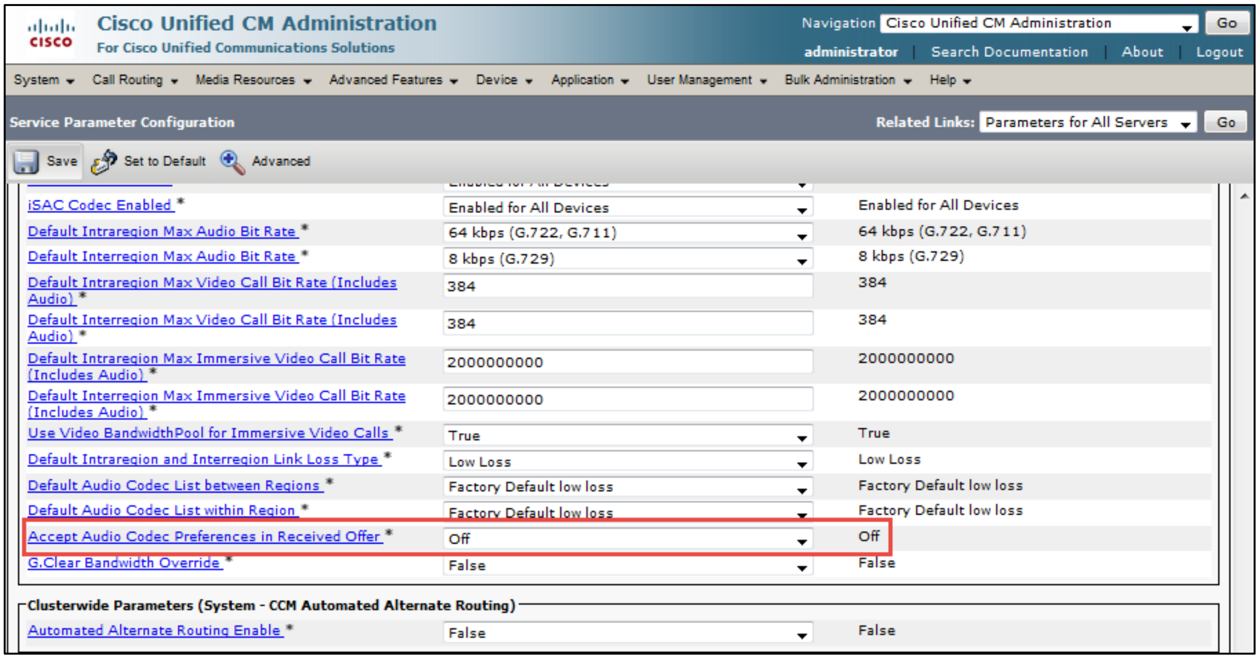


## Cisco Unified Communications Manager Audio Codec Preference List Configuration

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

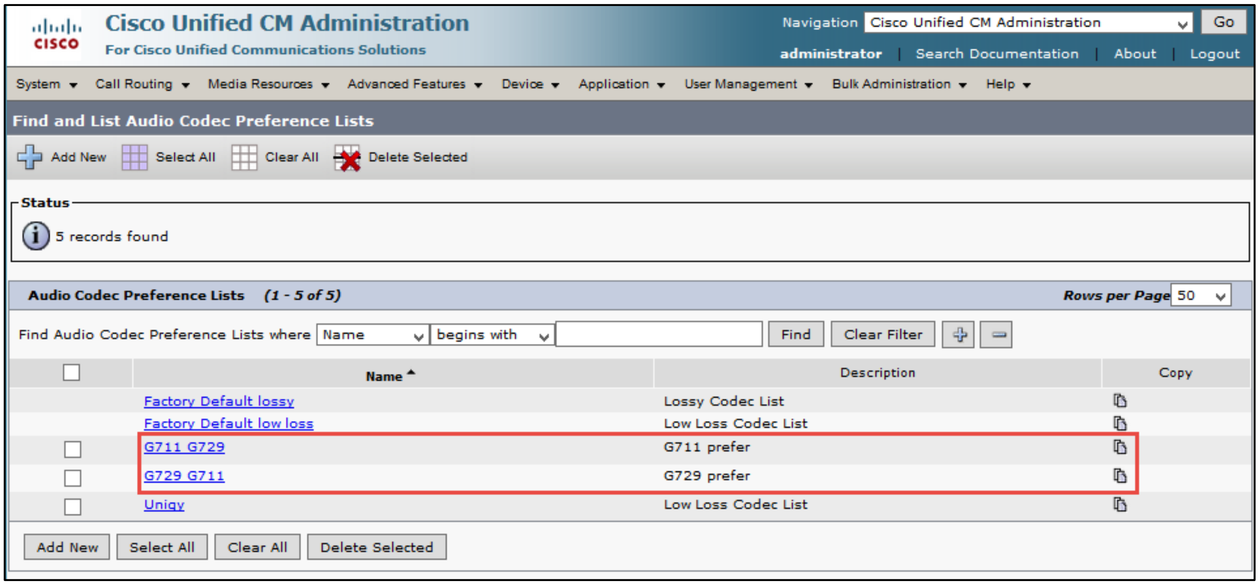
Set Accept Audio Codec Preference in Received Offer *= Off. This needs to be set when you are wanting to use the Codec Preference List created.



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

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

All other values are default.



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

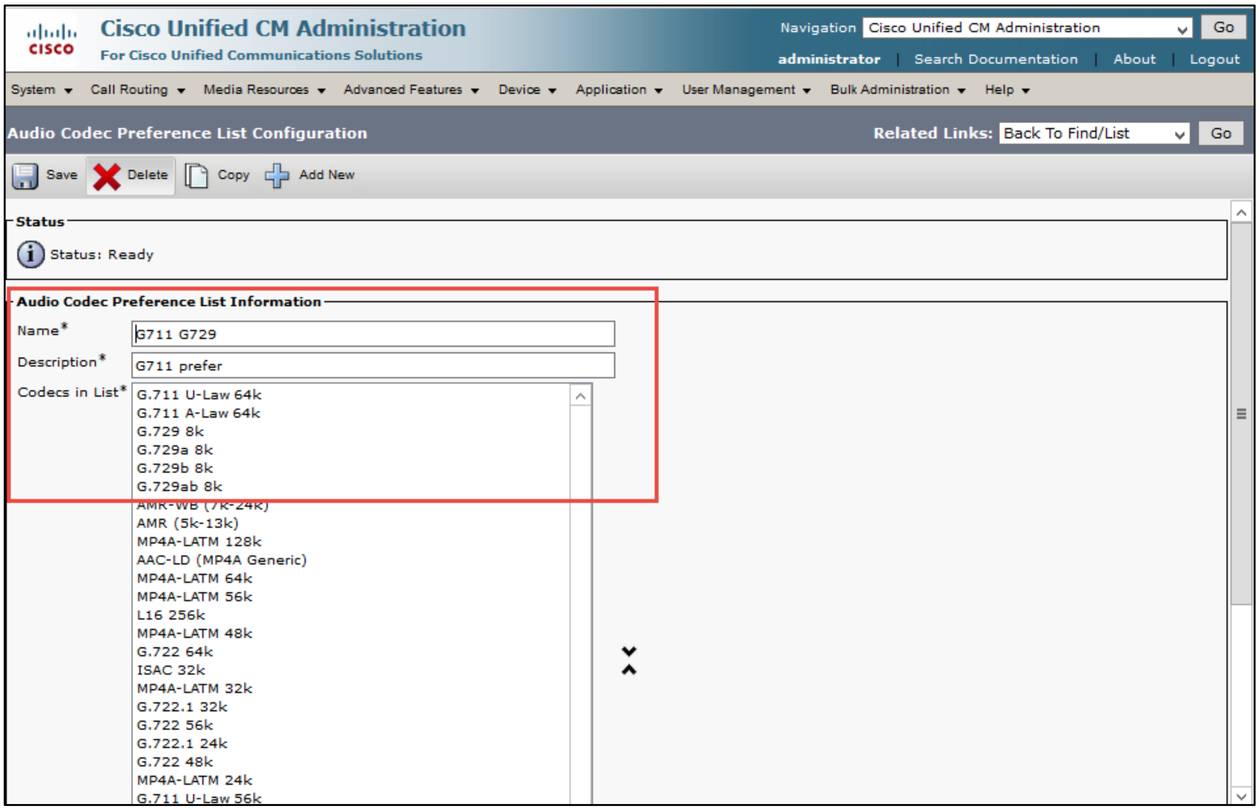
Set Name*= G711 G729.  This is used for this example

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

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

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

All other values are default.



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

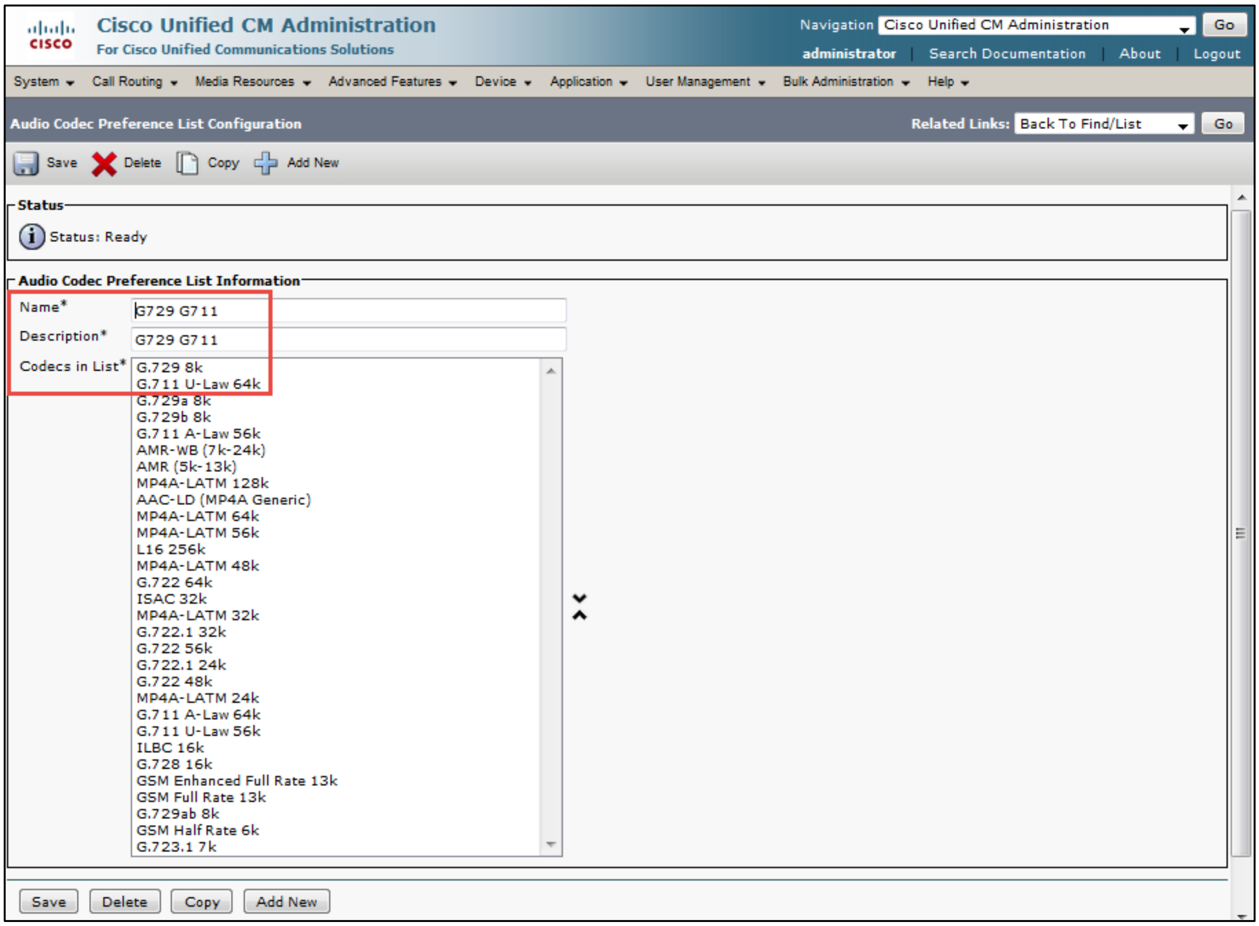
Set Name*= G729 G711.  This is used for this example.

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.




# Cisco UCM Extent and Connect

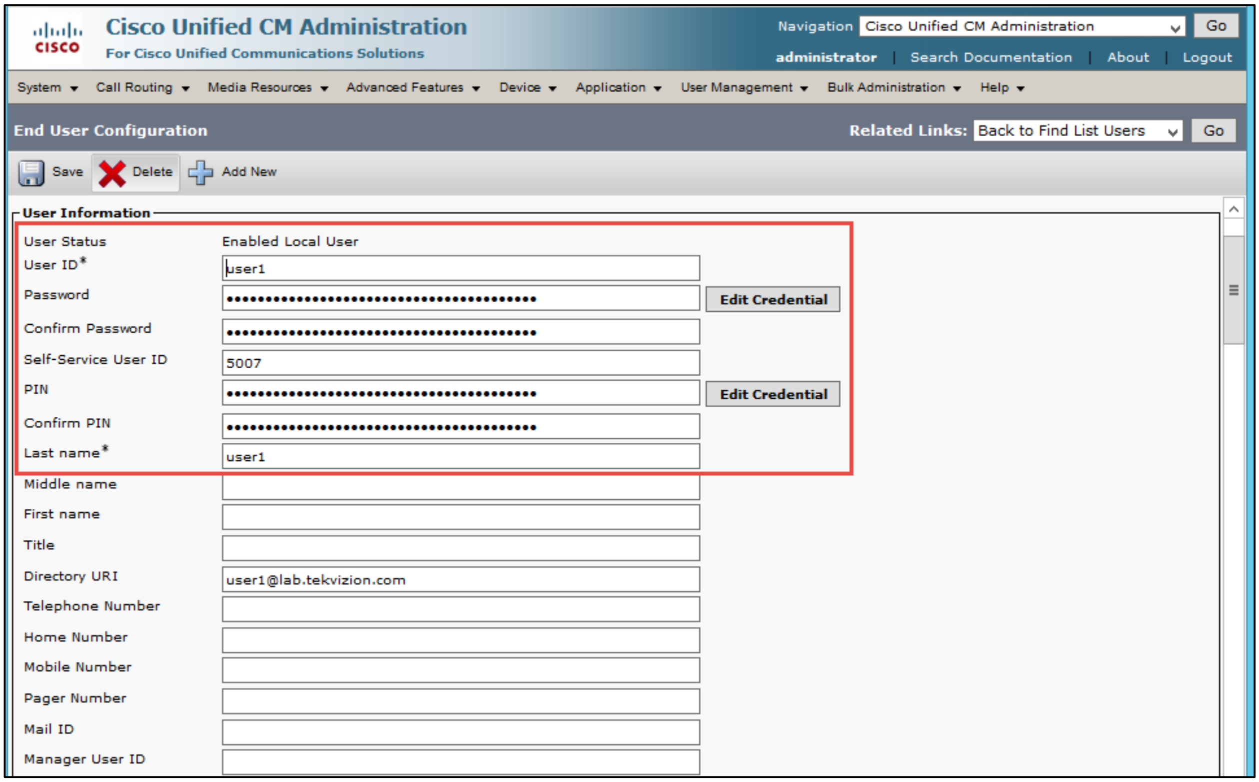
Extend and Connect is a feature that allows administrators to rapidly deploy UC Computer Telephony Integration (CTI) applications which interoperate with any endpoint. With Extend and Connect, users can leverage the benefits of UC applications from any location using any device. This feature also allows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

## Cisco UCM end user configuration

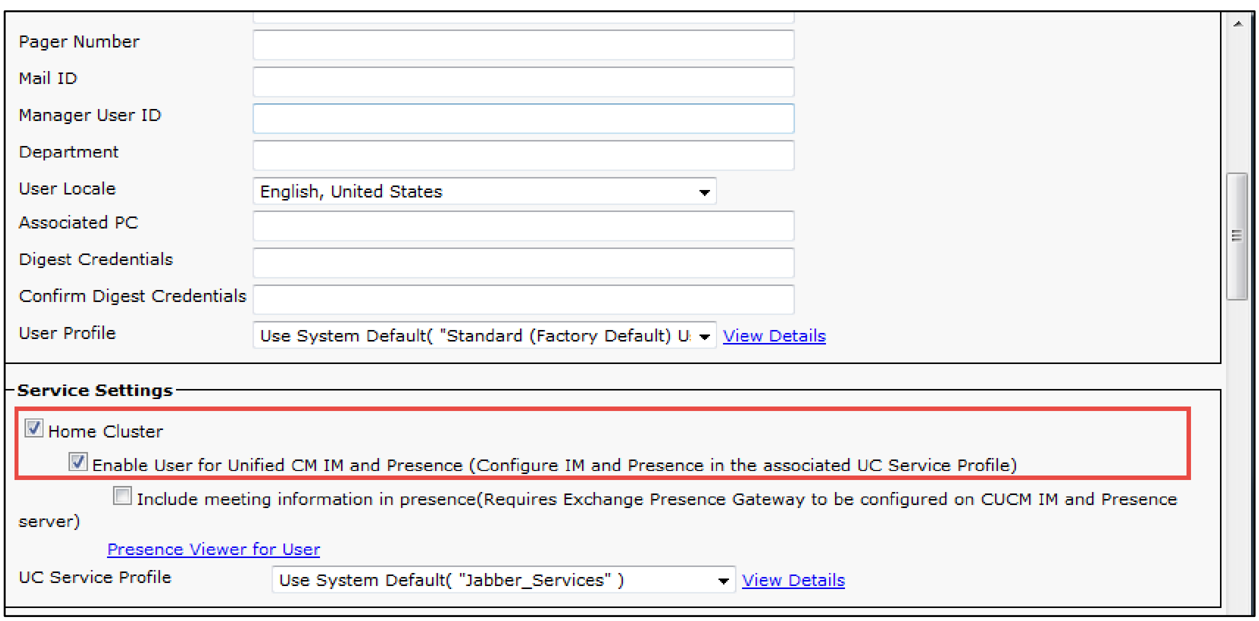
Add user to Cisco UCM

**Navigation Path:** User Management→ End user

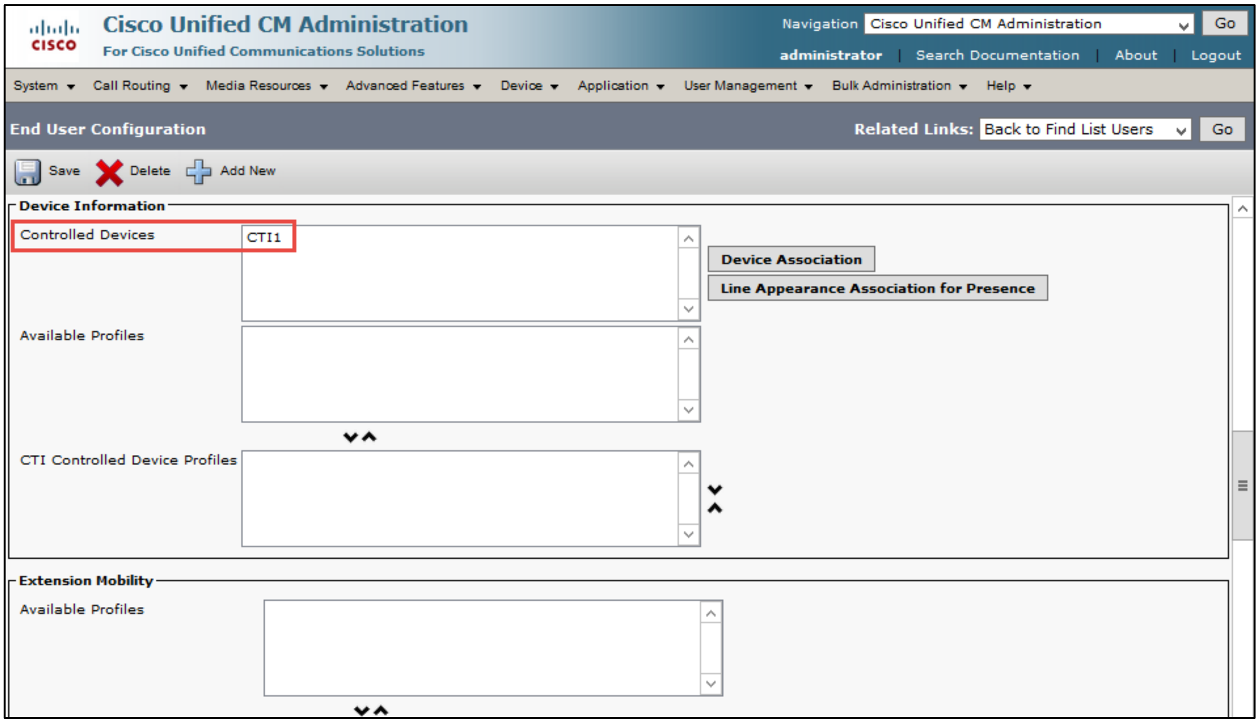
Set User ID*= user1. This is used for this example.
Set Last Name = user1. This is used for this example.
Check Home Cluster.




## Cisco UCM end user Configuration (Continued)

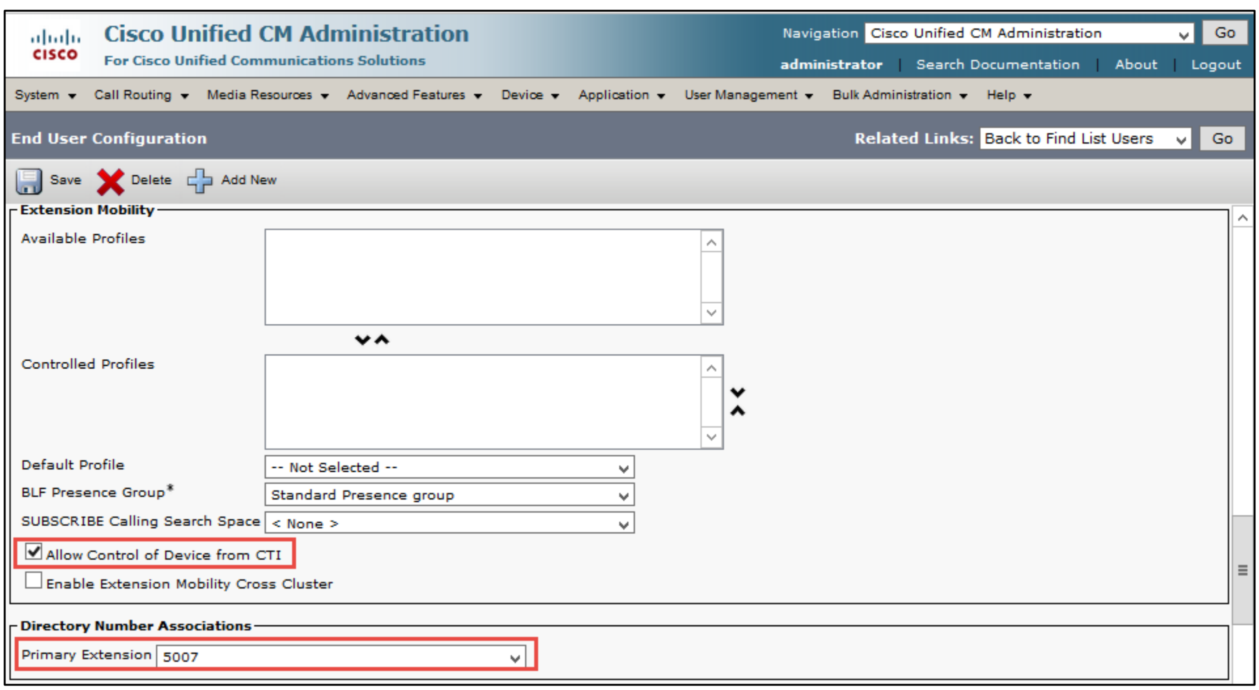


Set Controlled Devices = CTI1. This is used for this example.

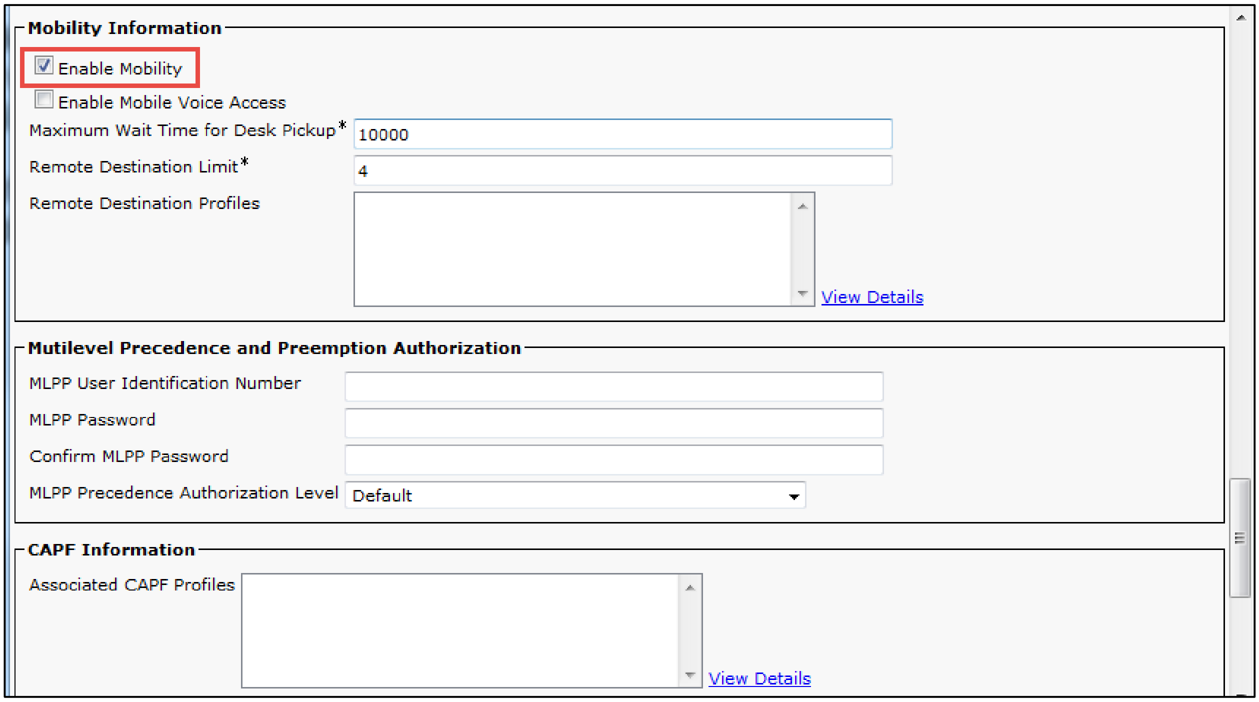



## Cisco UCM end user Configuration (Continued)

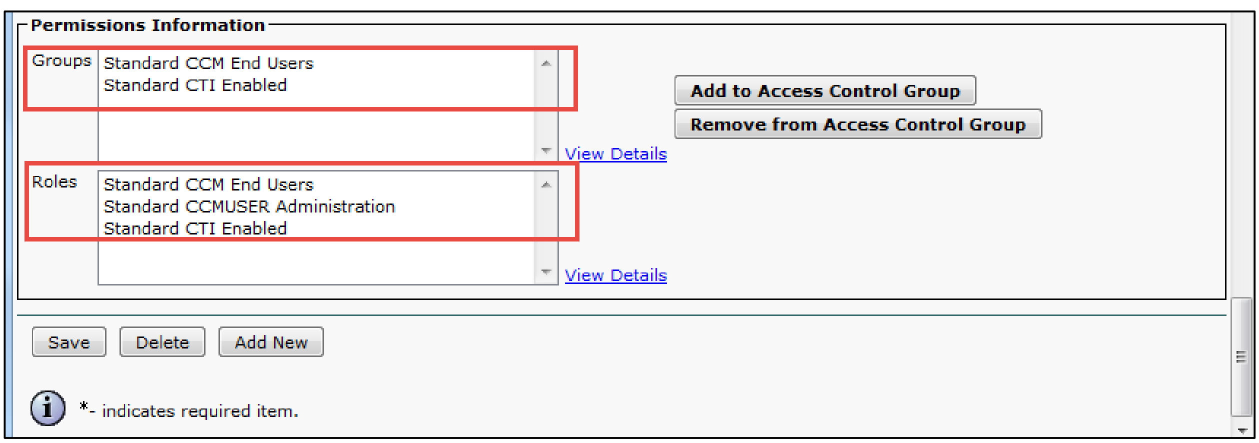
Check Allow Control of Device from CTI
Select the Primary Extension for this user.5007 is used for this example.



Check Enable Mobility

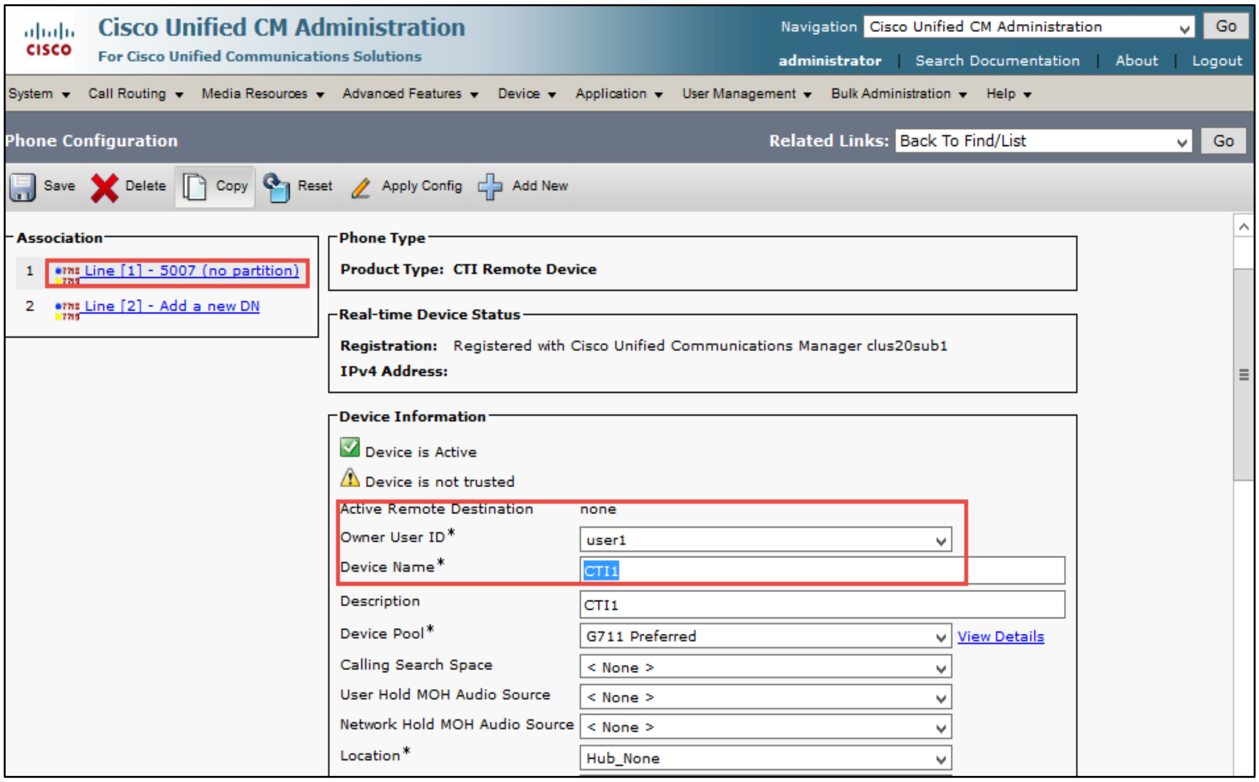



Add the following permissions for Standard Users:
–Standard CCM End-Users
–Standard CTI Enabled
–Standard CCMUSER Administration

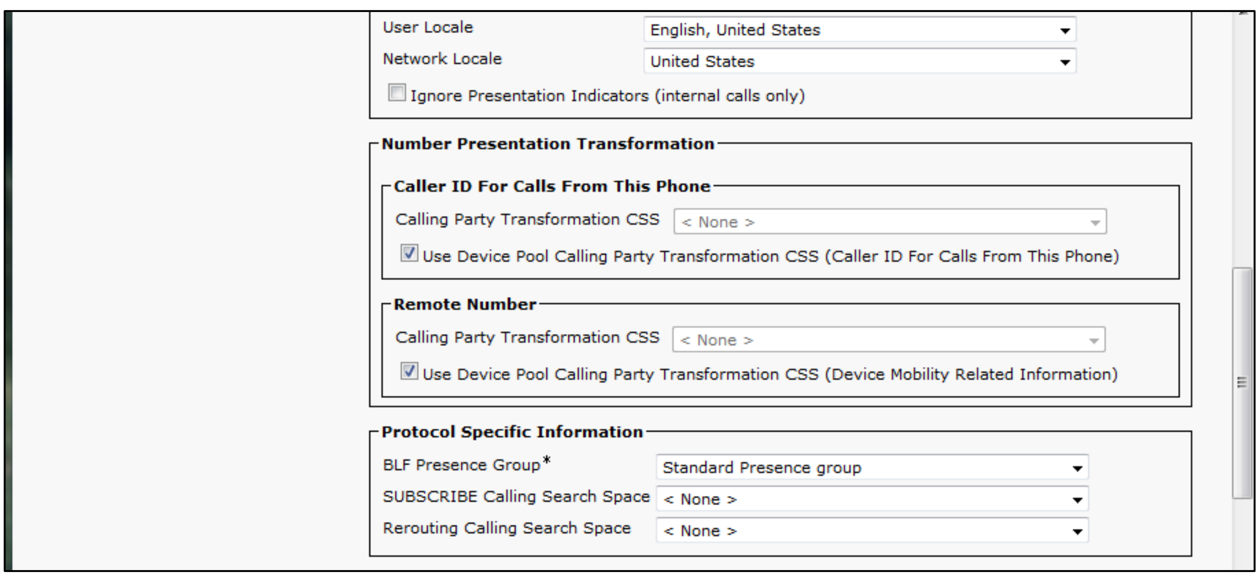



Add Phone: CTI Remote Device

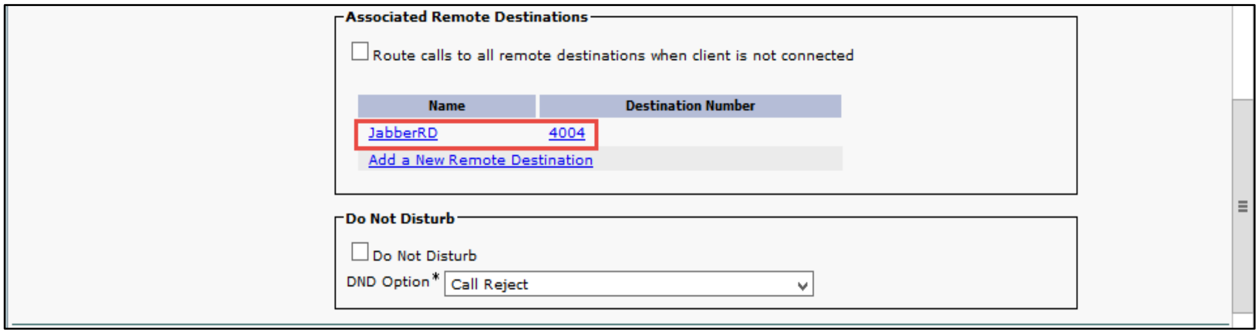
The CTI Remote Device type represents the user’s remote device(s).
Select the desired Owner User ID .User1 is used in this example.
Set the Device Name populated automatically. Modify if desired - CTI1 used this example.
Set Device Pool: Default. This is used in this example.




## Cisco UCM CTI remote device Configuration (Continued)

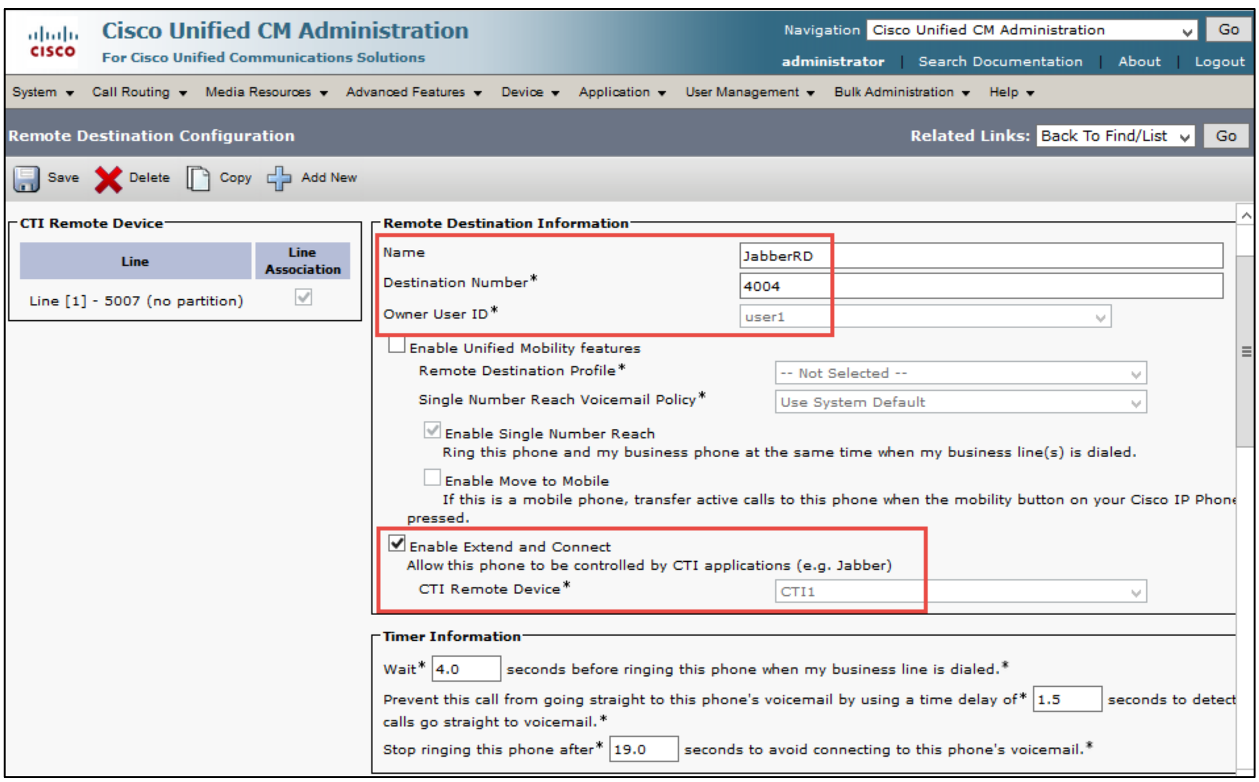


Set RD*= 4004. This is used for this example.4004 is the Alcatel extension.



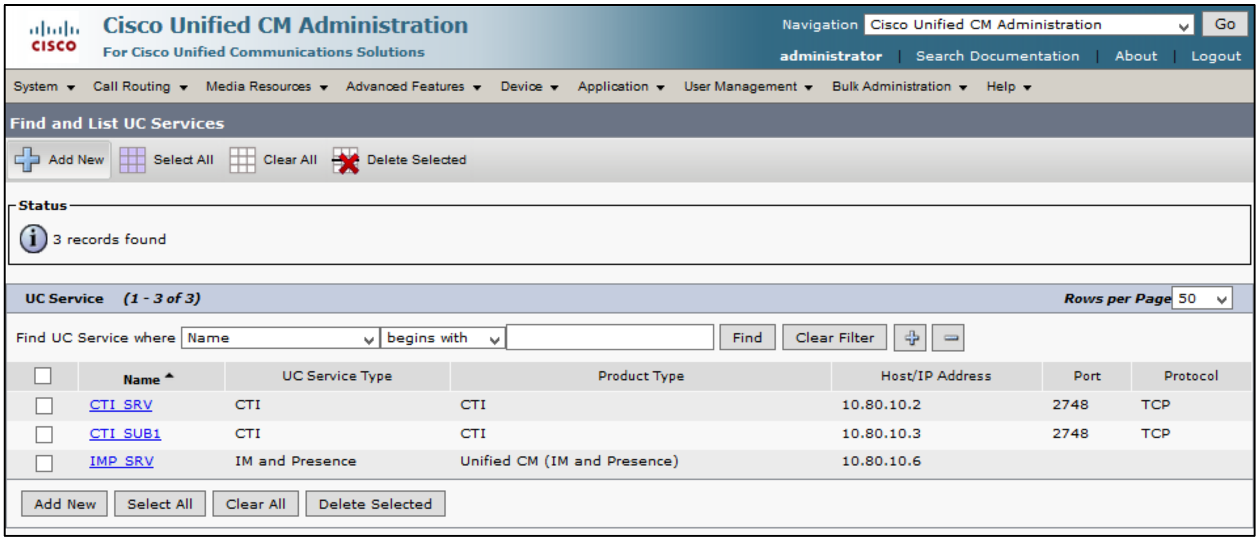

## Remote Destination Configuration

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



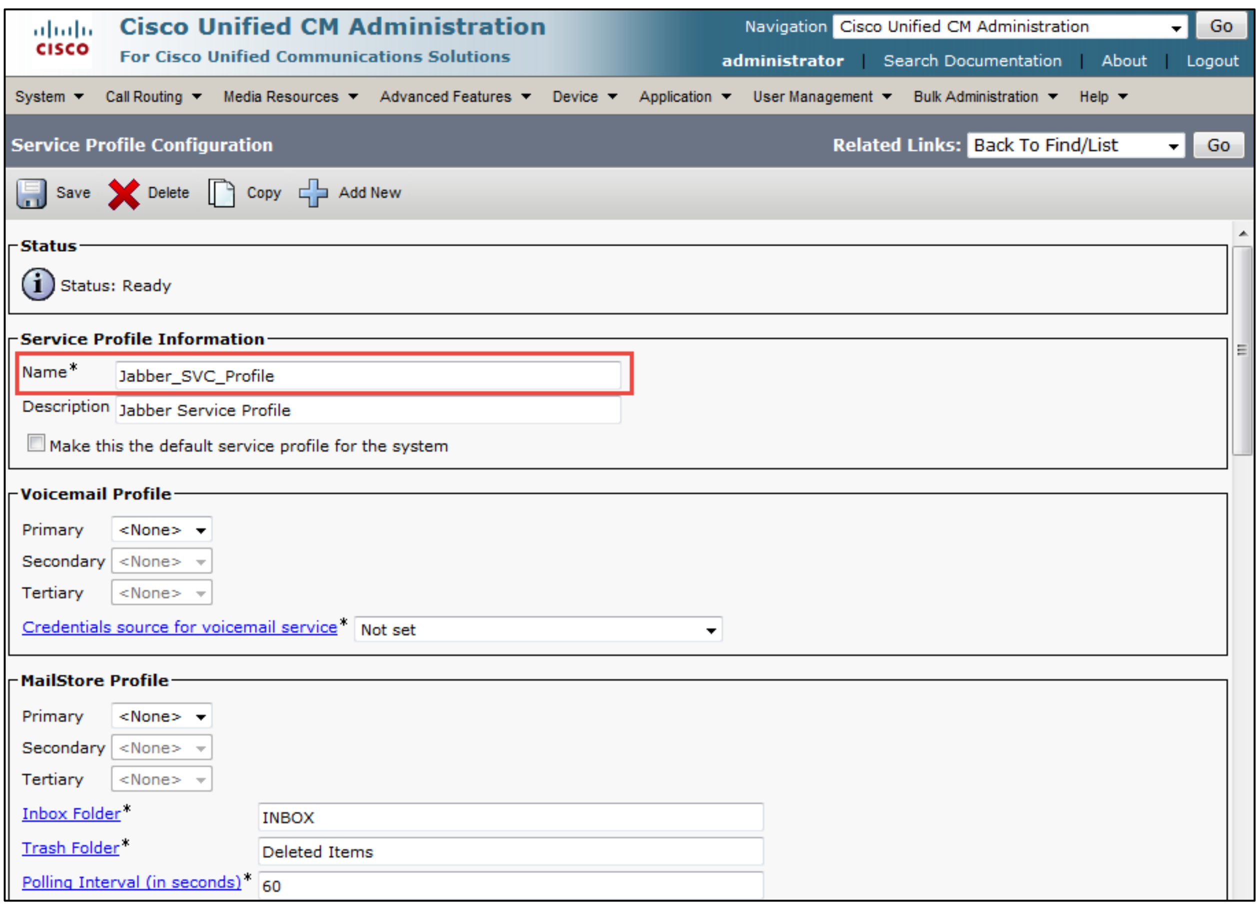
## Cisco UCM UC service Configuration

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

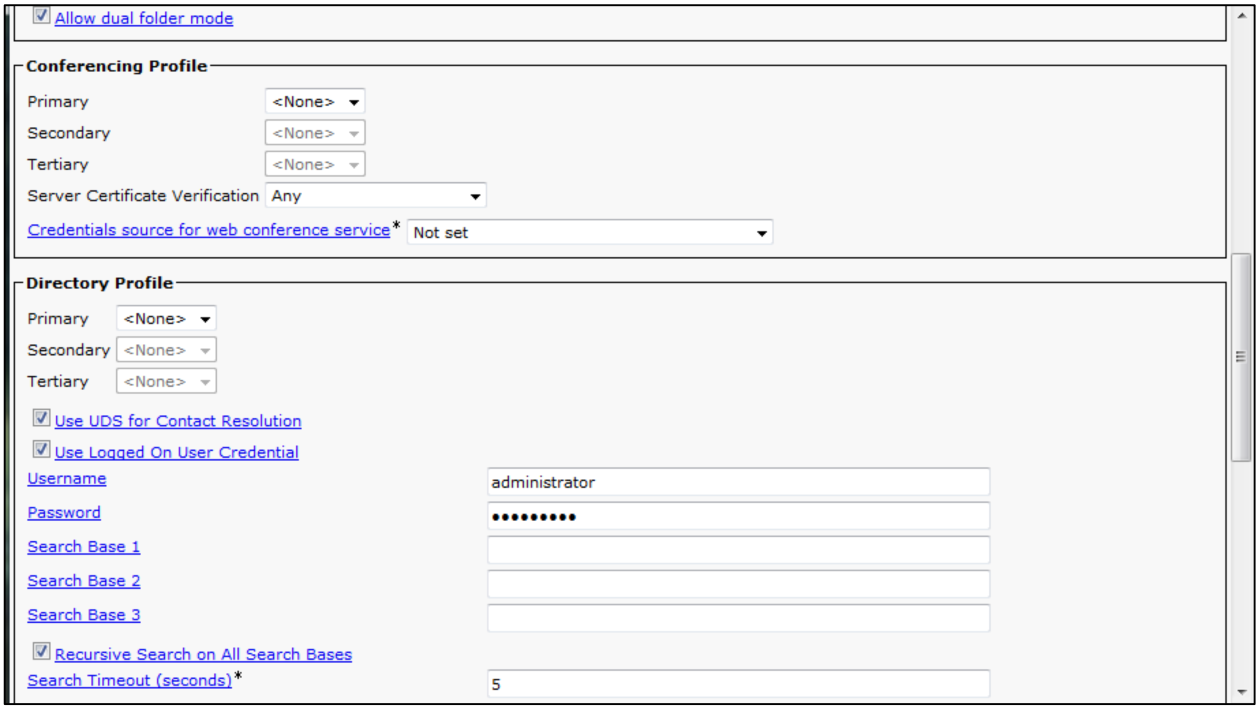


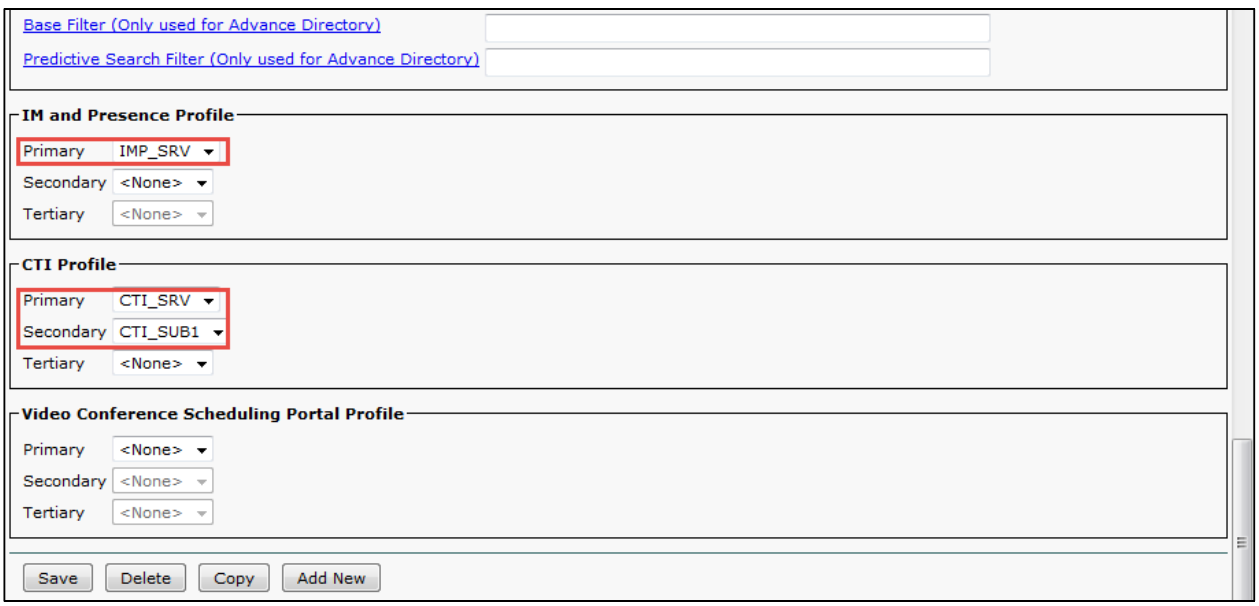

## Cisco UCM service Profile Configuration

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




**Cisco UCM service profile Configuration (Continued)**

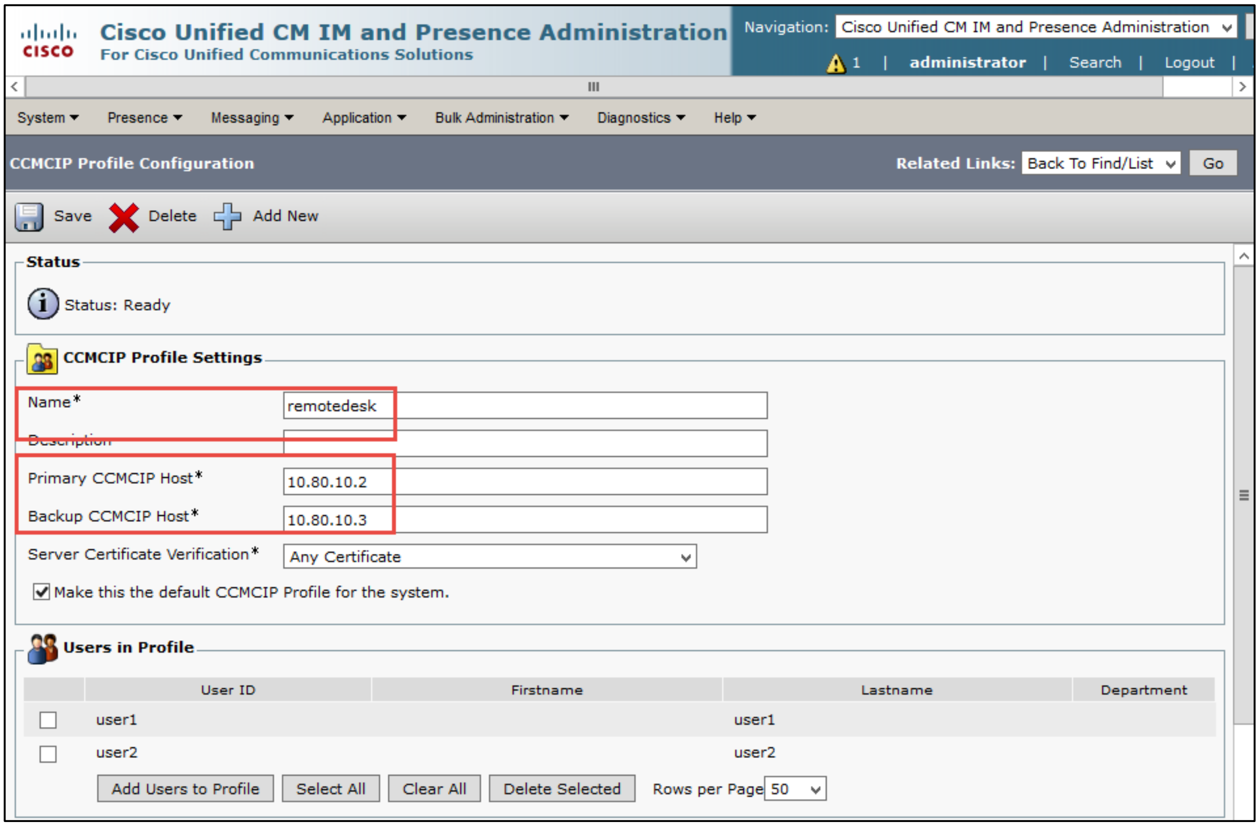

Allow dual folder mode
Conferencing Profile
Primary
<None>
Secondary
<None>
Tertiary
<None>
Server Certificate Verification
Any
Credentials source for web conference service*
Not set
Directory Profile
Primary
<None>
Secondary
<None>
Tertiary
<None>
Use UDS for Contact Resolution
Use Logged On User Credential
Username
administrator
Password
Search Base 1
Search Base 2
Search Base 3
Recursive Search on All Search Bases
Search Timeout (seconds)*
5


Base Filter (Only used for Advance Directory)
Predictive Search Filter (Only used for Advance Directory)
IM and Presence Profile
Primary
IMP_SRV
Secondary
<None>
Tertiary
<None>
CTI Profile
Primary
CTI_SRV
Secondary
CTI_SUB1
Tertiary
<None>
Video Conference Scheduling Portal Profile
Primary
<None>
Secondary
<None>
Tertiary
<None>
Save
Delete
Copy
Add New

## 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.

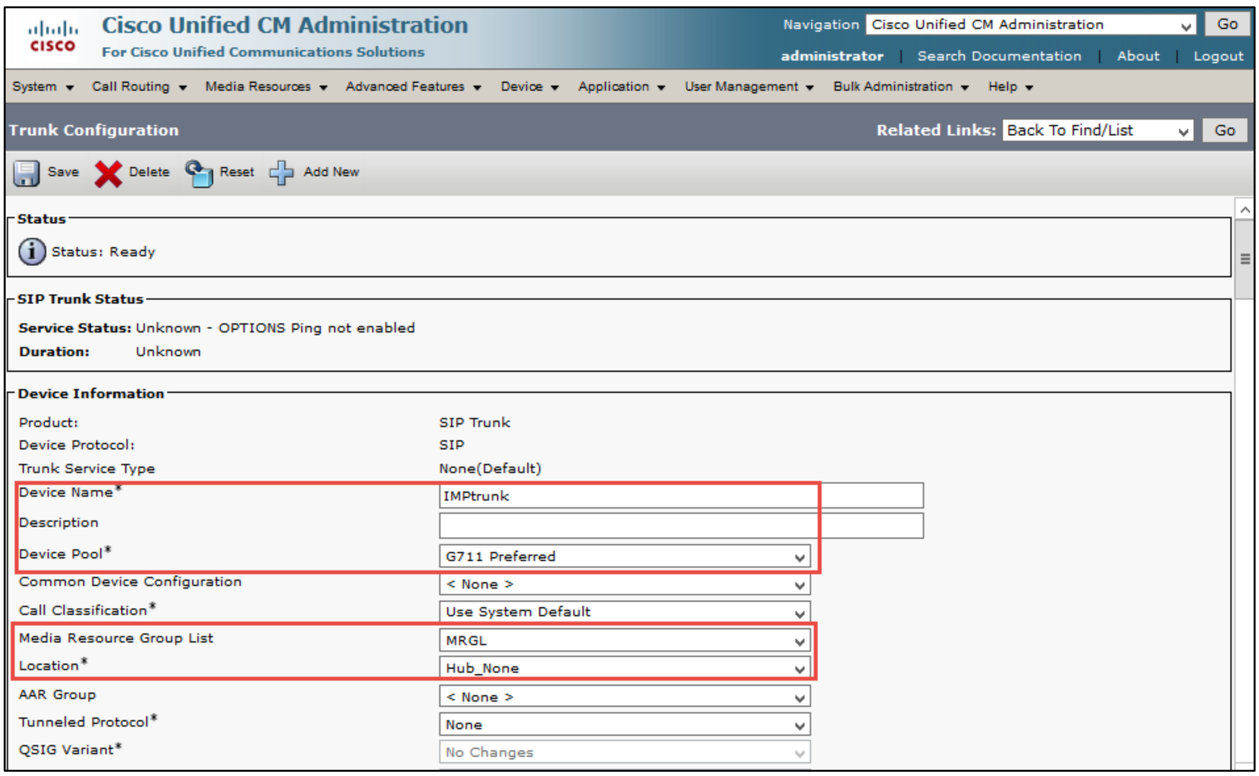


## Cisco UCM – SIP trunk to Cisco IM&Presence Trunk Configuration

**Navigation Path:** Device→ Trunk

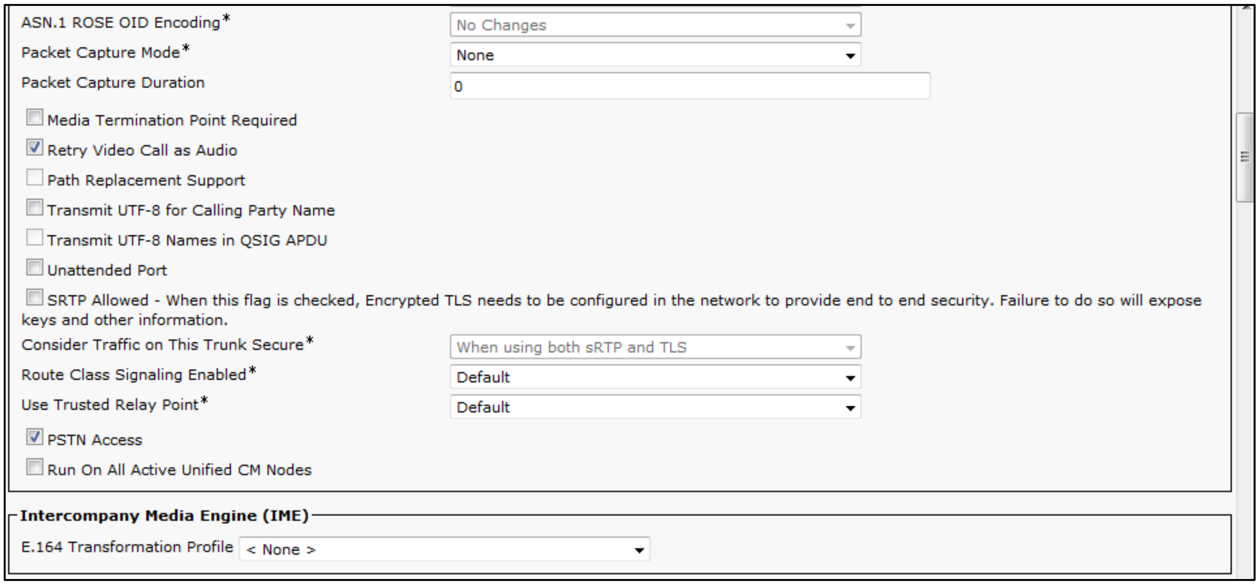
Set Device Name*= IMPTrunk.  This is used for this example.
Set Description = this text is used to identify this Trunk Group.
Set Device Pool* = Default. This is used for this example.
Set Media Resource Group List = MRGL. This is used for this example.

All other values are default.



## 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 |

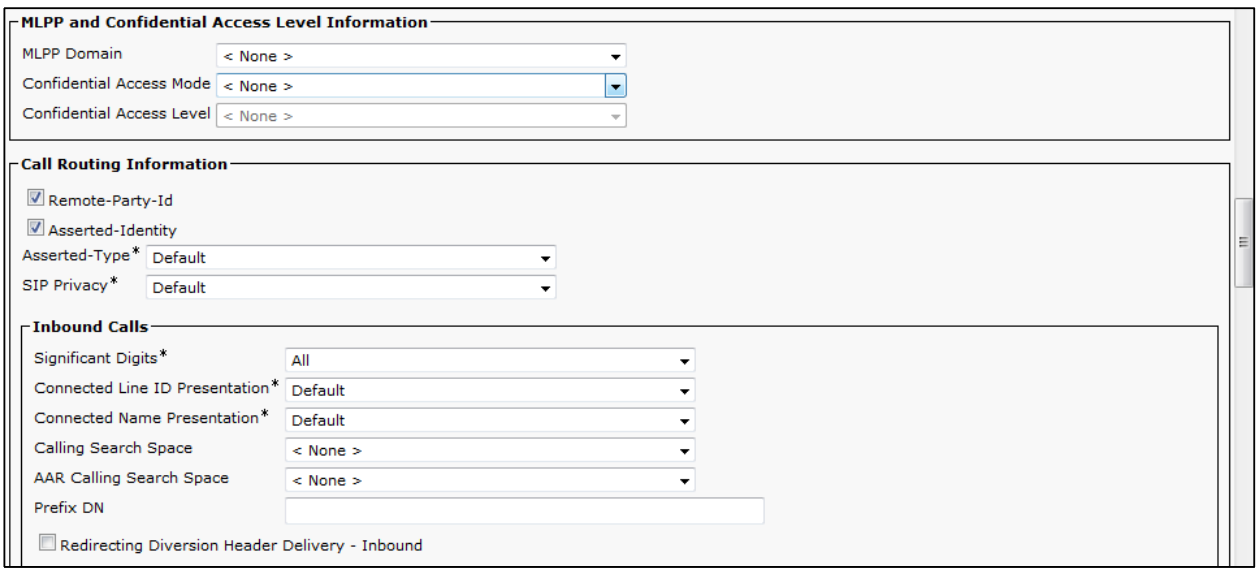
- [ ] Media Termination Point Required
- [x] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port
- [ ] SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

| | |
|---|---|
| Consider Traffic on This Trunk Secure* | When using both sRTP and TLS |
| Route Class Signaling Enabled* | Default |
| Use Trusted Relay Point* | Default |

- [x] PSTN Access
- [ ] 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**

- [x] Remote-Party-Id
- [x] Asserted-Identity

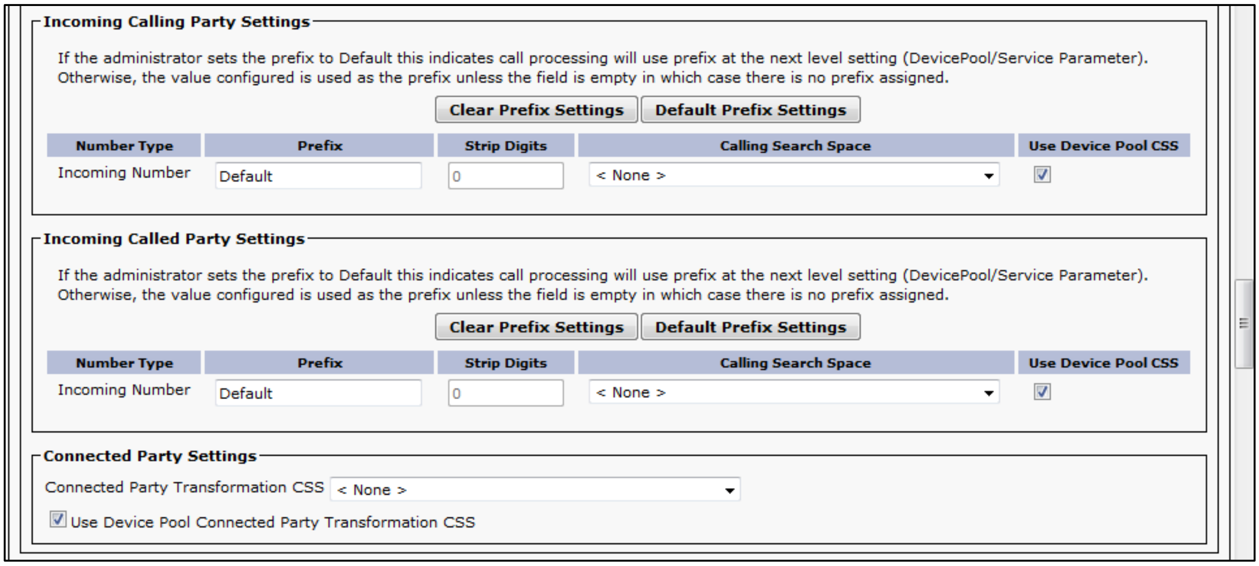
| | |
|---|---|
| Asserted-Type* | Default |
| SIP Privacy* | Default |

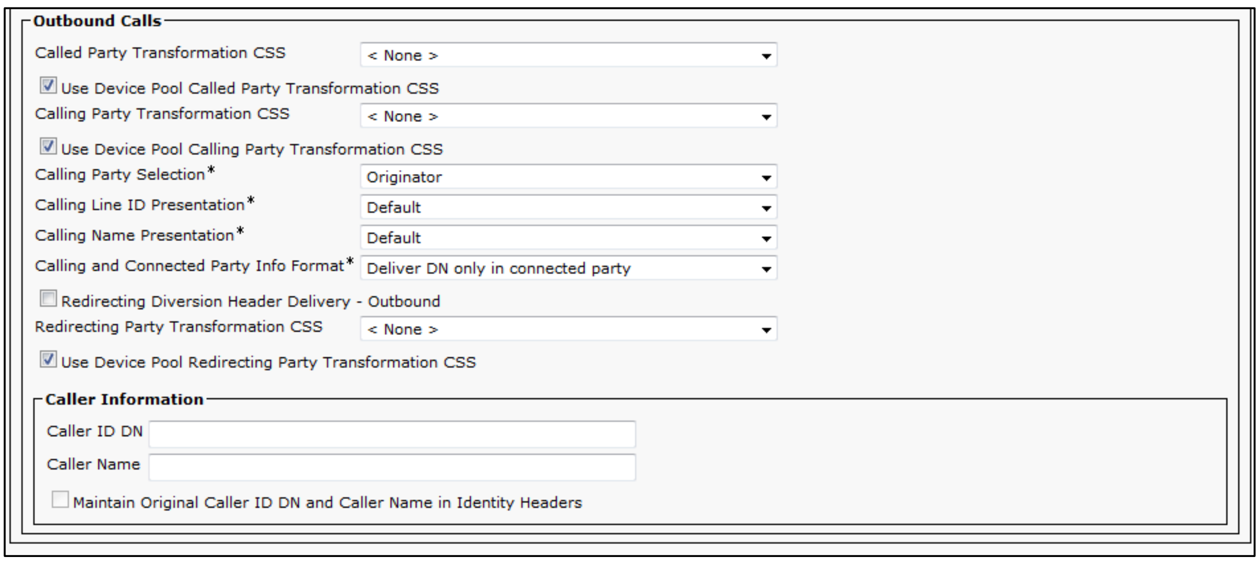
**Inbound Calls**

| | |
|---|---|
| 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

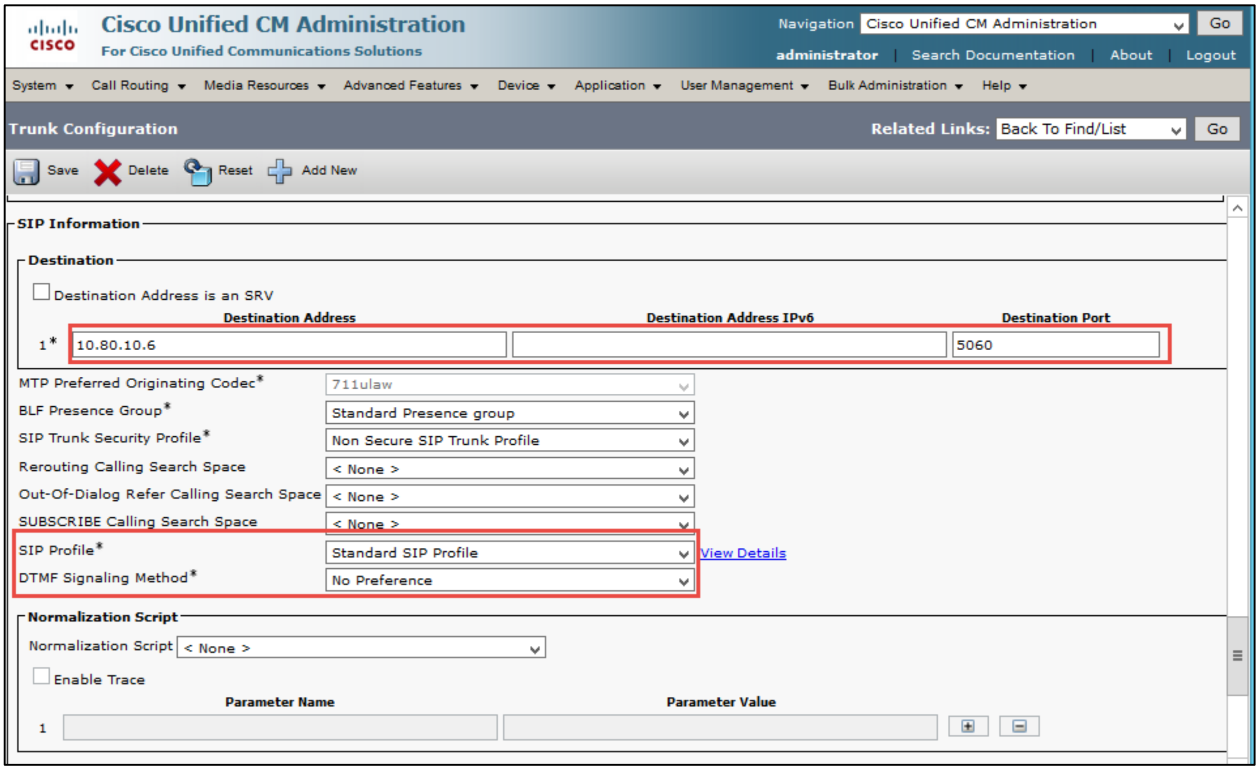
## Cisco UCM SIP Trunk to CUP Configuration (Continued)

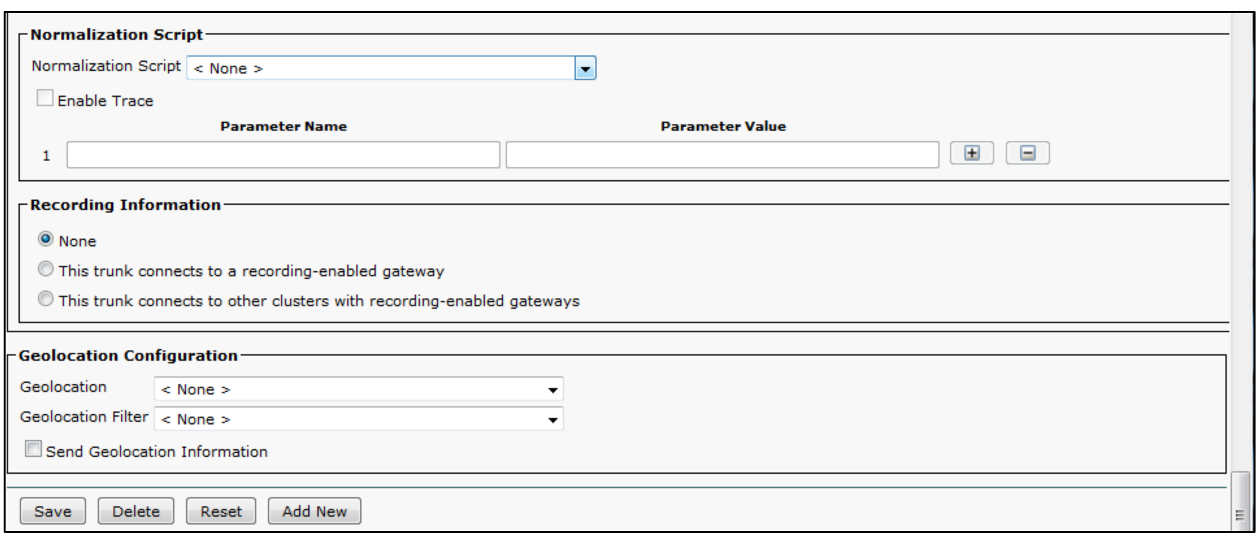


**Cisco UCM SIP Trunk to CUP Configuration (Continued)**

Set Destination Address = 10.80.10.6. This is used in this example.
Set SIP Trunk Security Profile*= Non Secure SIP Trunk Profile.
Set SIP Profile*= Standard SIP Profile.
Set DTMF Signaling Method*= No Preference.
All other values are default.



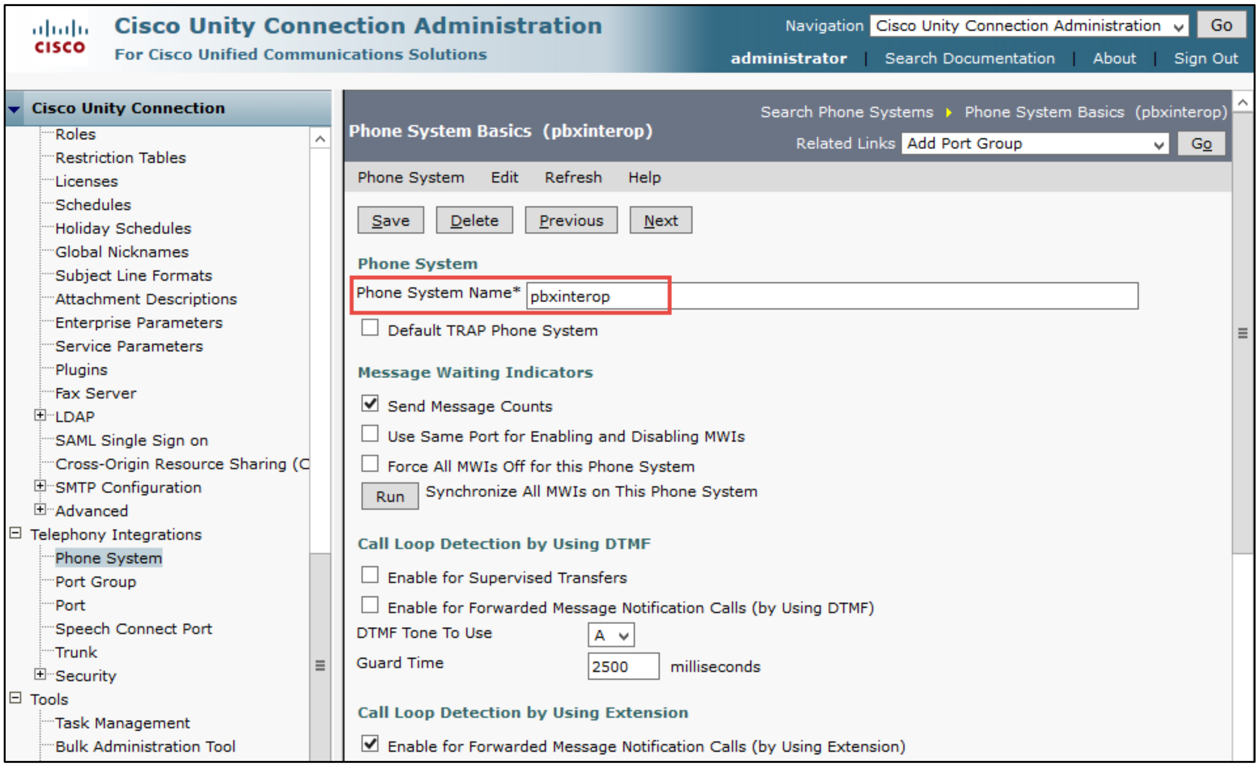



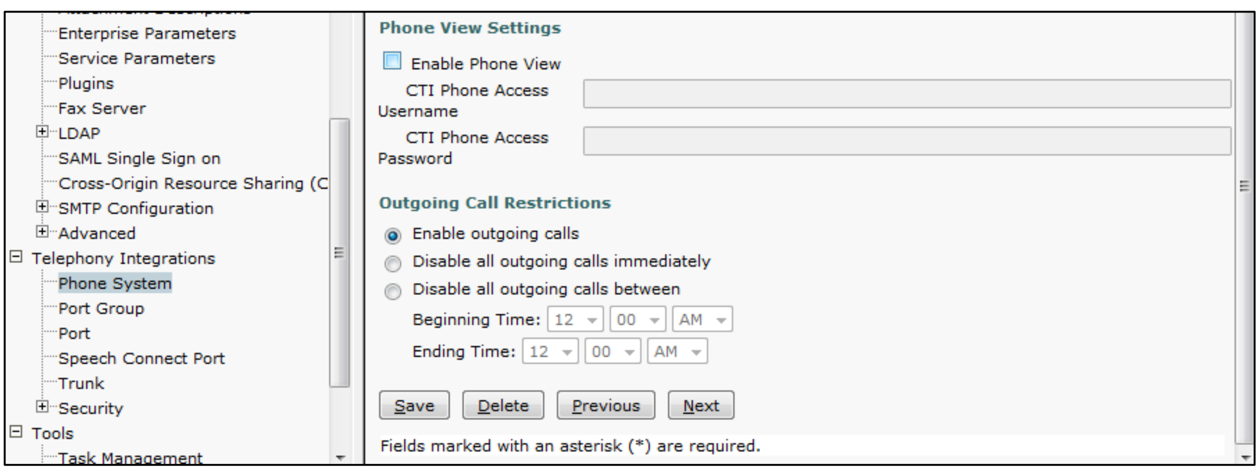
# Cisco Unity Connection

## Cisco Unity Connection Telephony Integration

**Navigation:** Telephony Integrations → Phone system

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



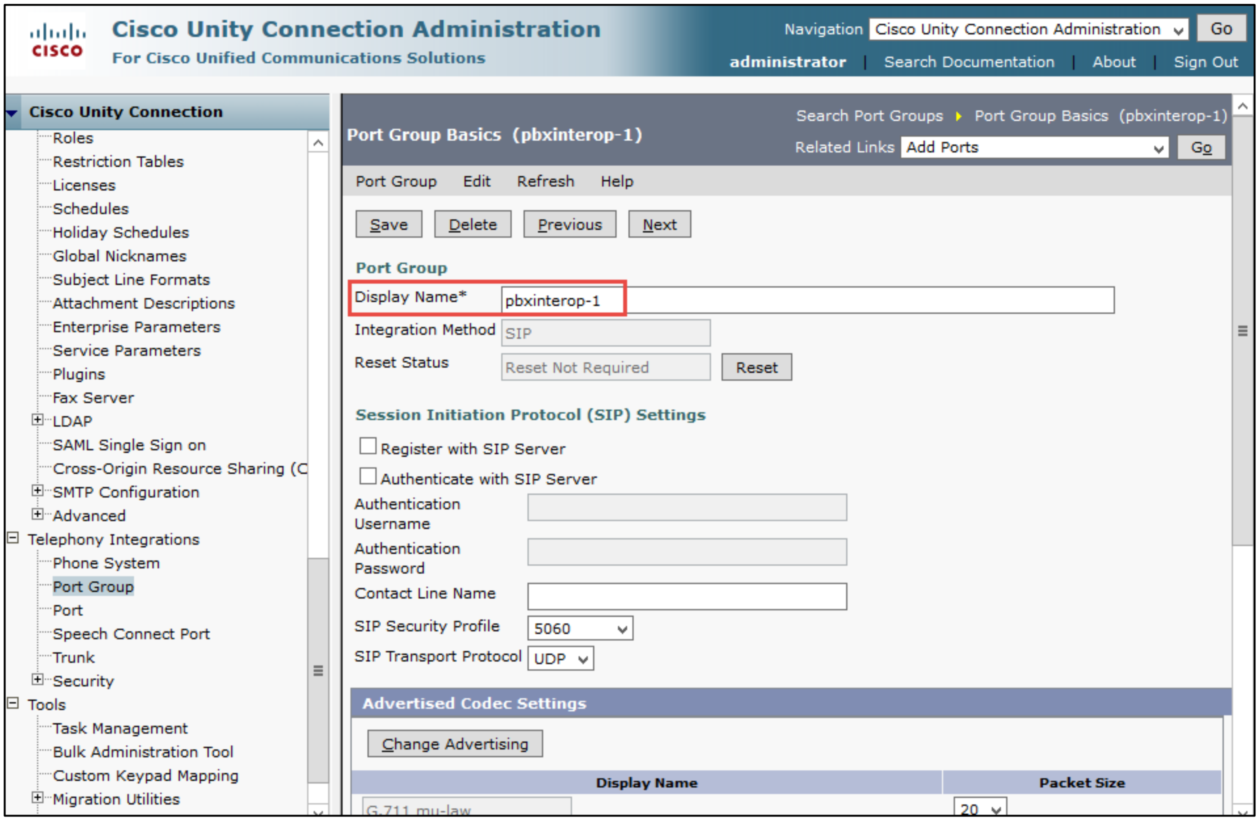



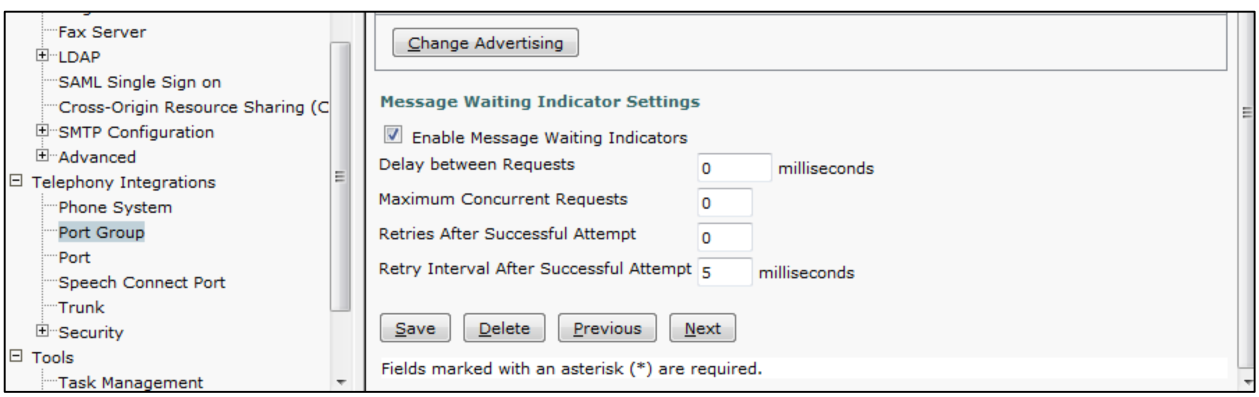
Port Group

**Navigation:** Telephony Integration → Port Group

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

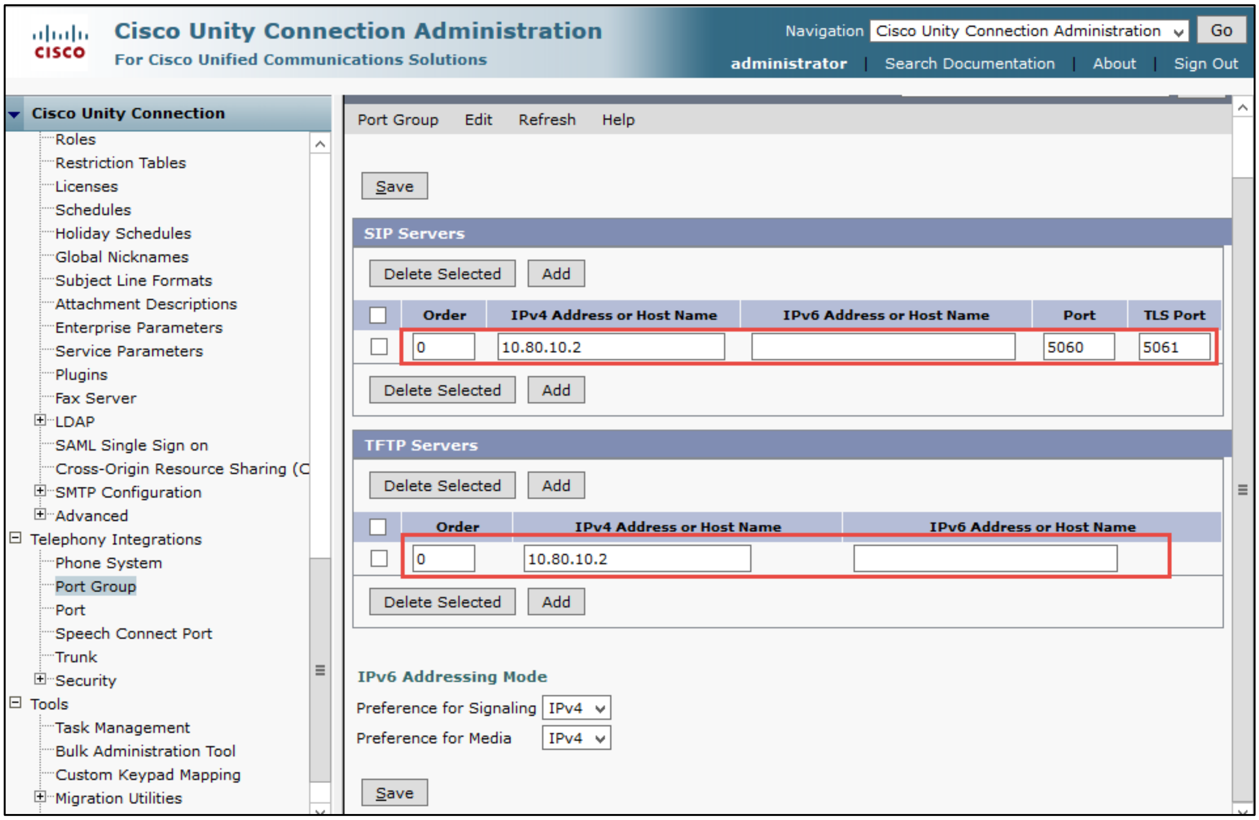
Check Register with SIP server



## Cisco Unity Telephony integration Configuration (Continued)

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



**Cisco Unity Telephony integration Configuration (Continued)**

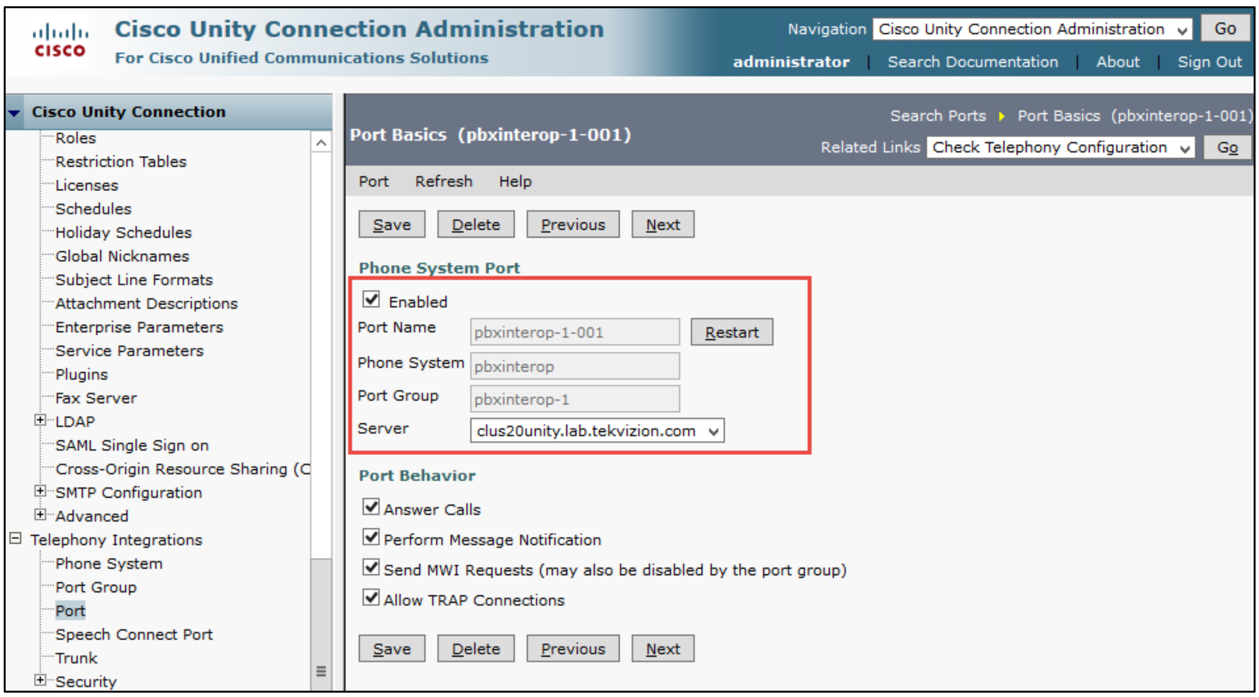
Port

Set Port Name = pbxinterop-1-001. This Name used for this example

Phone System = pbxinterop

Port Group = pbxinterop -1

Server = clus20unity.lab.tekvizion.com. This Name used for this example




## Cisco Unity Connection User Configuration

**Navigation:** Cisco Unity Connection → Users → Users

Set Alias*= 4001 This is used for this example.

Set First Name = This text is used to identify this User.

Set Last Name* = cisco. This is used for this example

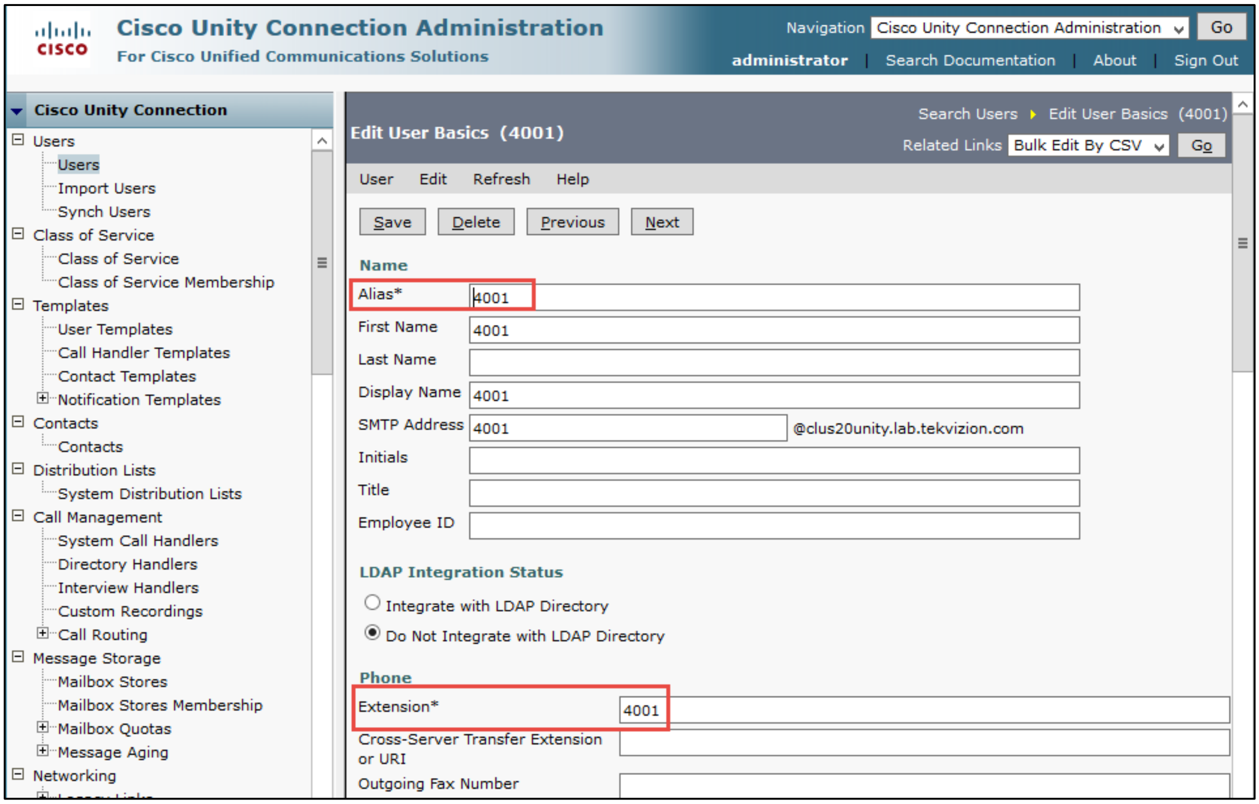
Set Display Name= 4001. This is used in this example.

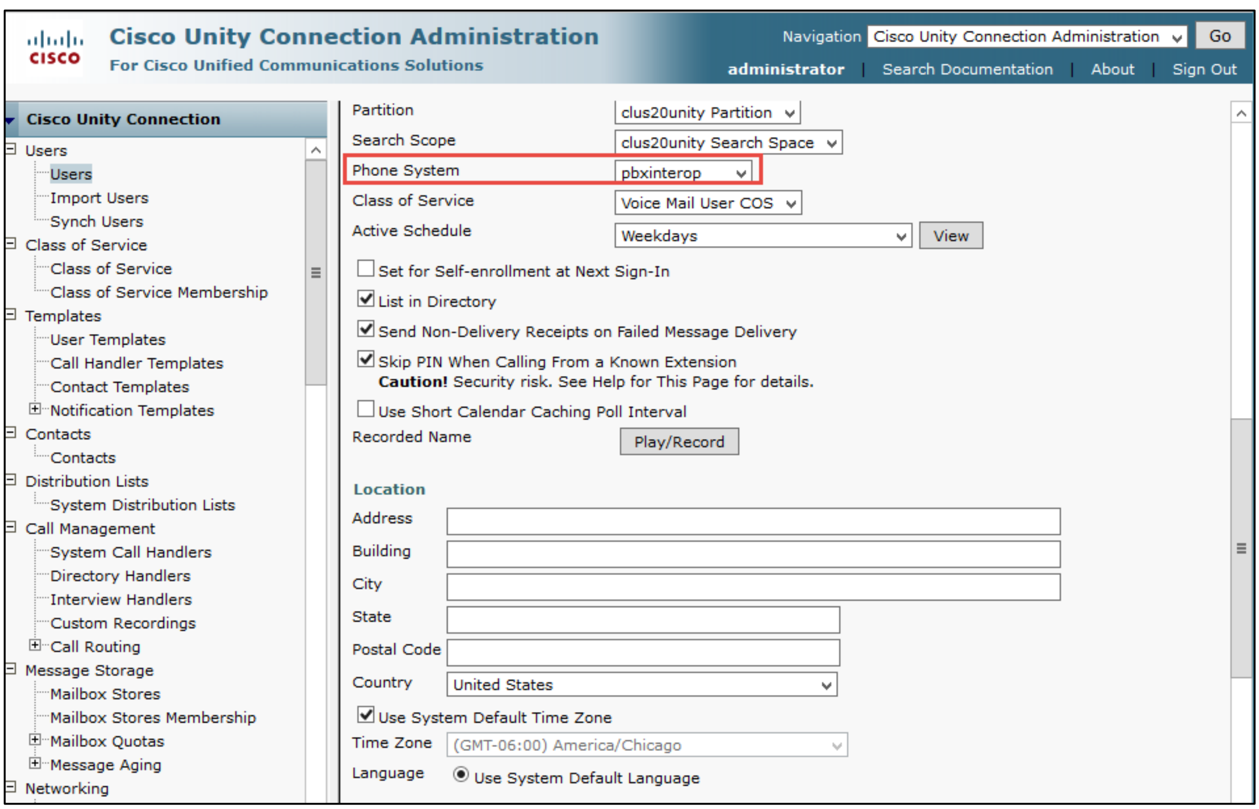
Set SMTP Address =4001. This is used in this example.

Set Phone System= SIP. This is used in this example.

All other values are default.

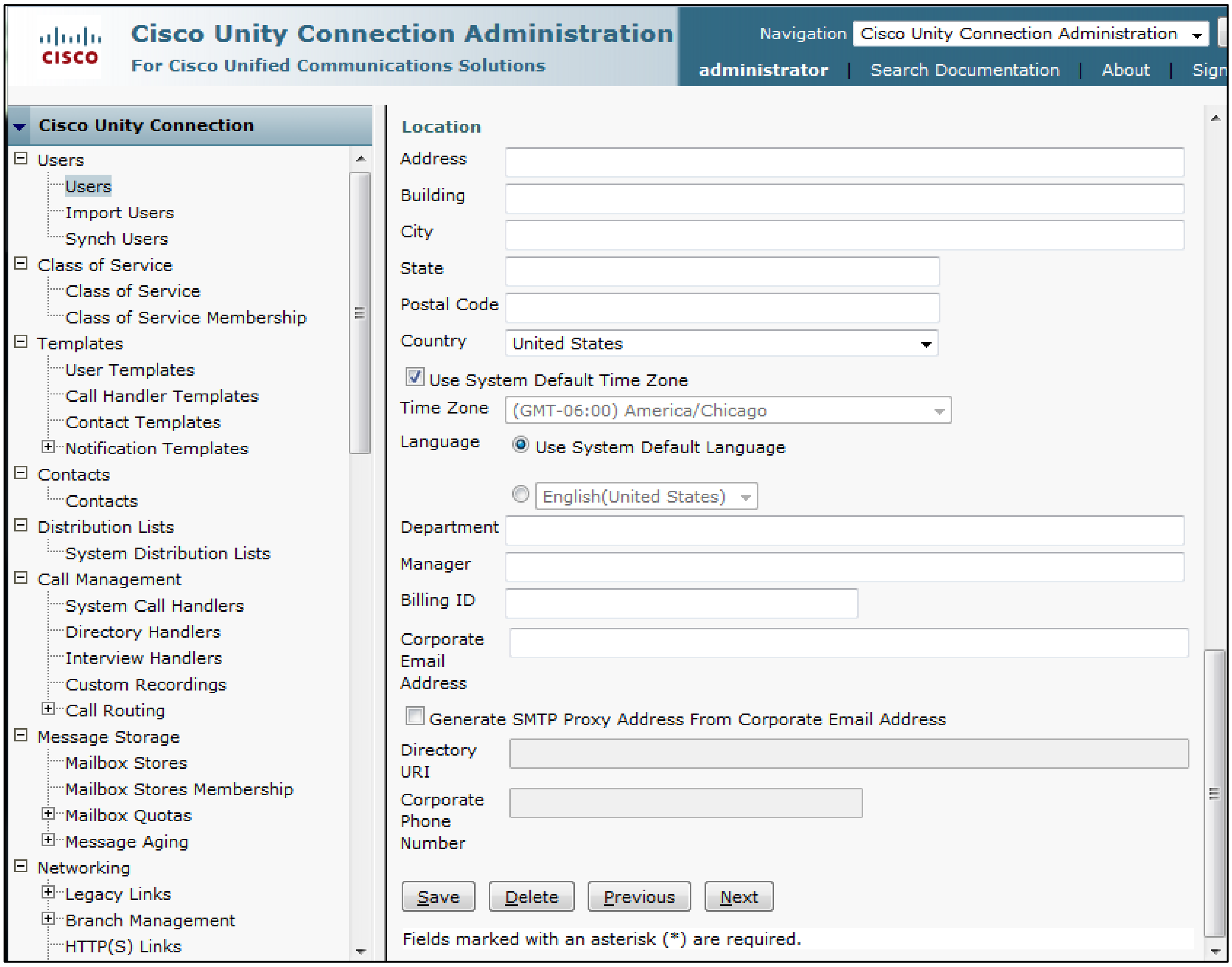
## Cisco Unity Connection User Configuration (Continued)



**Cisco Unity Connection User Configuration (Continued)**

All values are default.



## Acronyms

| Acronym | Definition |
|---|---|
| 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 |