

**AT&T IP Flexible Reach Service with Enhanced Features Using MIS / PNT or AT&T Virtual Private Network Transport with Cisco Unified Communications Manager v. 11.0 and Cisco UBE v. 11.1.0 on an ISR G2 Router with IPv4 SIP Interface**

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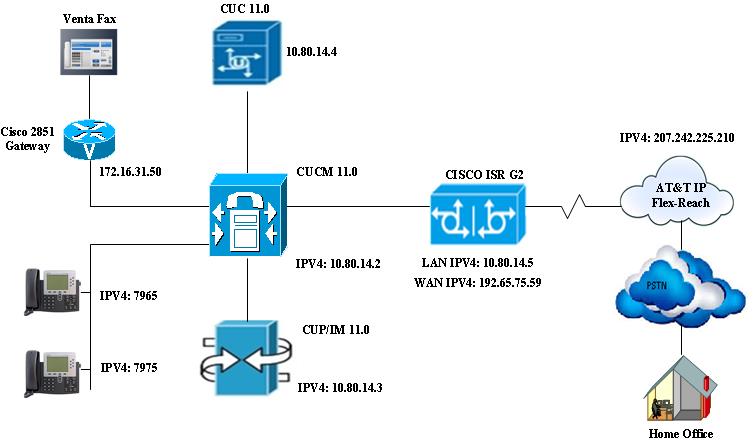
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# Introduction

Service Providers today, such as AT&T, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP Flexible Reach is a service provider offering that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity via either analog or T1 lines. A demarcation device between these services and customer owned services is recommended. The Cisco Unified Border Element (Cisco UBE) provides demarcation, security, interworking and session management services.

* This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 11.0, Cisco Unity Connection 11.0, Cisco Unified CM IM and Presence 11.0, Cisco Integrated Services Routers (ISR) Version 15.5(3) M1 with connectivity to AT&T’s IP Flex-Reach SIP trunk service. It also covers support and configuration example of Cisco Unity Connection (CUC) messaging integrated with Cisco Unified Communications Manager (Cisco UCM). The deployment model covered in this application note is Cisco Integrated Services Routers (ISR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.
* Testing was performed in accordance to AT&T’s IP Flexible Reach test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, CODEC negotiation, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), CISCO auto-attendant (BACD), fax G.711 and T38 (G3 and SG3 speeds), teleconferencing, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.
* The Cisco Unified Border Element function configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Integrated services router. The configurations described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying Cisco ISR to ensure these commands are set per each dial-peer required, to interoperate done AT&T SIP network.
* Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communication Manager with Cisco Unified Border Element components.

# Network Topology

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# Hardware Components

* UCS-C240 VMWare server running ESXi 5.5
* Cisco IP Phones. This solution was tested with Cisco 7965 & Cisco 9971 phones
* Cisco integrated Service Router G2 - Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory
* Processor board ID FTX1746AJCC 3 Gigabit Ethernet interfaces, 1 terminal line, 1 Virtual Private Network (VPN) Module, DRAM configuration is 64 bits wide with parity enabled, 255K bytes of non-volatile configuration memory

# Software Requirements

* Cisco UCM: System version: 11.0.1.10000-10, including Business Edition 6000 and Business Edition 7000.
* ISR: C2900 Software (C2900-UNIVERSALK9\_NPE-M), Version 15.5(3) M1, RELEASE SOFTWARE (fc2).
* Cisco UBE Software Release 11.1.0
* System image file is "flash:c2900-universalk9\_npe-mz.SPA.155-3.M1.bin"
* Cisco Unity Connection version: System version: 11.0.1.10000-10.
* Cisco Unified CM IM and Presence: System version: 11.0.1.10000-6.
* Cisco Jabber client version:11.0.0 Build 65527
* VentaFax client version: 7.3.233.582 I

# Features

## Features – Supported

* Basic Call using G.729 and G711
* Calling Party Number Presentation and Restriction
* Calling Name Presentation
* AT&T Advanced 8YY Call Prompter (8YY)
* Cisco UBE Delayed-Offer-to-Early-Offer conversion of an initial SIP INVITE without SDP
* Intra-site Call Transfer
* Intra-site Conference
* Call Hold and Resume
* Call Forward All, Busy and No Answer
* AT&T IP Teleconferencing
* Fax over G.711
* Fax using T.38
* Incoming DNIS Translation and Routing
* Outbound calls to AT&T’s IP and TDM networks
* Inbound calls from AT&T’s IP and TDM networks
* CPE voicemail managed service, leave and retrieve voice messages via incoming AT&T SIP trunk (Cisco Unity Connection)
* Auto-attendant transfer-to service (See Caveat section for details)
* Failover (From non-responsive SIP network to ATT SIP network)
* Inbound & Outbound Calls using Cisco Jabber
* Emergency and 411 calls were terminated to a voicemail platform in lab environment within AT&T for test
* RTCP

## Network Based Features - Supported

* Call forward (Unconditional, Busy, No Answer, Not reachable)
* Sequential Ringing
* Simultaneous Ringing

NOTE: Using the AT&T IP Flexible Reach Portal, provision TN(s) on the CPE with the Sequential Ring and simultaneous feature. Provisioning is self-explanatory. Please contact your AT&T representative, if you need help with the provisioning Network based feature.

## Features - Not Supported

* Cisco UCM Codec negotiation of G.722.1
* Network-Based Blind Call Transfer
* Network-Based Consultative Call Transfer

# Caveats

## Auto-Attendant

* The CUC auto-attendant feature was used to test attendant functionality using the default codec G711 for auto attendant prompts. G729 prompts can be used but was not tested.

## Hold/Resume & Music on Hold (MOH)

* Re-invites for hold/resume from PSTN network is potentially dependent on the carrier/network through which the call is traversing.

## Ring back Tone on Early Unattended Transfer

* Caller does not hear ring back tone when a call is transferred to PSTN user.

## PBX Based Call Forward Unconditional

* PBX Based Unconditional Call Forwarding test is temporarily blocked due to AT&T Flexible Reach network issue.

## SIP Provisional Acknowledgement/Early media

* To play early media sent by ATT, Cisco UCM needs to be enabled with PRACK if 1XX contains SDP on Cisco UCM SIP Profile.
* Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful early-media cut-through, the Cisco UCM to Cisco UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the Cisco UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”. The SIP Profile is found under Device>Device Settings>SIP Profile, This feature can be assigned on a per SIP trunk basis using SIP profiles. SIP PRACK provisioning on Cisco UCM 9.X and newer software versions is enabled under SIP Profile configuration page, while SIP PRACK support on Cisco UCM 7.X and older software versions is enabled under the Service Parameters configuration page.

## AT&T IP Teleconferencing (IPTC)

Following scenarios were not executed due to limitations on AT&T network

* IPTC - Hold & Resume
* IPTC - PBX-Based Attended Transfer
* IPTC - PBX-Based 3-way Call Conference

# Configuration Considerations

* To enable conference on AT&T IP Flexible Reach and Cisco UCM SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between end-points. See configuration section for details.
* Forwarded calls from Cisco UCM user to PSTN (out to AT&T’s IP Flexible Reach service), AT&T serviced areas require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption has been made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using Cisco UBE translation profile. This is because the Cisco UCM uses 4-digit extensions on Cisco UCM IP phones and it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP profile in Cisco UBE (See configuration section for details).
* Upon receiving inbound calls, AT&T SIP network will always have the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. Customers wishing to place/receive G.711-only calls must configure separate voice class codec on Cisco UBE with G.711 as the first choice.
* SIP Profiles may also be employed to advertise desired RTP payload packet size.
* “voice-class sip privacy id” needs to configure in Cisco UBE dial peer to make call From a CPE Phone to PSTN phone, Pass Calling Party Number (CPN), marked private.
* This test environment is not configured with Cisco UBE High Availability (HA)
* Cisco UCM sends a SIP UPDATE message to Cisco UBE for a call transfer. AT&T network does not support the SIP UPDATE message causing the Cisco UBE to timeout and the call transfer is not completed. As a workaround, SIP profile has been applied on the Cisco UBE to remove UPDATE from the allowed headers (See configuration section for details).

# Emergency 911/E911 Services Limitations and Restrictions

* Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor
* While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at <http://new.serviceguide.att.com>. Such circumstances include, but are not limited to, relocation of the end user’s CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power and delays that may occur in updating the Customer’s location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions

# ISR Configuration

ATT-IPV4#sh version

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9\_NPE-M), Version 15.5(3)M1, RELEASE SOFTWARE (fc2)

Technical Support: http://www.cisco.com/techsupport

Copyright (c) 1986-2015 by Cisco Systems, Inc.

Compiled Mon 16-Nov-15 19:25 by prod\_rel\_team

ROM: System Bootstrap, Version 15.0(1r)M16, RELEASE SOFTWARE (fc1)

ATT-IPV4 uptime is 5 weeks, 1 day, 11 minutes

System returned to ROM by reload at 12:06:12 UTC Tue Jan 5 2016

System image file is "flash:c2900-universalk9\_npe-mz.SPA.155-3.M1.bin"

Last reload type: Normal Reload

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United

States and local country laws governing import, export, transfer and

use. Delivery of Cisco cryptographic products does not imply

third-party authority to import, export, distribute or use encryption.

Importers, exporters, distributors and users are responsible for

compliance with U.S. and local country laws. By using this product you

agree to comply with applicable laws and regulations. If you are unable

to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:

http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to

export@cisco.com.

Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory.

Processor board ID FTX1746AJCB

3 Gigabit Ethernet interfaces

1 terminal line

DRAM configuration is 64 bits wide with parity enabled.

255K bytes of non-volatile configuration memory.

250880K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

-------------------------------------------------

Device# PID SN

-------------------------------------------------

\*1 CISCO2921/K9 FTX1746AJCB

Suite License Information for Module:'c2900'

--------------------------------------------------------------------------------

Suite Suite Current Type Suite Next reboot

--------------------------------------------------------------------------------

FoundationSuiteK9\_npe None None None

securityk9\_npe

datak9

AdvUCSuiteK9 None None None

uck9

cme-srst

cube

Technology Package License Information for Module:'c2900'

------------------------------------------------------------------------

Technology Technology-package Technology-package

Current Type Next reboot

------------------------------------------------------------------------

ipbase ipbasek9 Permanent ipbasek9

security None None None

uc uck9 Permanent uck9

data None None None

Configuration register is 0x2102

ATT-IPV4#sh run

Building configuration...

Current configuration : 11605 bytes

!

! Last configuration change at 23:33:01 UTC Tue Feb 9 2016 by cisco

!

version 15.5

service timestamps debug datetime msec

service timestamps log datetime msec

no service password-encryption

service sequence-numbers

!

hostname ATT-IPV4

!

boot-start-marker

boot system flash c2900-universalk9\_npe-mz.SPA.155-3.M1.bin

boot-end-marker

!

aqm-register-fnf

!

logging queue-limit 1000000000

logging buffered 30000000

logging rate-limit 10000

no logging console

no logging monitor

enable secret 4 Pe0NhiWw5IXZpE.k5VhTSCoGPcuVeRyrer9kEPz20Z6

!

no aaa new-model

ethernet lmi ce

!

!

!

!

!

!

!

!

!

!

!

!

no ip domain lookup

ip cef

no ipv6 cef

multilink bundle-name authenticated

!

!

!

!

!

cts logging verbose

!

!

voice-card 0

dspfarm

dsp services dspfarm

!

!

!

voice service voip

no ip address trusted authenticate

address-hiding**[[1]](#footnote-1)**

mode border-element**[[2]](#footnote-2)**

allow-connections sip to sip**[[3]](#footnote-3)**

redirect ip2ip

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no fax-relay sg3-to-g3

sip

header-passing

error-passthru[[4]](#footnote-4)

early-offer forced[[5]](#footnote-5)

midcall-signaling passthru[[6]](#footnote-6)

privacy-policy passthru[[7]](#footnote-7)

g729 annexb-all

!

voice class codec 2

codec preference 1 g711ulaw

codec preference 2 g729r8 bytes 30

!

voice class codec 3

codec preference 1 g711ulaw

!

voice class codec 1[[8]](#footnote-8)

codec preference 1 g729r8 bytes 30

codec preference 2 g711ulaw

!

!

voice class sip-profiles 1

response ANY sip-header Allow-Header modify "UPDATE," ""

request INVITE sip-header Diversion modify "<sip:(.\*)@(.\*)>" "<sip:732320\1@\2>"[[9]](#footnote-9)

request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"[[10]](#footnote-10)

response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"

request INVITE sdp-header Audio-Attribute add "a=ptime:30"[[11]](#footnote-11)

!

!

!

license udi pid CISCO2921/K9 sn FTX1746AJCB

hw-module pvdm 0/0

!

!

!

username cisco privilege 15 password 0 cisco

!

redundancy

!

!

!

!

interface Embedded-Service-Engine0/0

no ip address

shutdown

!

interface GigabitEthernet0/0[[12]](#footnote-12)

description WAN to ATT

ip address 192.65.79.59 255.255.255.224

duplex auto

speed auto

!

interface GigabitEthernet0/1[[13]](#footnote-13)

description LAN Interface

ip address 10.80.14.5 255.255.0.0[[14]](#footnote-14)

duplex auto

speed auto

!

interface GigabitEthernet0/2

no ip address

shutdown

duplex auto

speed auto

!

ip forward-protocol nd

!

no ip http server

no ip http secure-server

!

ip route 0.0.0.0 0.0.0.0 192.65.79.33

ip route 10.64.0.0 255.255.0.0 10.80.14.1

ip route 10.80.0.0 255.255.0.0 10.80.14.1

ip route 172.16.0.0 255.255.0.0 10.80.14.1

!

!

!

!

control-plane

!

!

!

!

!

!

mgcp behavior rsip-range tgcp-only

mgcp behavior comedia-role none

mgcp behavior comedia-check-media-src disable

mgcp behavior comedia-sdp-force disable

!

mgcp profile default

!

!

!

!

dial-peer voice 100 voip

description "Outgoing To AT&T .IP PBX facing side"

session protocol sipv2

incoming called-number [2-9]T

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/1

voice-class sip bind media source-interface GigabitEthernet0/1

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!

dial-peer voice 101 voip[[15]](#footnote-15)

description "Outgoing To AT&T"-AT&T facing side

destination-pattern [2-9]T

session protocol sipv2[[16]](#footnote-16)

session target ipv4:207.242.225.210

voice-class codec 1[[17]](#footnote-17)

voice-class sip asymmetric payload full[[18]](#footnote-18)

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru[[19]](#footnote-19)

voice-class sip early-offer forced

voice-class sip profiles 1[[20]](#footnote-20)

voice-class sip bind control source-interface GigabitEthernet0/0[[21]](#footnote-21)

voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte[[22]](#footnote-22)

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none[[23]](#footnote-23)

no vad

!

dial-peer voice 400 voip

description " Incoming AT&T to IP-PBX AT&T facing side "

huntstop

session protocol sipv2

incoming called-number [37][13][24]320435.

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip privacy-policy passthru

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/0

voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!

dial-peer voice 401 voip

description " Incoming AT&T to IP-PBX - IP-PBX facing side "

huntstop

destination-pattern [37][13][24].......

session protocol sipv2

session target ipv4:10.80.14.2

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip privacy-policy passthru

voice-class sip early-offer forced

voice-class sip bind control source-interface GigabitEthernet0/1

voice-class sip bind media source-interface GigabitEthernet0/1

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!

dial-peer voice 600 voip

description "Network Feature"

destination-pattern \*..

session protocol sipv2

session target ipv4:207.242.225.210

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip early-offer forced

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/0

voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!

dial-peer voice 300 voip

description " Int'l calls to AT&T - AT&T facing side "

destination-pattern 011T

session protocol sipv2

session target ipv4:207.242.225.210

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip early-offer forced

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/0

voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte

no vad

!

dial-peer voice 2002 voip

description "Outgoing To AT&T"-AT&T facing side

destination-pattern 7323204...

session protocol sipv2

session target ipv4:207.242.225.210

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip early-offer forced

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/0

voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!

dial-peer voice 2004 voip

description "Outgoing To AT&T"-AT&T facing side

destination-pattern 8772888362

session protocol sipv2

session target ipv4:207.242.225.210

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip early-offer forced

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/0

voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!

dial-peer voice 2005 voip

description "Outgoing To AT&T"-AT&T facing side

destination-pattern 911

session protocol sipv2

session target ipv4:207.242.225.210

voice-class codec 1

voice-class sip asymmetric payload full

voice-class sip asserted-id pai

voice-class sip privacy-policy passthru

voice-class sip early-offer forced

voice-class sip profiles 1

voice-class sip bind control source-interface GigabitEthernet0/0

voice-class sip bind media source-interface GigabitEthernet0/0

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

!

gateway

media-inactivity-criteria all

timer receive-rtcp 5

timer receive-rtp 86400

!

sip-ua

no remote-party-id

retry invite 2

timers expires 1800000

!

gatekeeper

shutdown

!

line con 0

logging synchronous

line aux 0

line 2

no activation-character

no exec

transport preferred none

transport output pad telnet rlogin lapb-ta mop udptn v120 ssh

stopbits 1

line vty 0 4

exec-timeout 960 0

logging synchronous

login local

transport input all

!

scheduler allocate 20000 1000

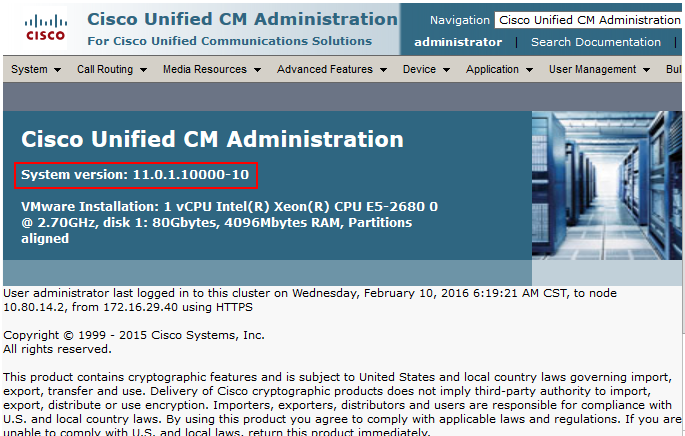
!

end

# Cisco UCM Configuration

The configuration screen shots shows general over view of lab configuration for this interoperability testing.

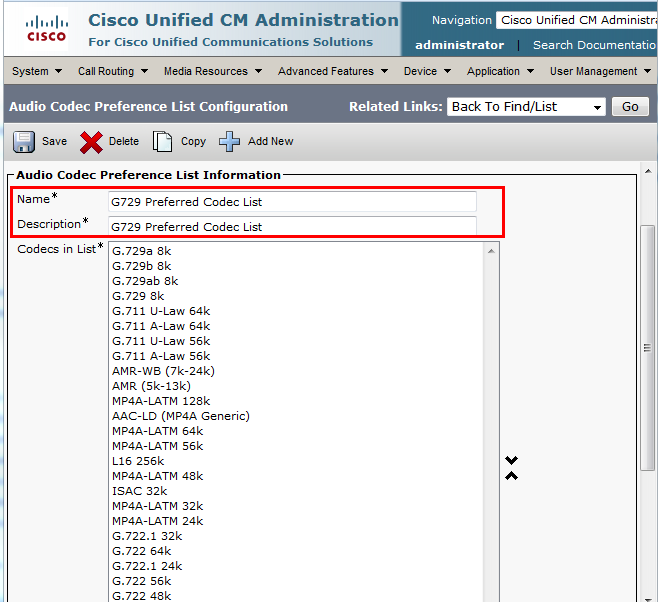
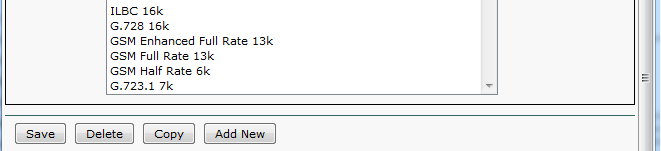
## Cisco UCM Version



## Cisco UCM Audio Codec Preference List

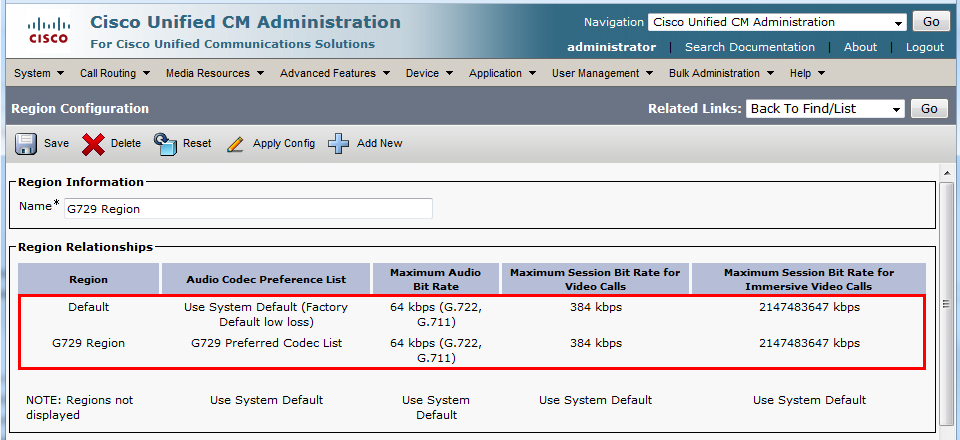
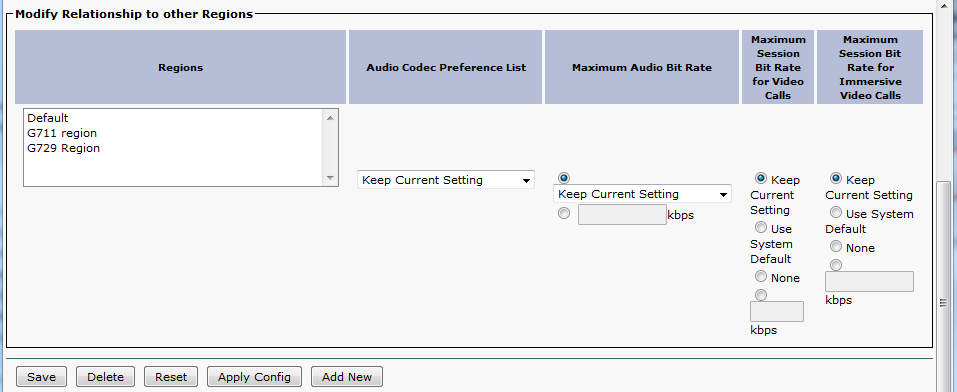
**Navigation Path:** System 🡪 Region Information 🡪 Audio codec preference list

Cisco UCM 11.0 has a feature called Audio Codec Preference List. This feature allows to configure the order of audio codec preference both for Inter and Intra Region calls. Audio Codec Preference list is assigned to the Region used by the Device Pool for Phones and by Conference Bridges. Based on user requirement, different codec regions can be assigned as their first choice codec with this configuration for inbound calls as well as conferences initiated by Cisco IP phones. Audio codec preference for outbound calls is determined by Cisco UBE (via configuration of voice-class codec)

## Cisco UCM Region Configuration

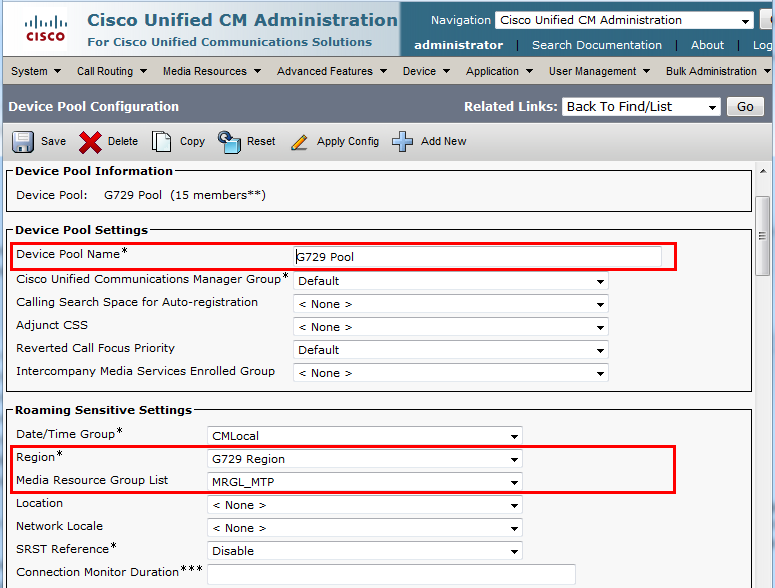
**Navigation Path:** System 🡪 Region Information 🡪 Region

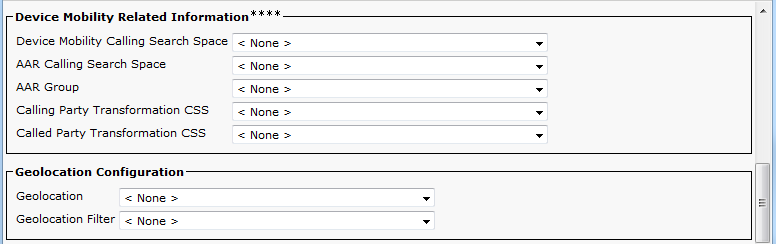
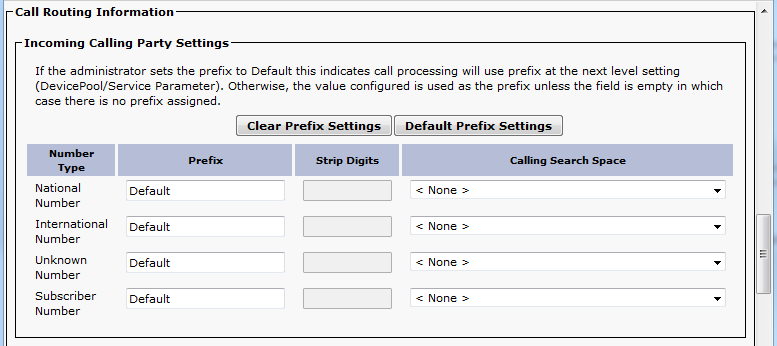
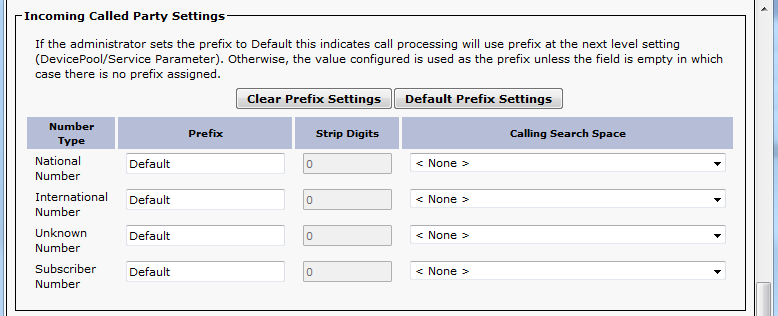
## Device Pool Configuration

**Navigation Path:** System 🡪 Device Pool

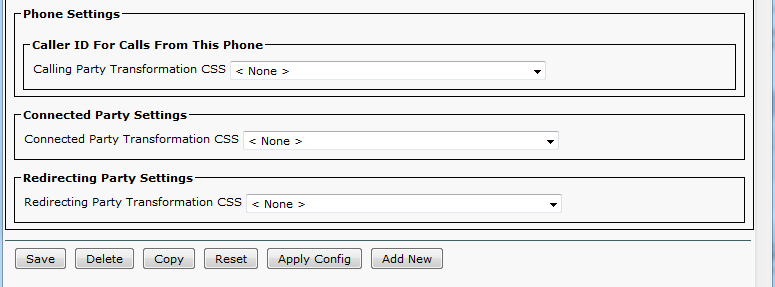
“G729” Device Pool is configured for testing the interoperability. No special consideration needs to be taken when configuring the Device Pools. Optionally, a Media Resource Group List can be added to the Device Pools, if needed, to assign selected Media Resources (Conference Bridges, Transcoders, MoH servers, Annunciators) to devices.

Device Pool Configuration (continued…)

Device Pool Configuration (continued…)



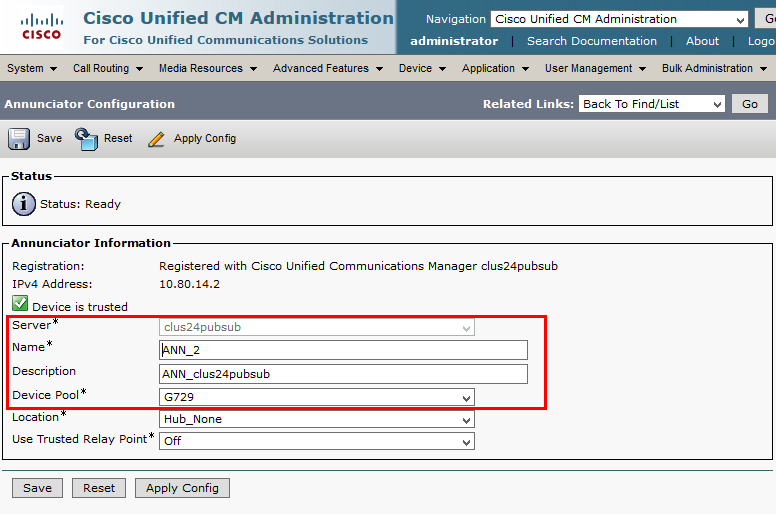
## Annunciator Configuration

**Navigation:** Media Resource 🡪 Annunciator

Set Name\* = ANN\_2.

Set Description = ANN\_clus24pubsub. This is used for this example

Set Device Pool\* = G729.



## Conference Bridge Configuration

**Navigation:** Media Resources 🡪 Conference Bridge

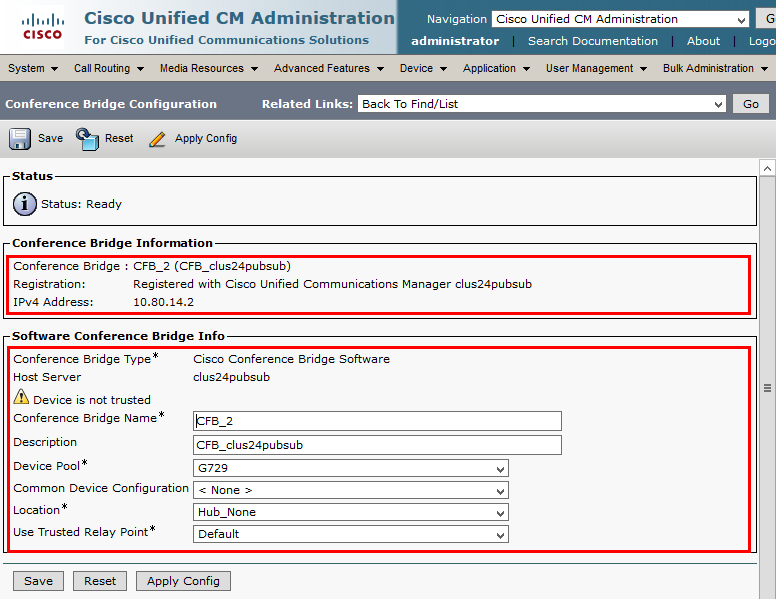
Set Conference Bridge Type\* = Cisco Conference Bridge Software.

Set Host Server = clus24pubsub. This is used for this example.

Set Conference Bridge Name\* = CFB\_2.

Set Description = CFB\_clus24pubsub. This is used in this example.

Set Device Pool\* = G729.



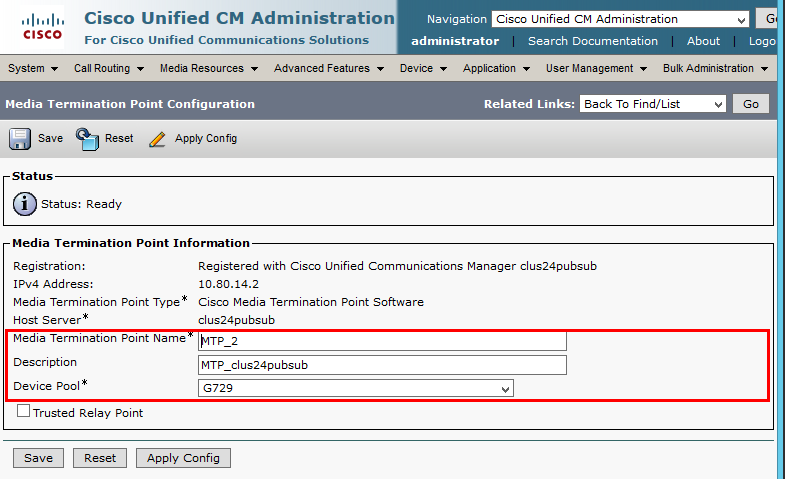
## Media Termination Point Configuration

**Navigation:** Media Resource🡪Media Termination Point

Set Media Termination Point Name\* = MTP\_2

Set Description = MTP\_clus24pubsub. This is used for this example

Set Device pool\* = G729



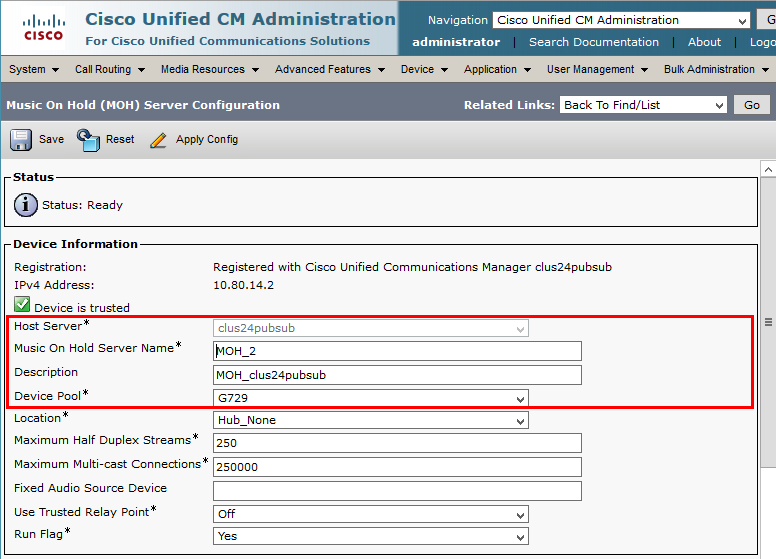
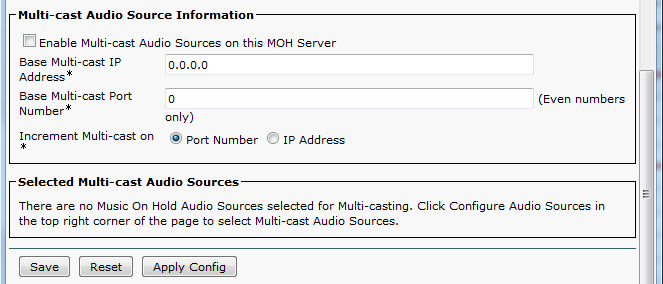
## Music on Hold Server Configuration

**Navigation:** Media Resources 🡪 Music on Hold Server

Set Music on Hold Server Name\* = MOH\_2.

Set Description = MOH\_clus24pubsub. This is used for this example.

Set Device Pool\* = G729

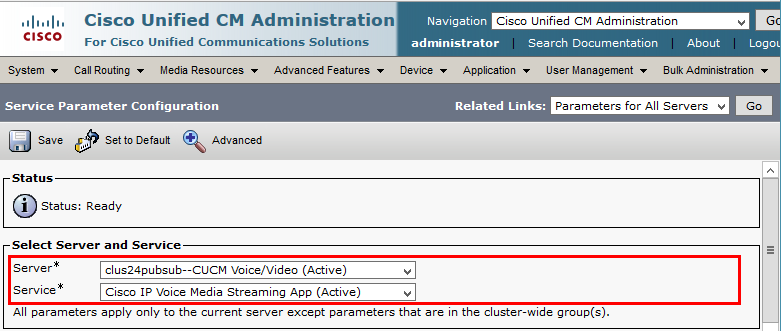
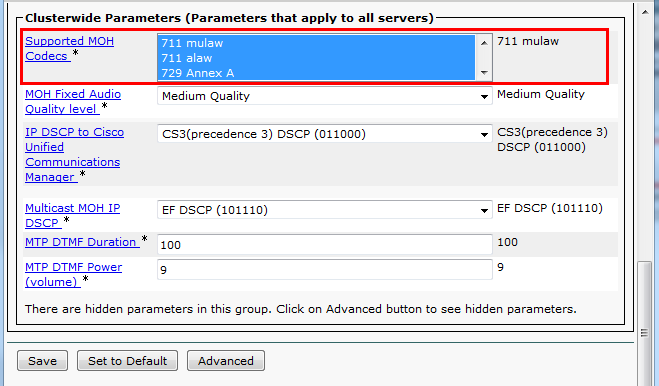
 

## Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

**Navigation:** System 🡪 Service Parameter

Note: Make sure codecs G.729 Annex A and G.711 mulaw are configured in parameter Supported MOH Codecs.

Select Server\* = clus24pubsub--CUCM Voice/Video (Active). This is used in this example.

Select Service\* = Cisco IP Voice Media Streaming App (Active 

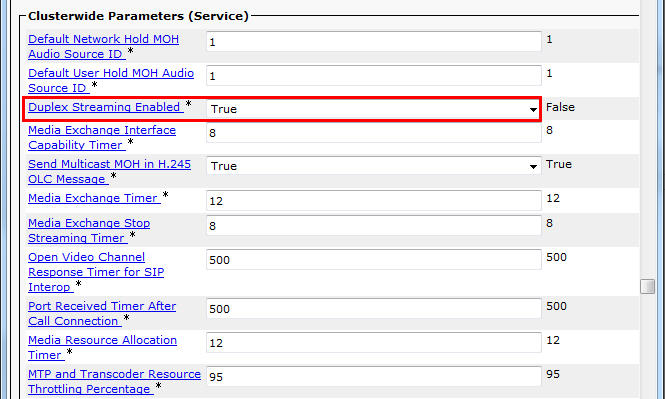
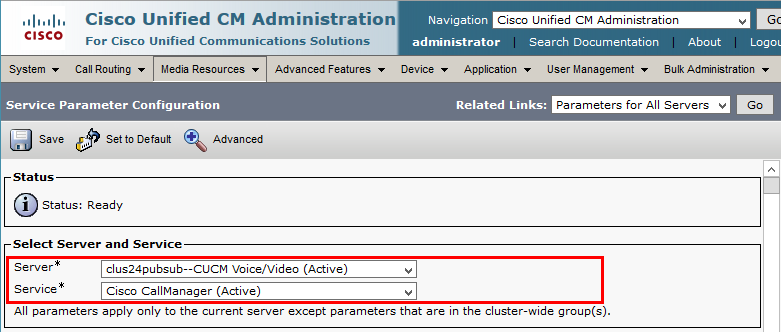
## Music on Hold Service (Duplex Streaming) Parameter Settings

**Navigation:** System 🡪 Service Parameter

Select Server\* = clus24pubsub--CUCM Voice/Video (Active). This is used in this example.

Select Service\* = Cisco CallManager (Active).

Select Duplex Streaming Enabled \* = True



## Media Resource Group Configuration

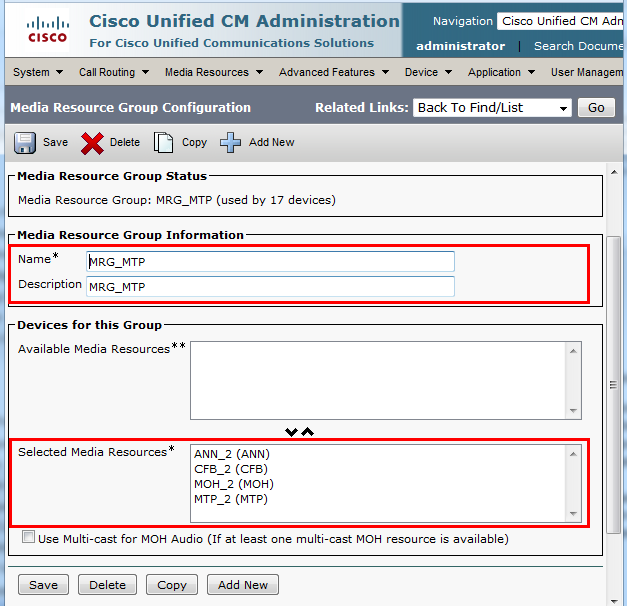
**Navigation Path:** Media Resources 🡪 Media Resources group

The Media Resource Group (MRG) contains media resources, such as Conference Bridge, Transcoder, MoH server and Annunciator. It will be assigned to a Media Resource Group List (MRGL) which is used to allocate media resources to groups of devices through Device Pools, or individually by configuring a valid MRGL at the device configuration page.

Set Name\*= MRG\_MTP - This is used for this example.

Set Description = MRG\_MTP - This text is used to define this Media Resource Group List.

Set all Resources in the selected Media Resources Box.

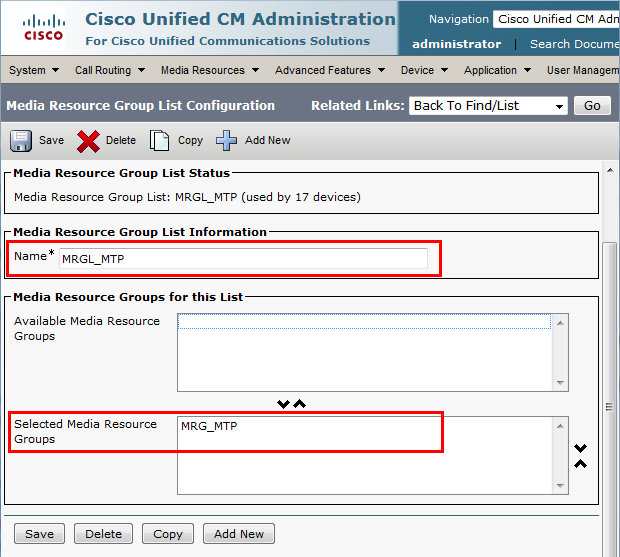


## Media Resource Group List Configuration

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

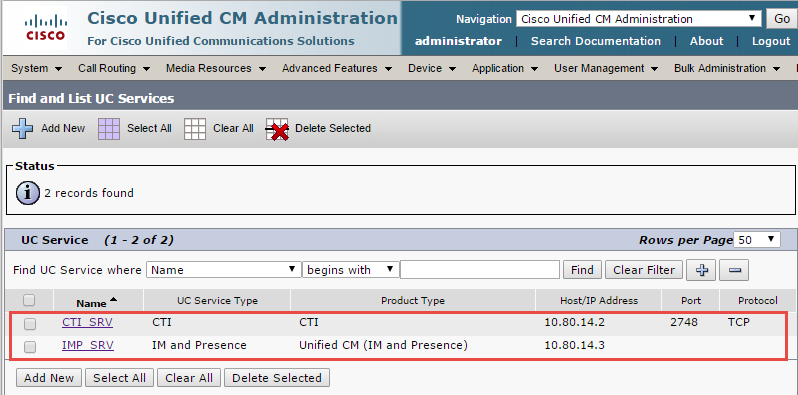
Set Name = MRGL\_MTP.

Set selected Media Resource Groups = MRG\_MTP.



## UC Service Configuration

**Navigation:** User Management 🡪 User Settings 🡪 UC Service



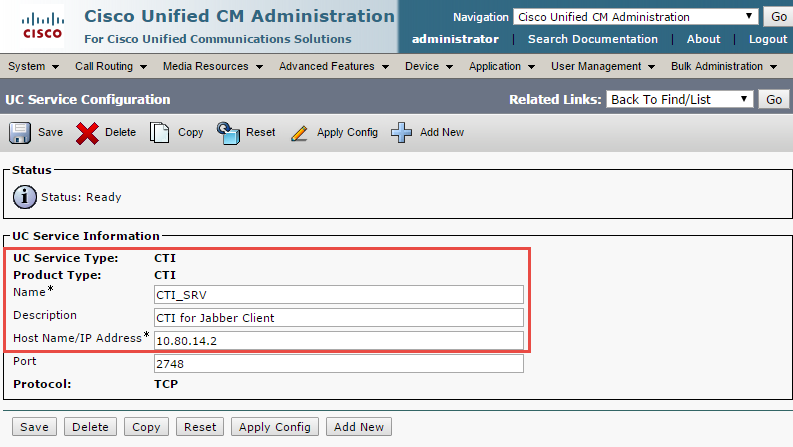
UC Service Configuration (Contd…)

Select UC Service Type: = CTI

Set Name\* = CTI\_SRV. This is used in this example.

Set Description = CTI for Jabber Clients. This is used in this example.

Set Host Name/IP Address\* = 10.80.14.2 (Cisco UCM Address)



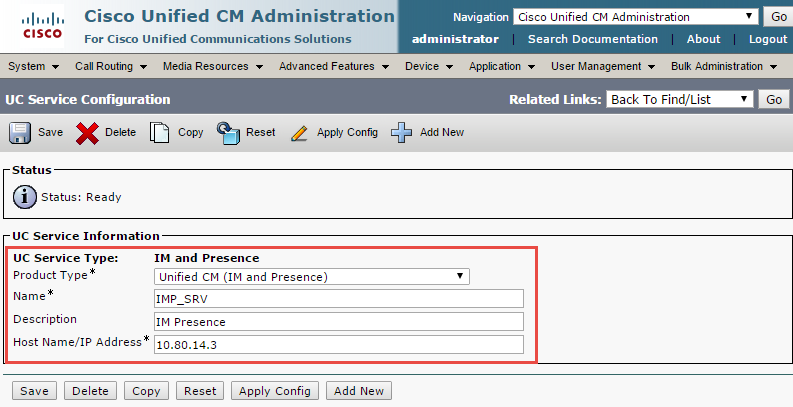
UC Service Configuration (Contd…)

Select UC Service Type: = IM and Presence

Set Name\* = IMP\_SRV. This is used in this example.

Set Description = IM Presence. This is used in this example.

Set Host Name/IP Address\* = 10.80.14.3 (Cisco UCM IM & Presence IP Address)



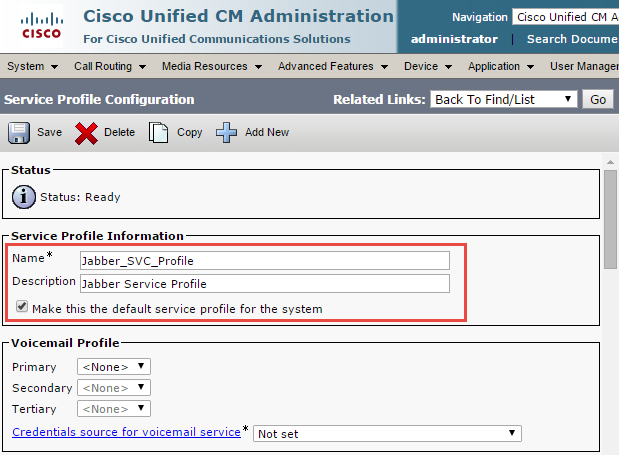
## Service Profile Configuration

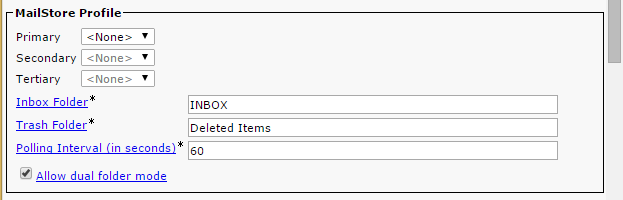
**Navigation:** User Management 🡪 User Settings 🡪 Service Profile

Set Name\* = Jabber\_SVC\_Profile. This is used in this example.

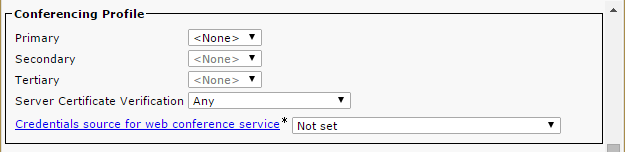
Set Description = Jabber Service Profile. This is used in this example.

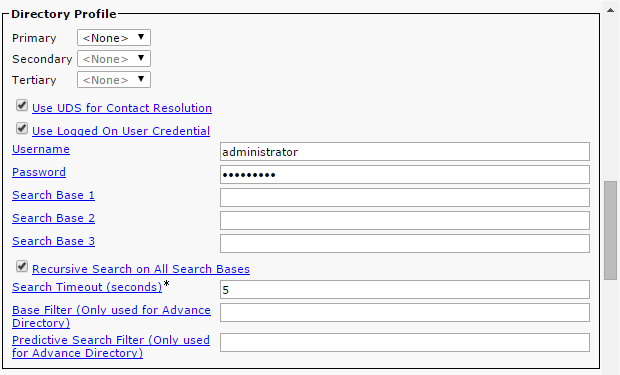
Check - Make this the default service profile for the system.

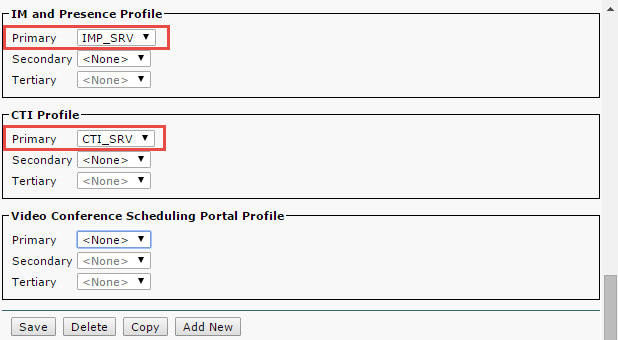




Service Profile Configuration (Contd…)







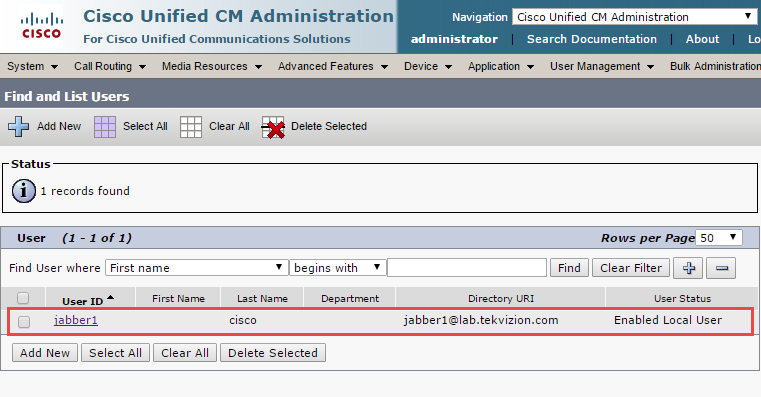
## End User Configuration

**Navigation:** User Management 🡪 End User

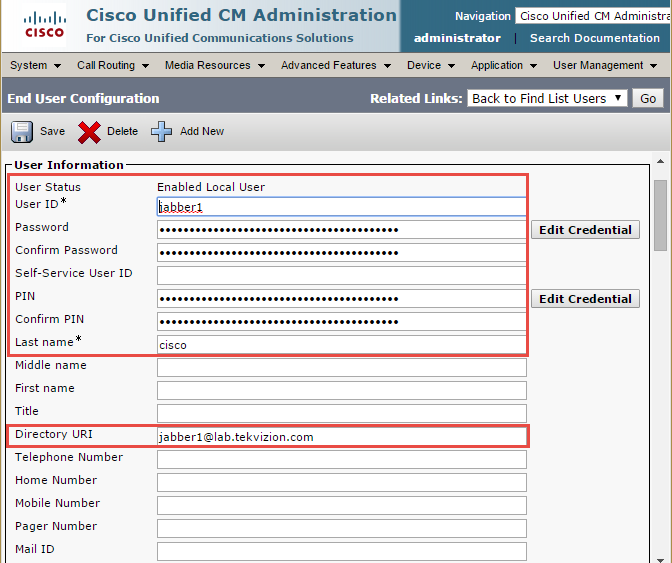
Set User ID\* = jabber1 – This is used in this example.

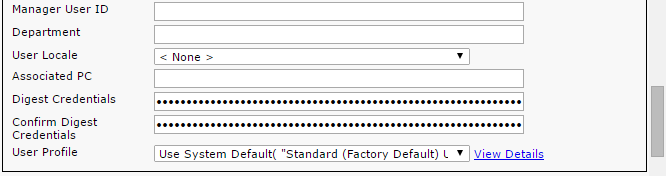
Set Password = Password for profile.

Set Directory URI = [jabber1@lab.tekvizion.com](mailto:jabber1@lab.tekvizion.com).

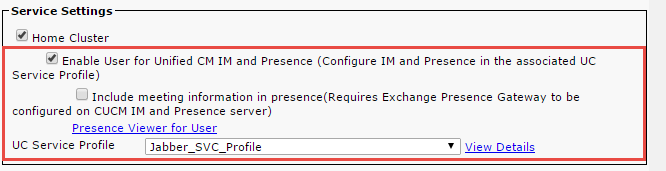


End User Configuration (continued…)

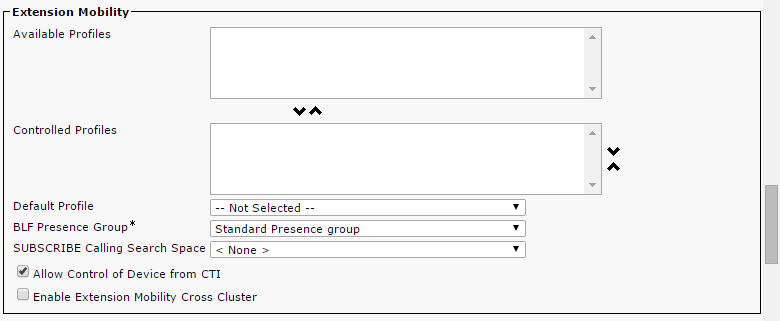




End User Configuration (continued…)

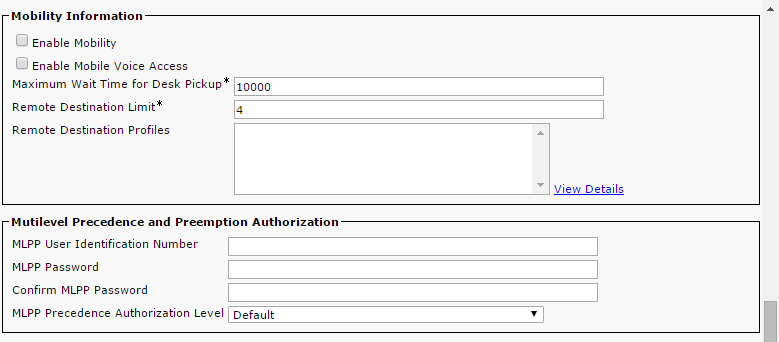


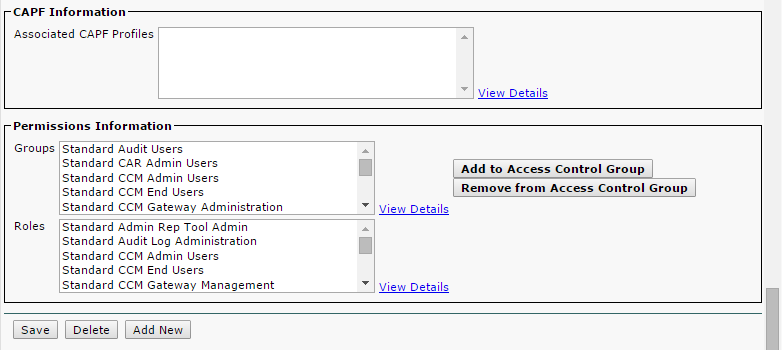






End User Configuration(continued… )





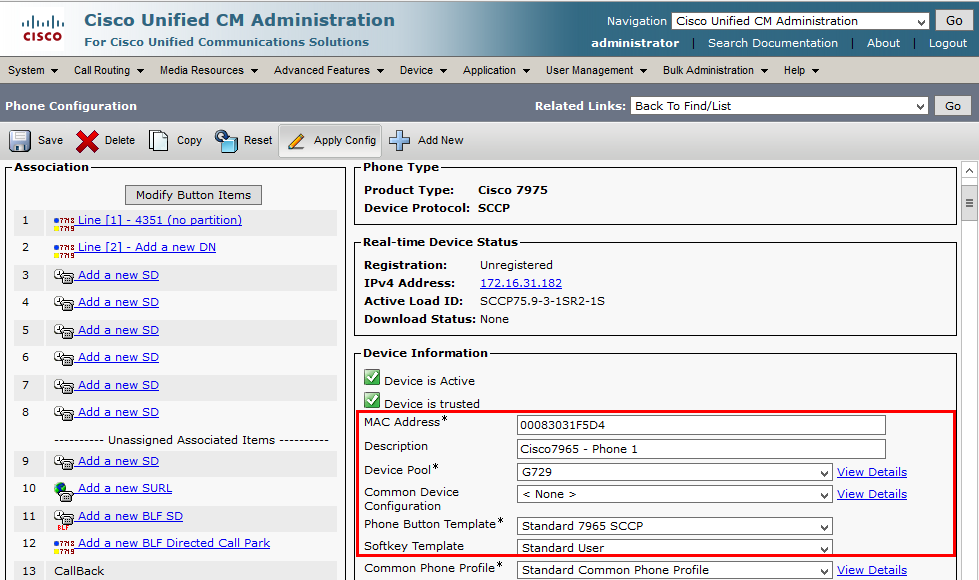
## Cisco IP Phone 7965 SCCP Configuration

Set MAC Address\* = the below mac is used in this example.

Set Description = Cisco7965\_Phone. This text is used to identify this Phone.

Set Device Pool\*= G729 pool. This is used in this example.

Set Phone Button Template\*= Standard 7965 SCCP. This is used in this example. Set Soft key Template = Standard User. This is used in this example.



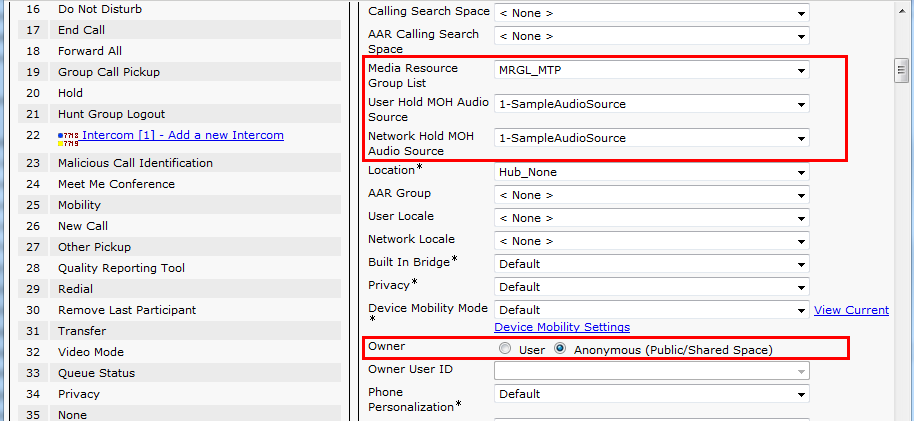
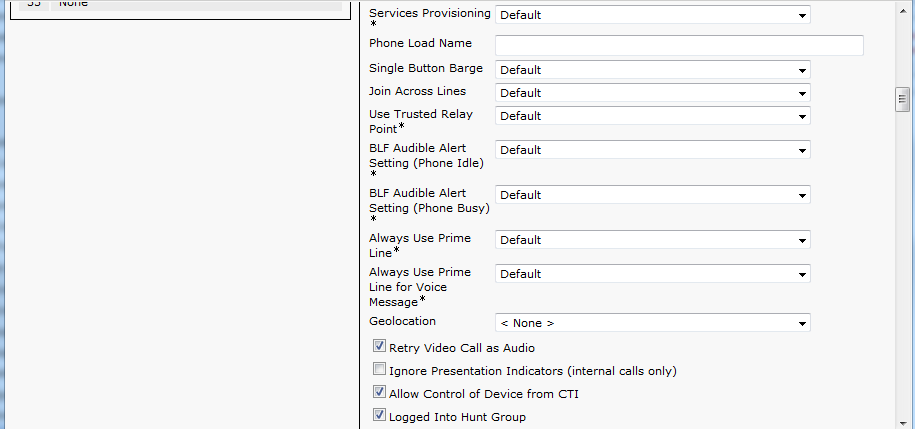
Cisco IP Phone 7965 SCCP Configuration (Continued…)

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

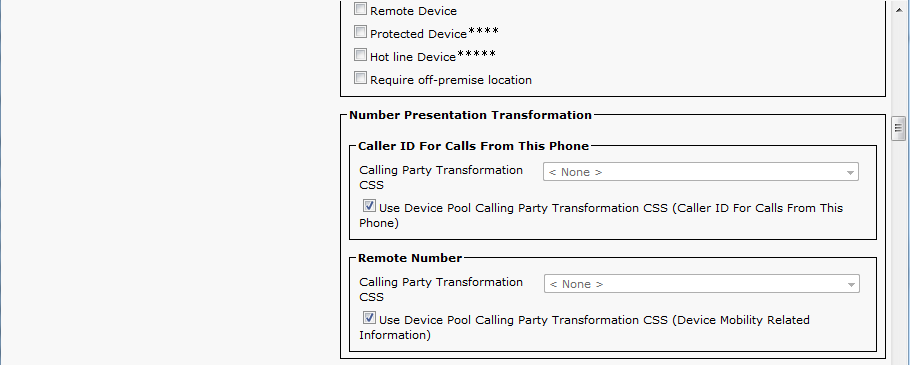
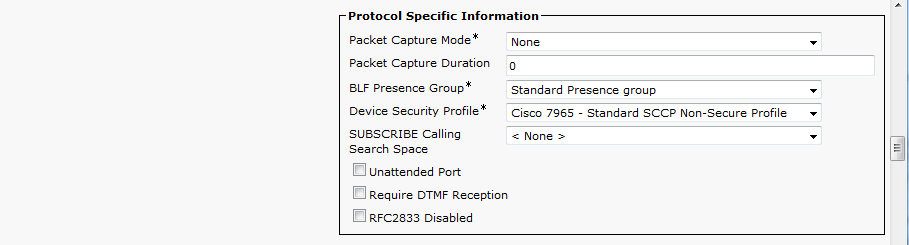
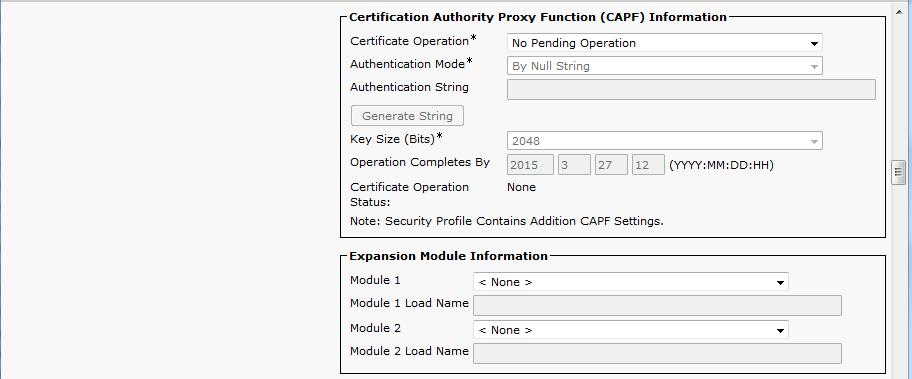
Set User Hold MOH Audio Source = 1-SampleAudioSource.

Set Network Hold MOH Audio Source = 1-SampleAudioSource.

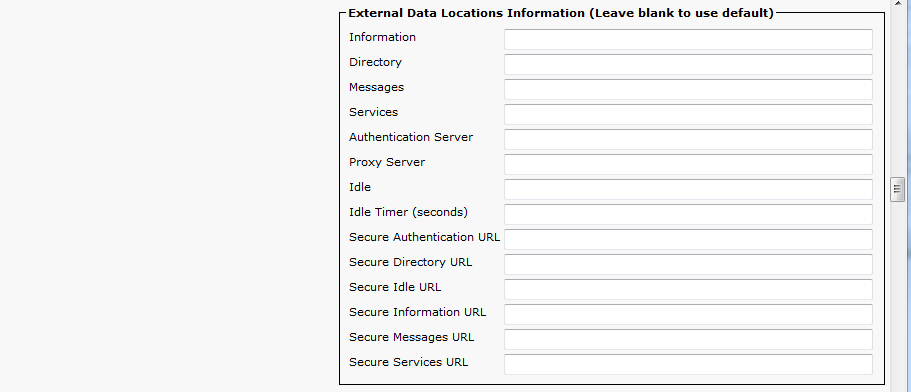
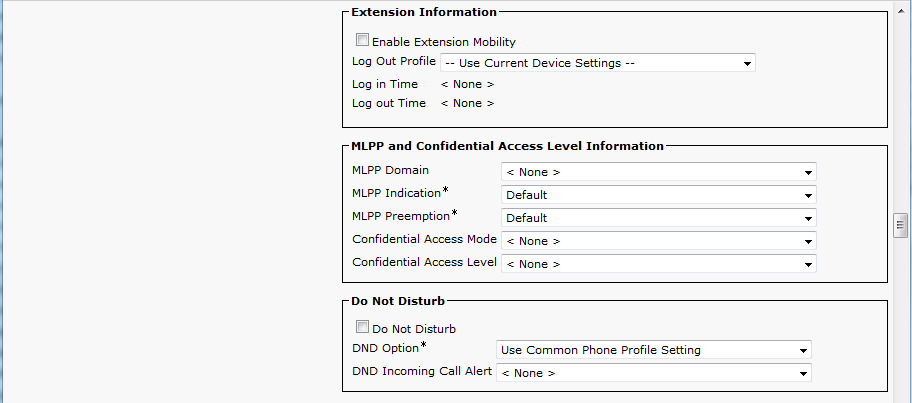
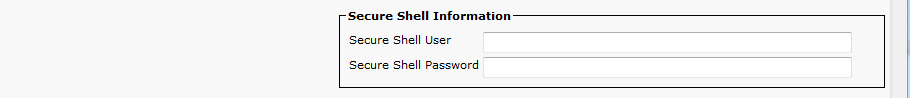
Check Owner = Anonymous (Public/Shared Space). This is used in this example.

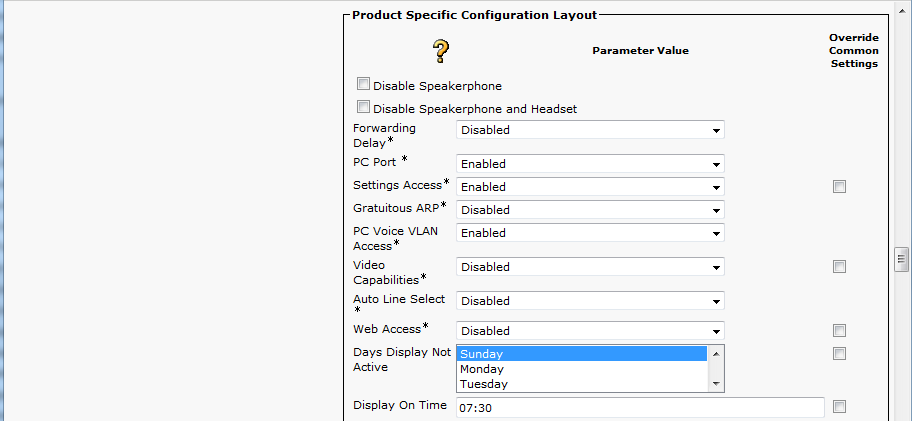
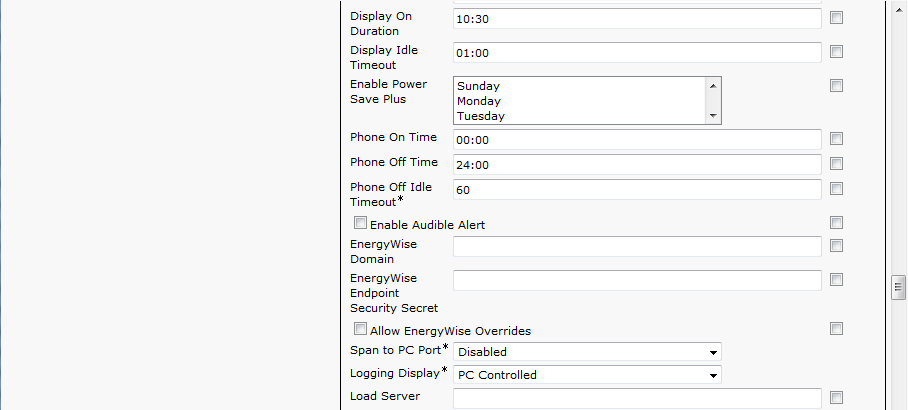
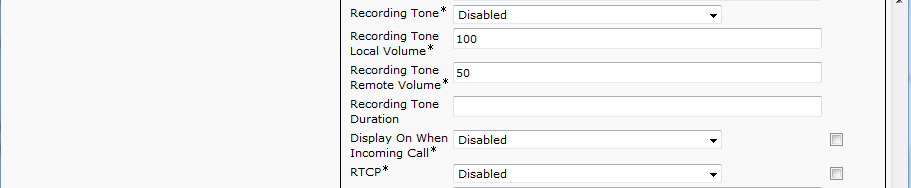
Cisco IP Phone 7965 SCCP Configuration (Continued…)

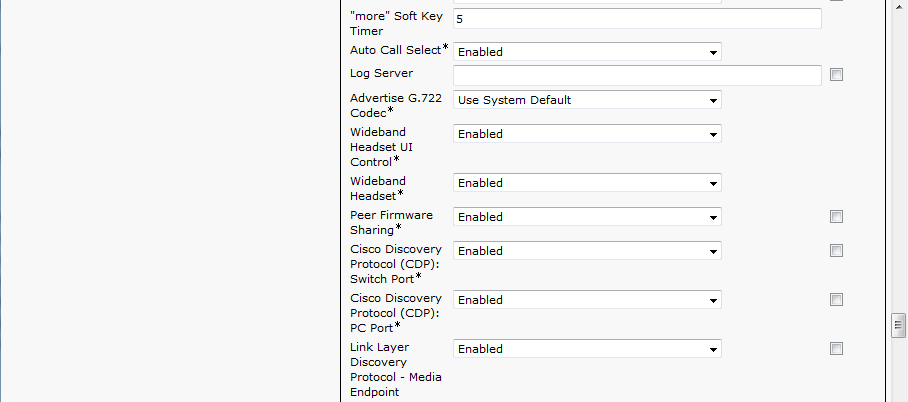
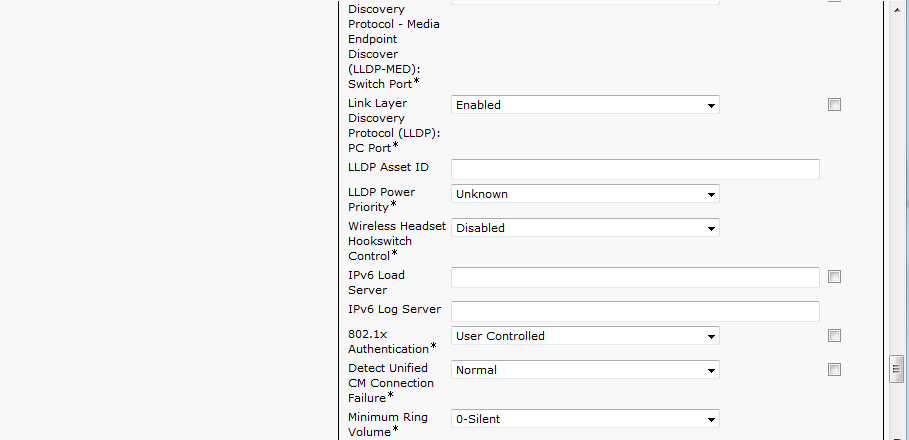
Cisco IP Phone 7965 SCCP Configuration (Continued…)

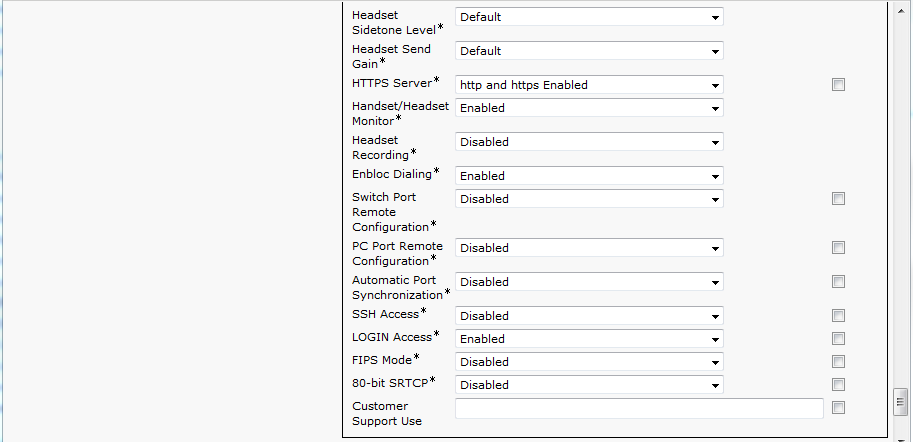
Cisco IP Phone 7965 SCCP Configuration (Continued…)

Cisco IP Phone 7965 SCCP Configuration (Continued…)

Cisco IP Phone 7965 SCCP Configuration (Continued…)

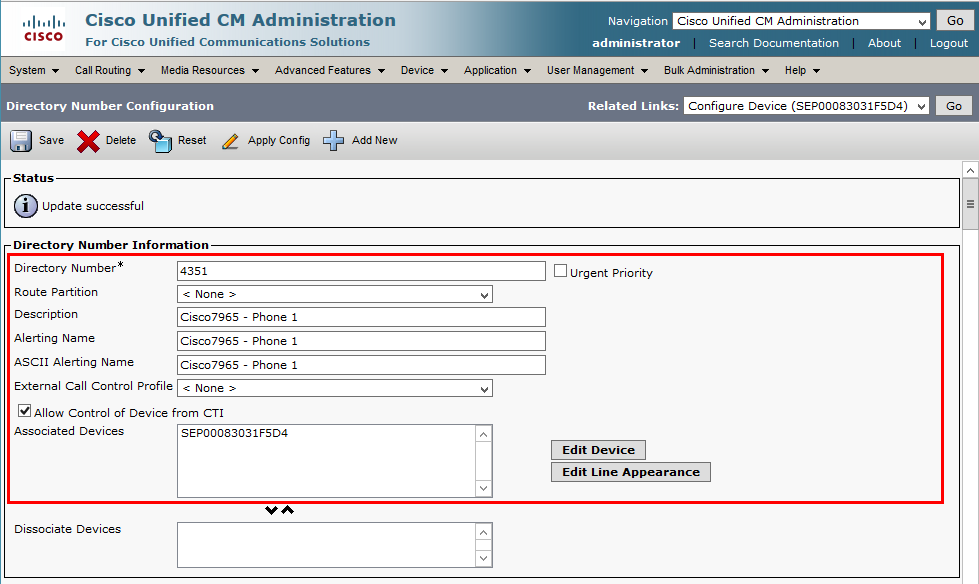
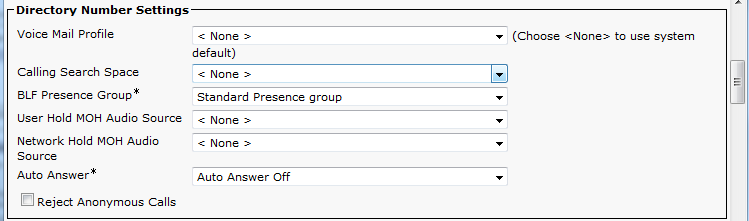
Cisco IP Phone 7965 SCCP Configuration (Continued…)

Set Directory Number\* = 4351. This is used in this example.

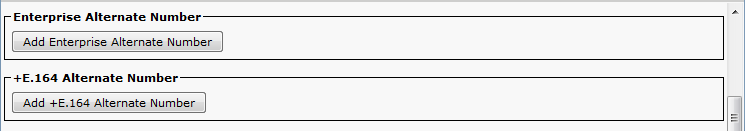
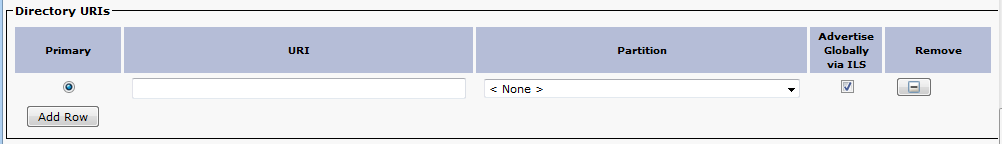
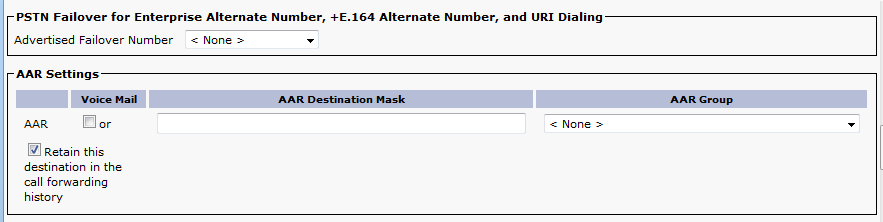
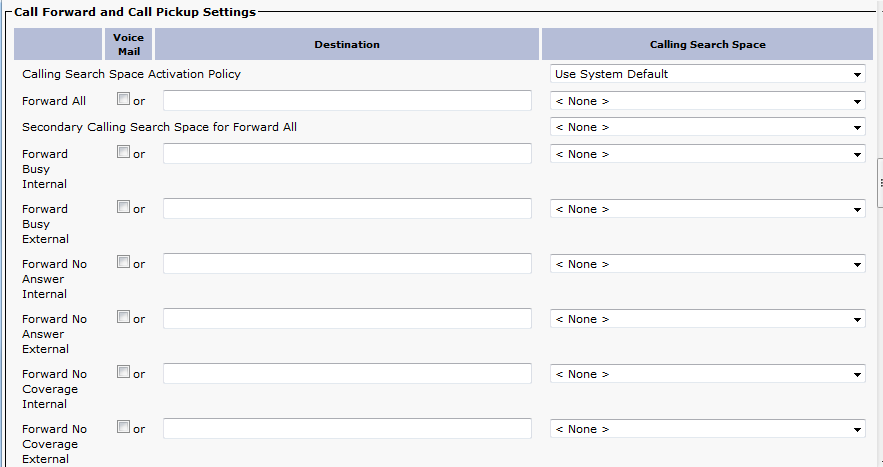
Set Description = 7323204351. This is used in this example.

Set Alerting Name = Cisco 7965 Phone. This is used in this example.

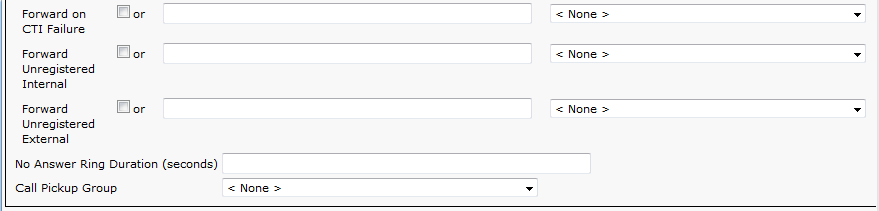
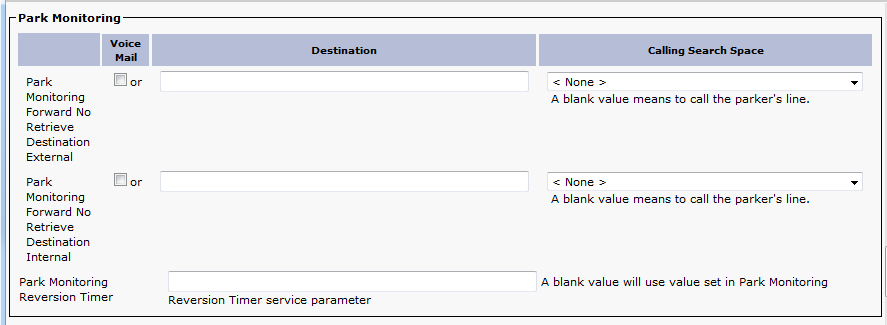
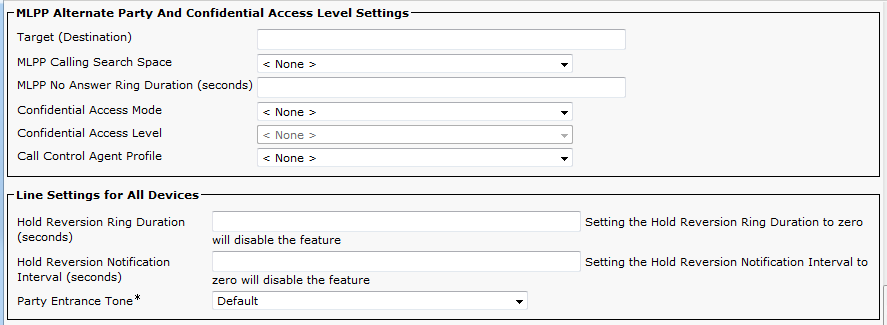
Set ASCII Alerting Name = Cisco 7965 Phone. This is used in this example.

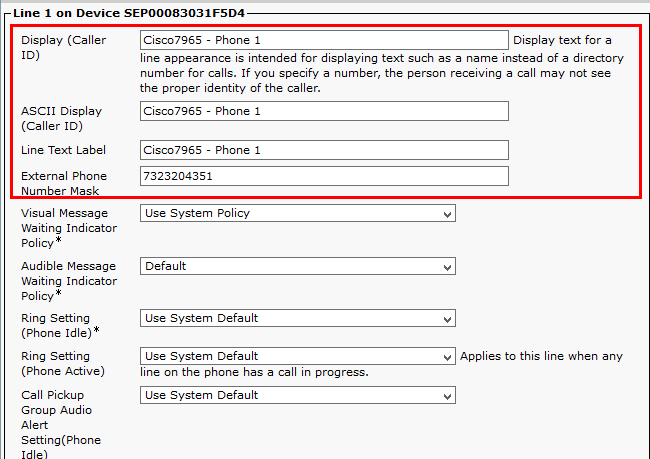
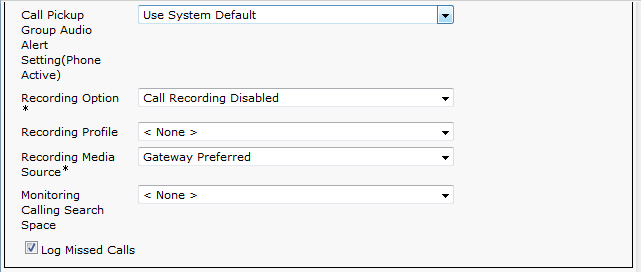
Cisco IP Phone 7965 SCCP Configuration (Continued…)

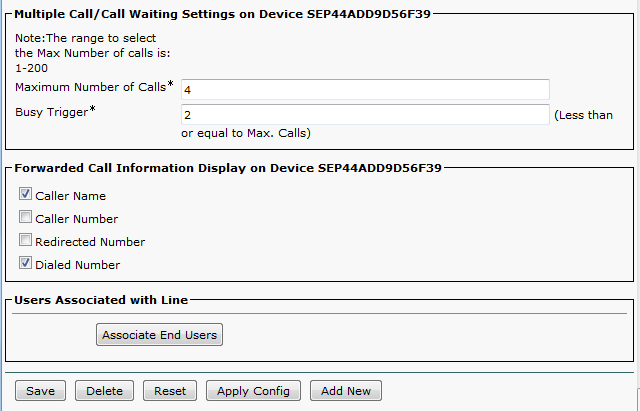
Cisco IP Phone 7965 SCCP Configuration (Continued…)

Cisco IP Phone 7965 SCCP Configuration (Continued…)

Cisco IP Phone 7965 SCCP Configuration (Continued…)



## Cisco IP Phone 7975 SCCP Configuration

Set MAC Address\* = the below mac is used in this example.

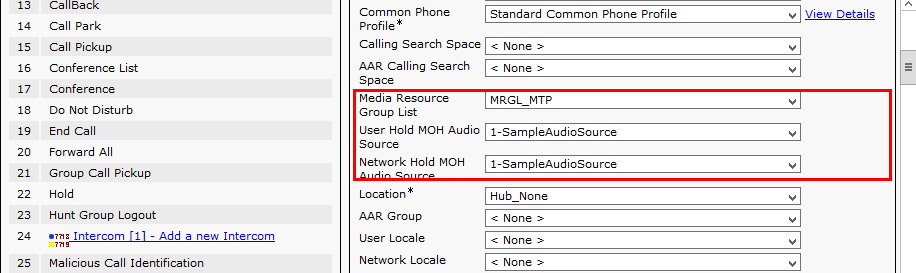
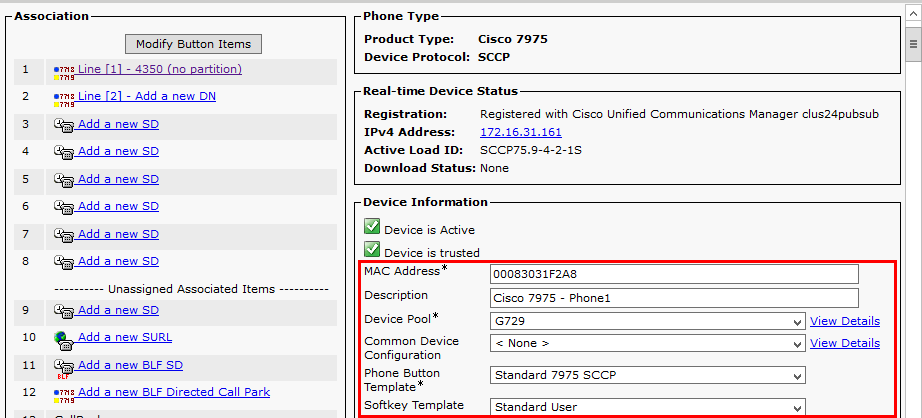
Set Description = Cisco 7975 Phone. This text is used to identify this Phone.

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

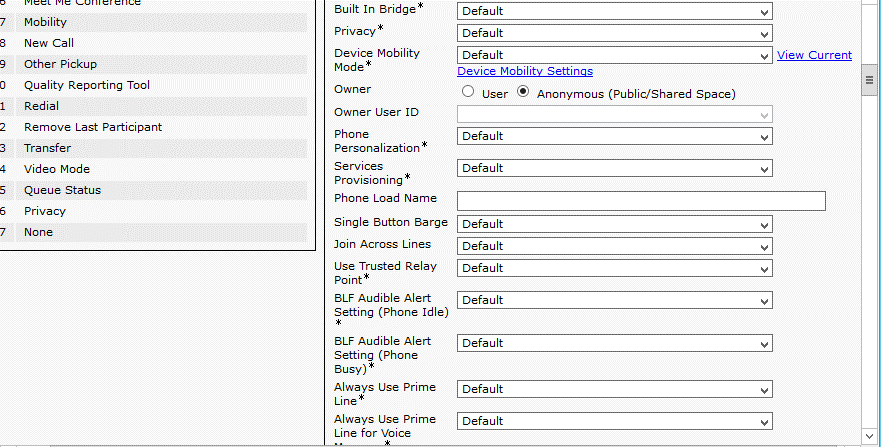
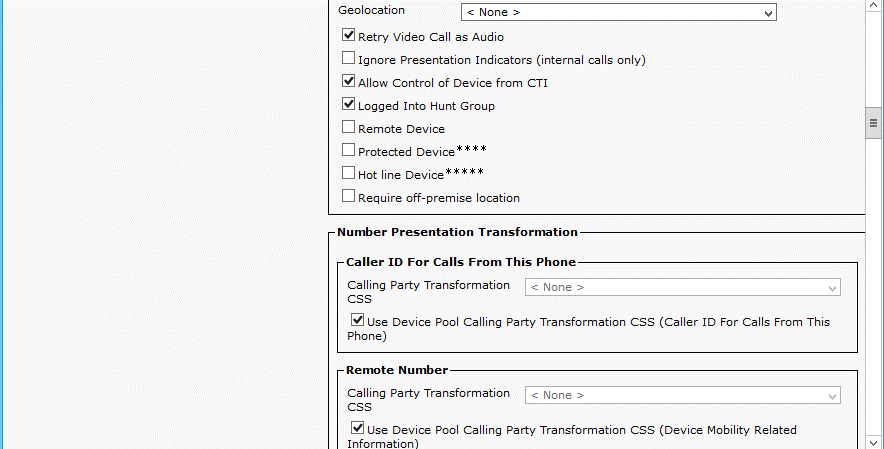
Set Phone Button Template\*= Standard 7975 SCCP. This is used in this example. Set Media Resource Group List = MRGL\_MTP. This is used in this example.

Set User Hold MOH Audio Source = 1-SampleAudioSource.

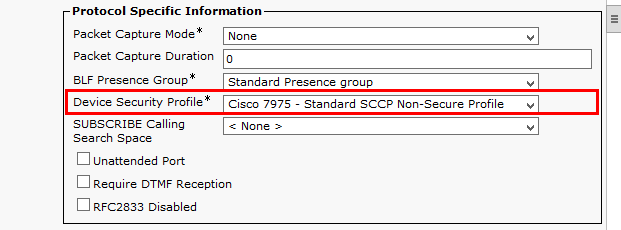
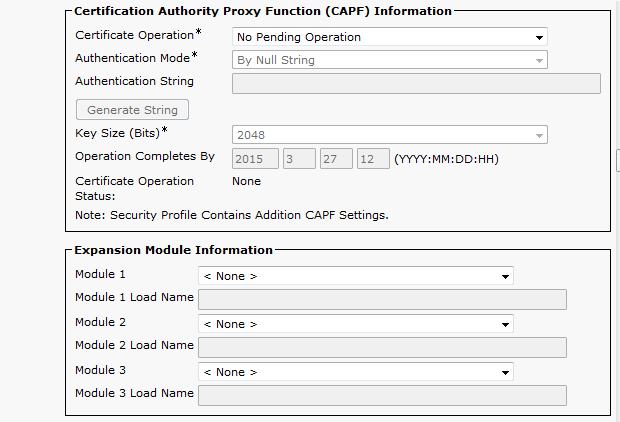
Set Network Hold MOH Audio Source = 1-SampleAudioSource

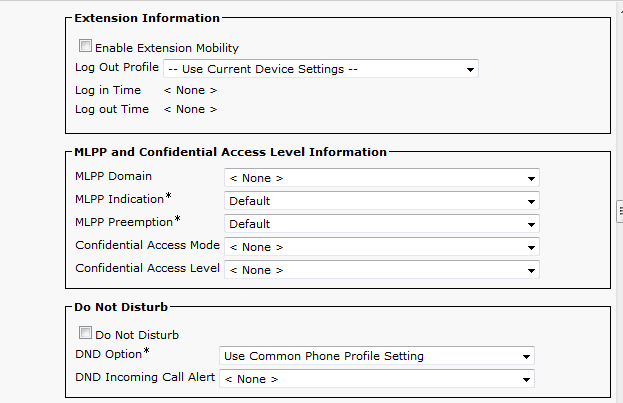
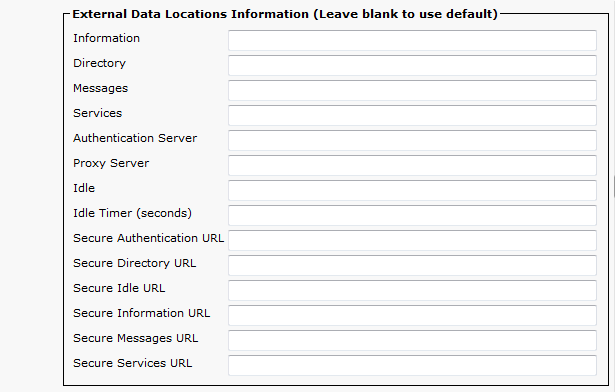


Cisco IP Phone 7975 SCCP Configuration (Continued…)

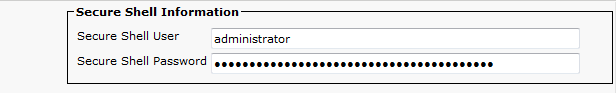
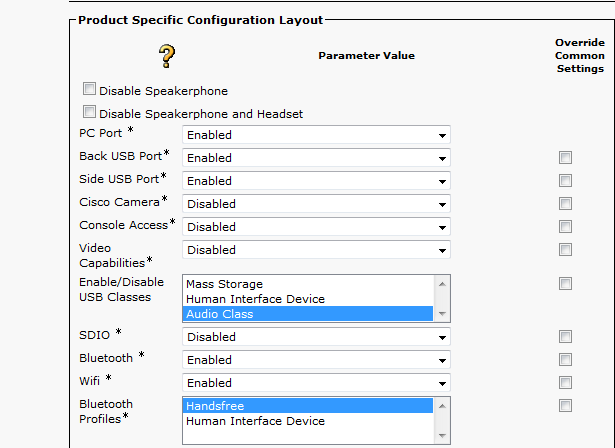
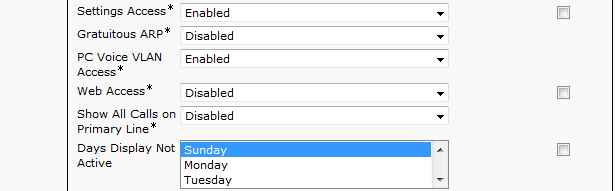
 

Cisco IP Phone 7975 SCCP Configuration (Continued…)

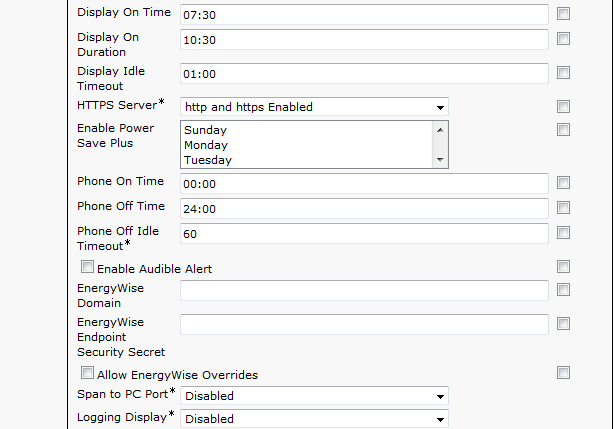
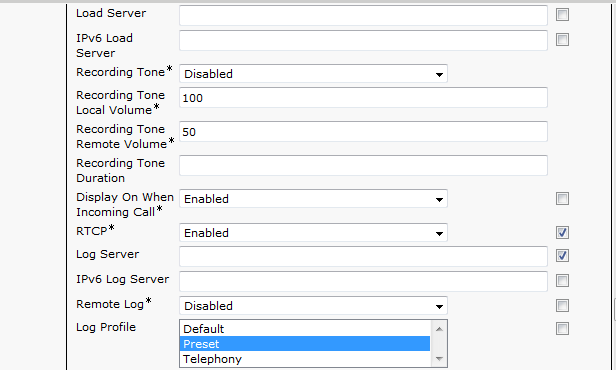
 

Cisco IP Phone 7975 SCCP Configuration (Continued…)

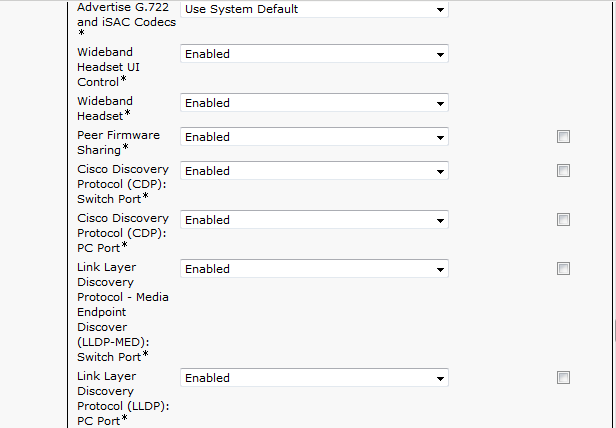
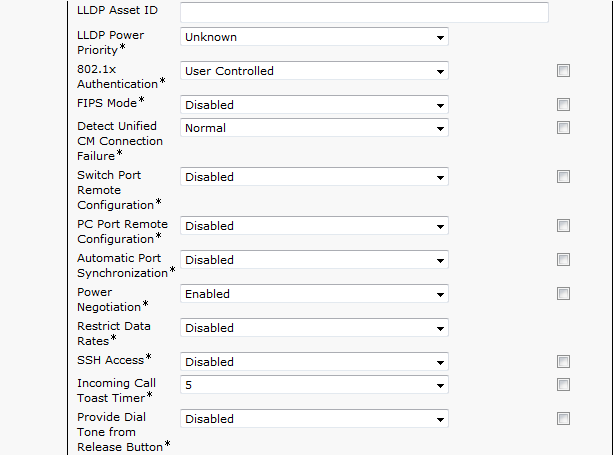
Cisco IP Phone 7975 SCCP Configuration (Continued…)

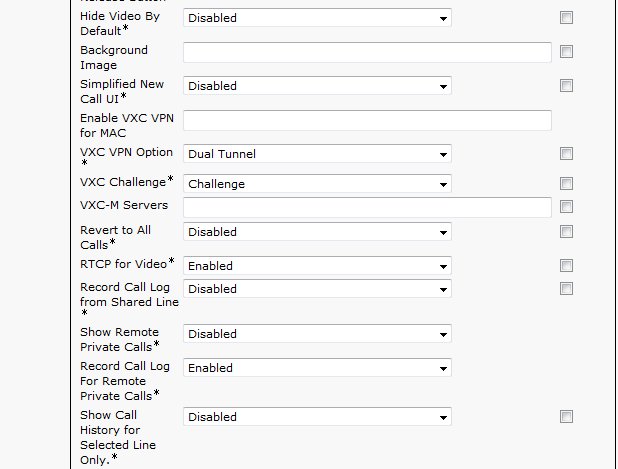
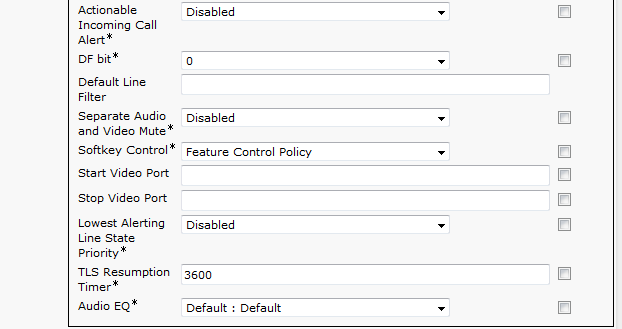
Cisco IP Phone 7975 SCCP Configuration (Continued…)

Cisco IP Phone 7975 SCCP Configuration (Continued…)

Cisco IP Phone 7975 SCCP Configuration (Continued…)

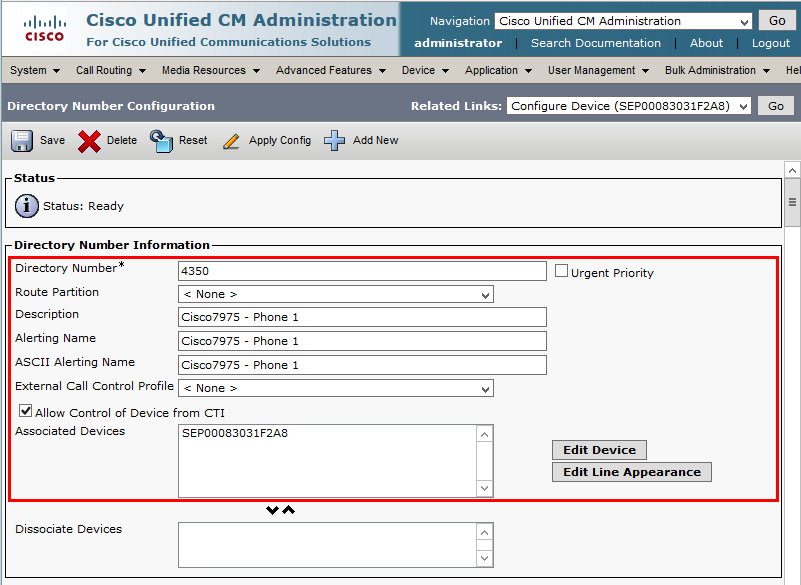
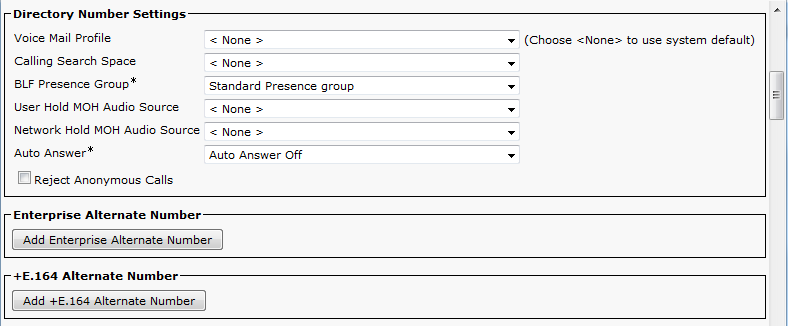
Cisco IP Phone 7975 SCCP Configuration (Continued…)

Set Directory Number\* = 4350. This is used in this example.

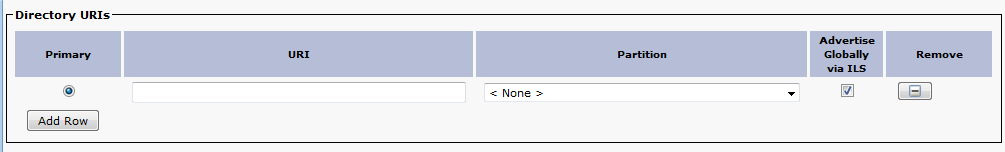
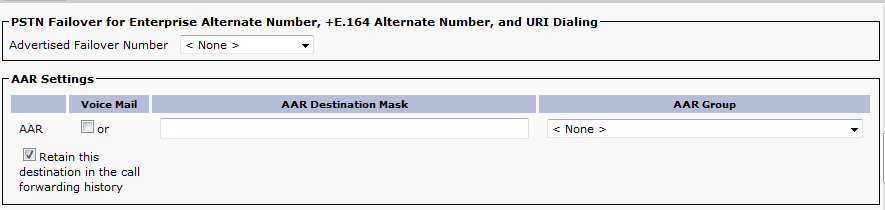
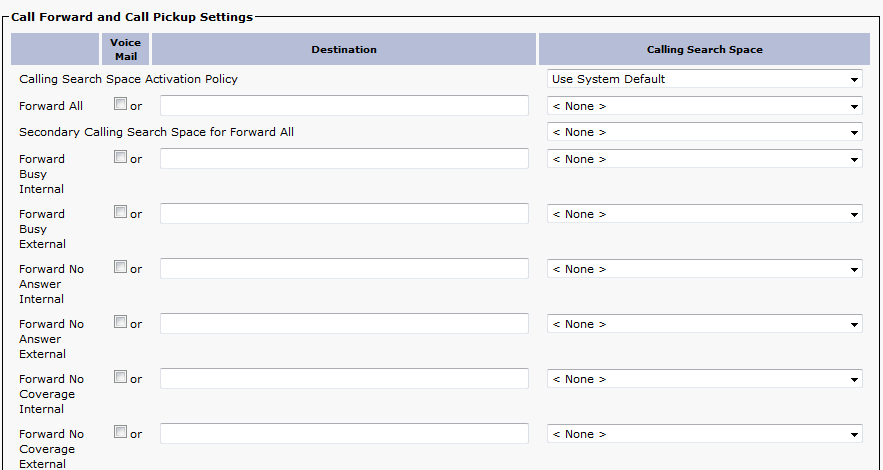
Set Description = 7323204350. This is used in this example.

Set Alerting Name = Cisco7975 Phone. This is used in this example.

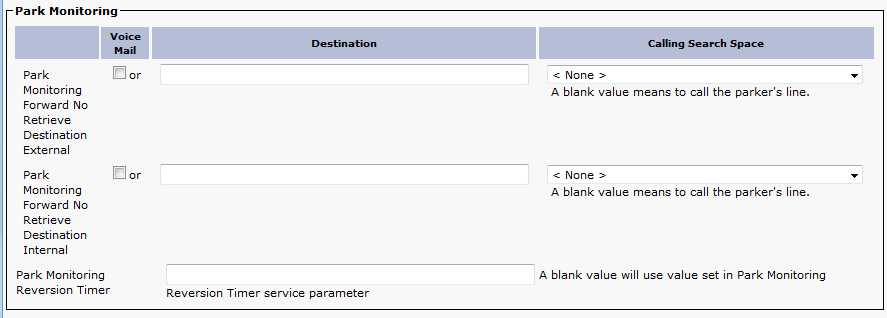
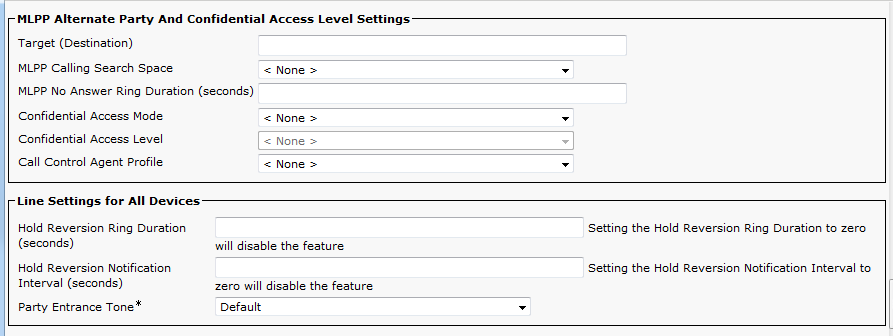
Set ASCII Alerting Name = Cisco7975 Phone. This is used in this example.

Cisco IP Phone 7975 SCCP Configuration (Continued…)

Cisco IP Phone 7975 SCCP Configuration (Continued…)

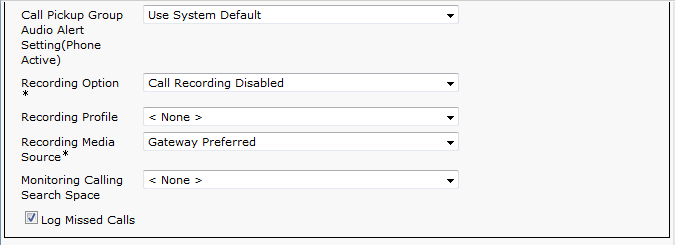
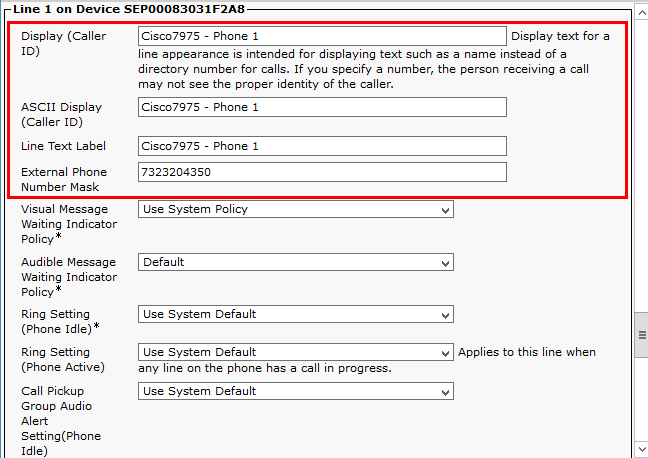
Cisco IP Phone 7975 SCCP Configuration (Continued…)

Set Display (caller ID) = Cisco7975-Phone 1. This is used in this example.

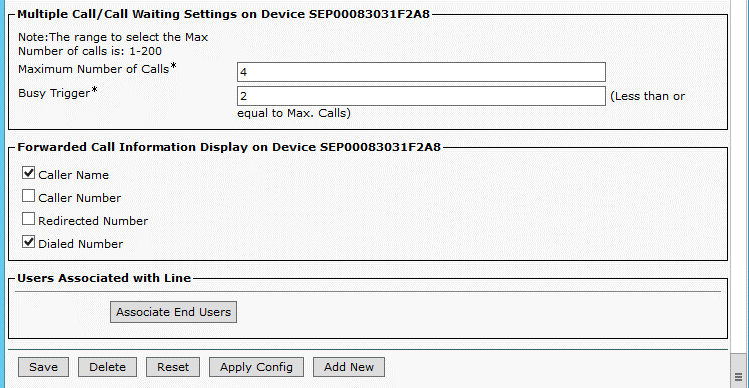
Set ASCII Display (caller ID) = Cisco7975-Phone 1. This is used in this example.

Set Line Text Label = Cisco7975-Phone 1. This is used in this example.

Set External Phone Number Mask = 7323204350. This is used in this example.



Cisco IP Phone 7975 SCCP Configuration (Continued…)



## Cisco IP Phone 9971 SIP Configuration

Set MAC Address\* = the below mac is used in this example.

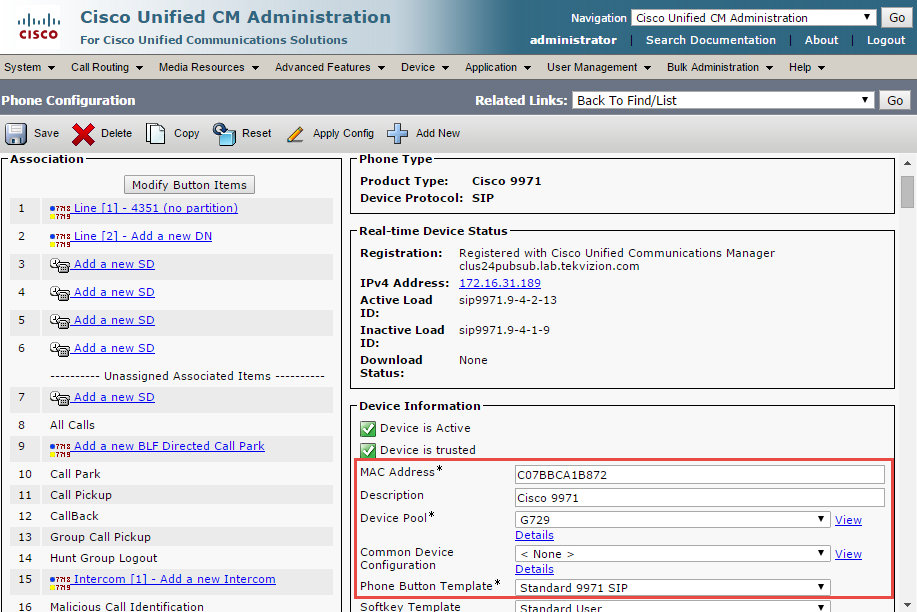
Set Description = Cisco 9971 Phone 2. This text is used to identify this Phone.

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

Set Phone Button Template\*= Standard 9971 SIP. This is used in this example. Set Media Resource Group List = MRGL\_MTP. This is used in this example.

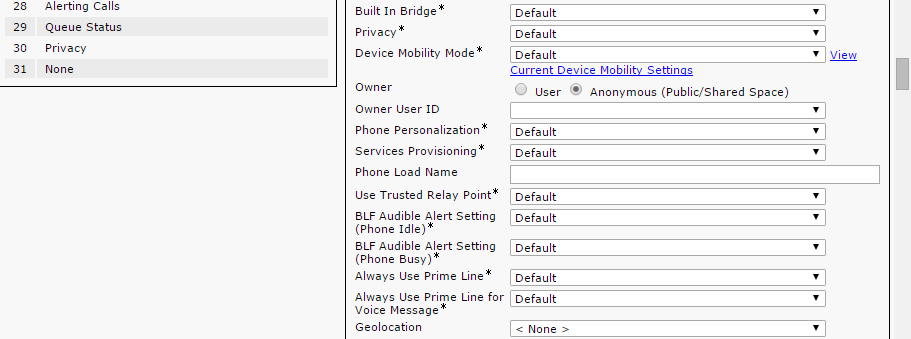
Set User Hold MOH Audio Source = 1-SampleAudioSource.

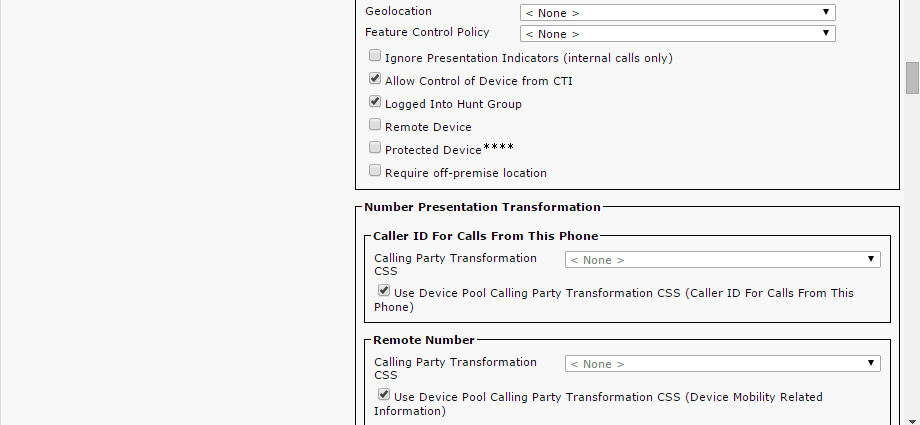
Set Network Hold MOH Audio Source = 1-SampleAudioSource



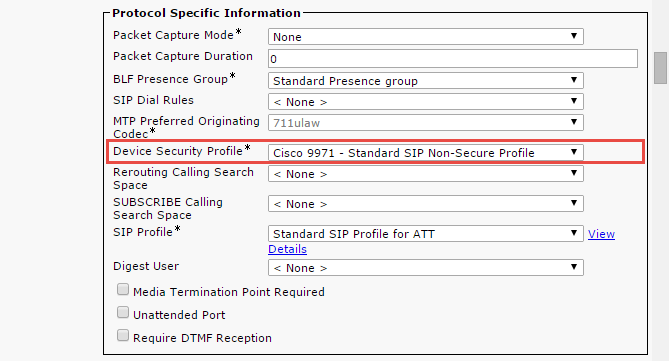


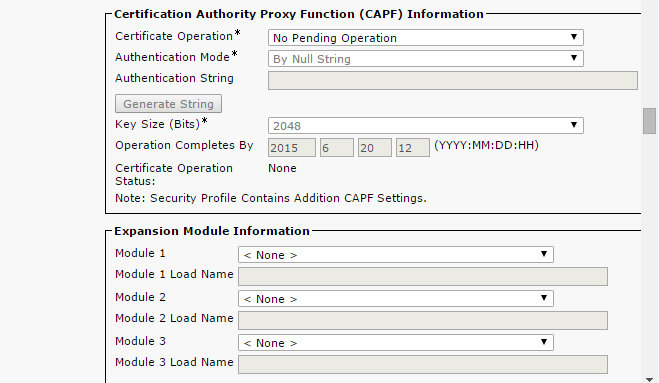
Cisco IP Phone 9971 SIP Configuration(Continued…)



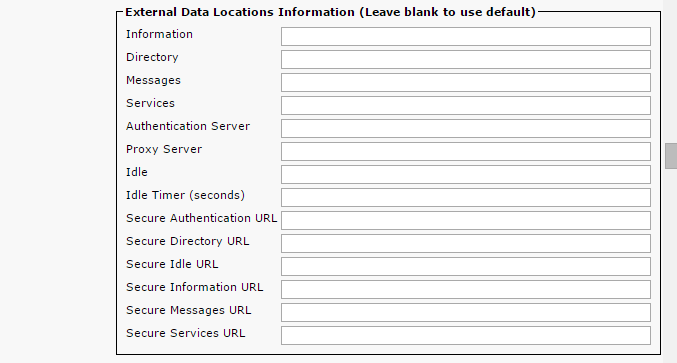


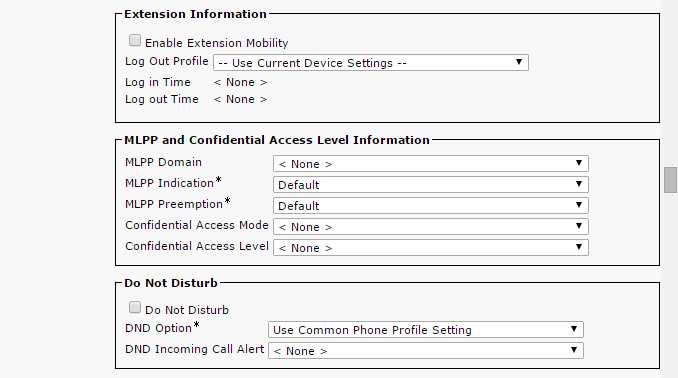
Cisco IP Phone 9971 SIP Configuration(Continued…)



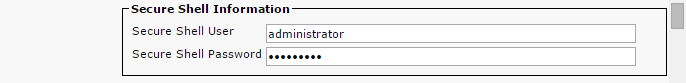


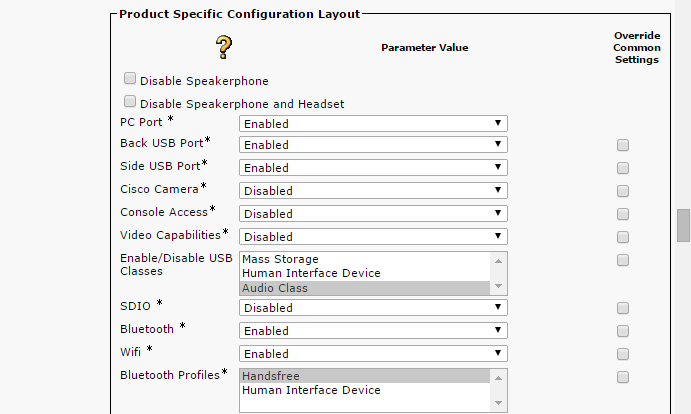
Cisco IP Phone 9971 SIP Configuration(Continued…)

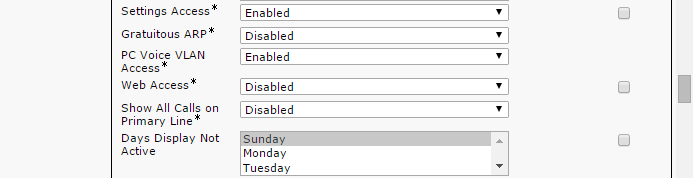




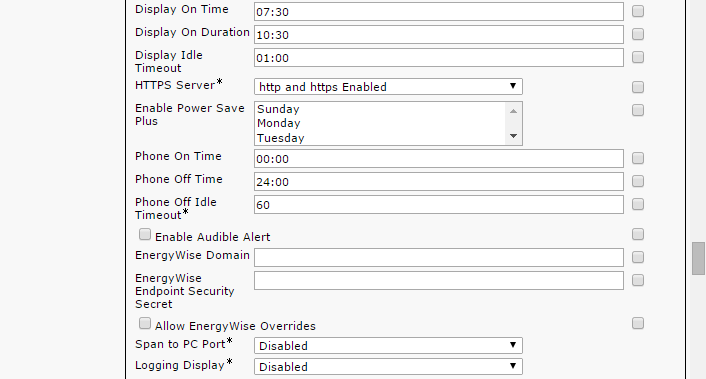
Cisco IP Phone 9971 SIP Configuration(Continued…)

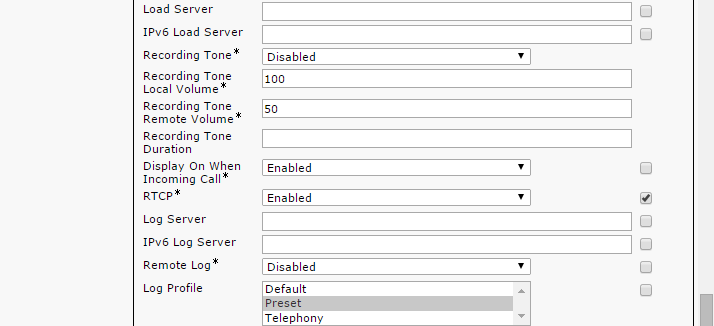




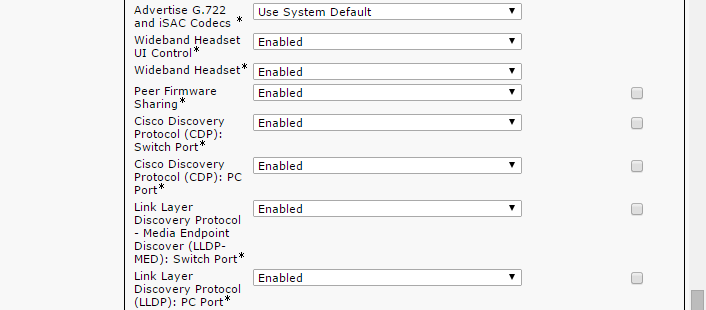


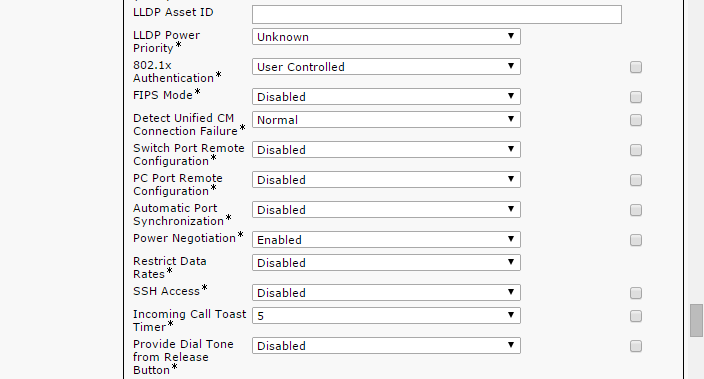
Cisco IP Phone 9971 SIP Configuration(Continued…)



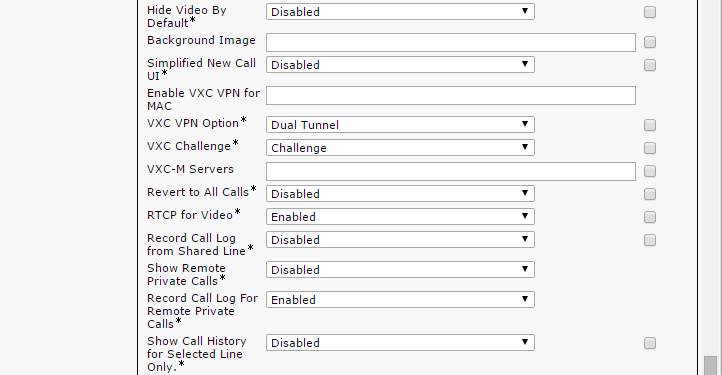


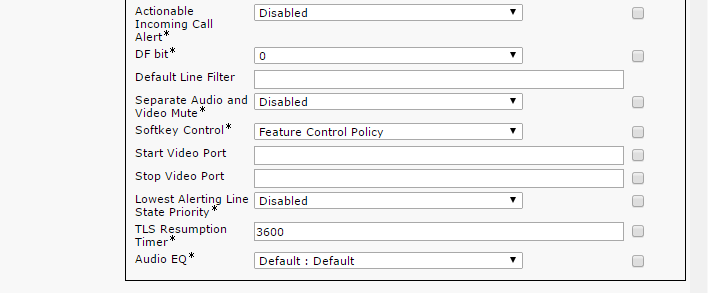
Cisco IP Phone 9971 SIP Configuration (Continued…)





Cisco IP Phone 9971 SIP Configuration (Continued…)







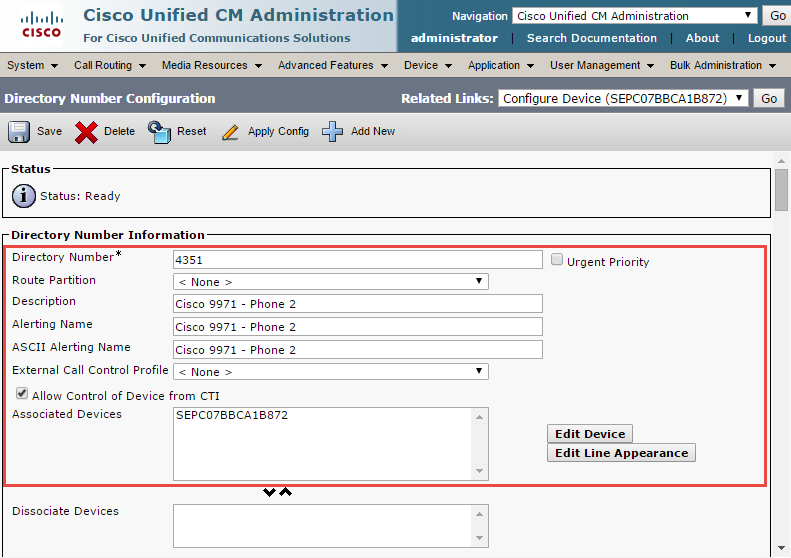
Cisco IP Phone 9971 SIP Configuration (Continued…)

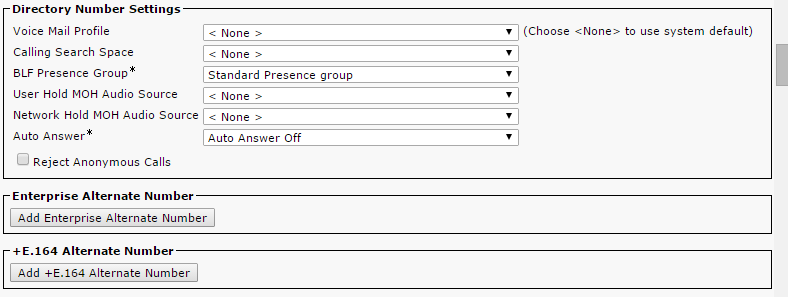
Set Directory Number\* = 4351. This is used in this example.

Set Description = 7323204351. This is used in this example.

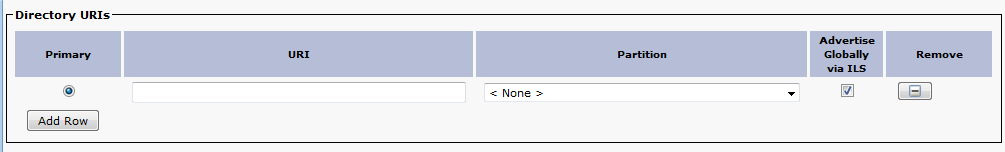
Set Alerting Name = Cisco 9971 Phone 2. This is used in this example.

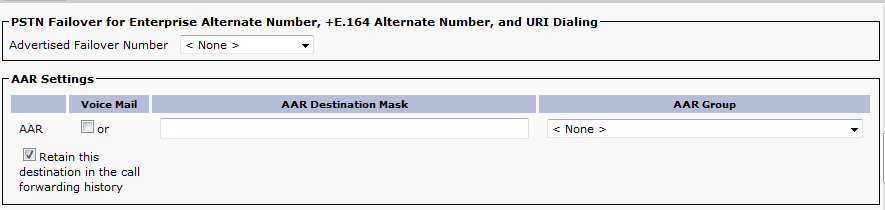
Set ASCII Alerting Name = Cisco 9971 Phone 2. This is used in this example.

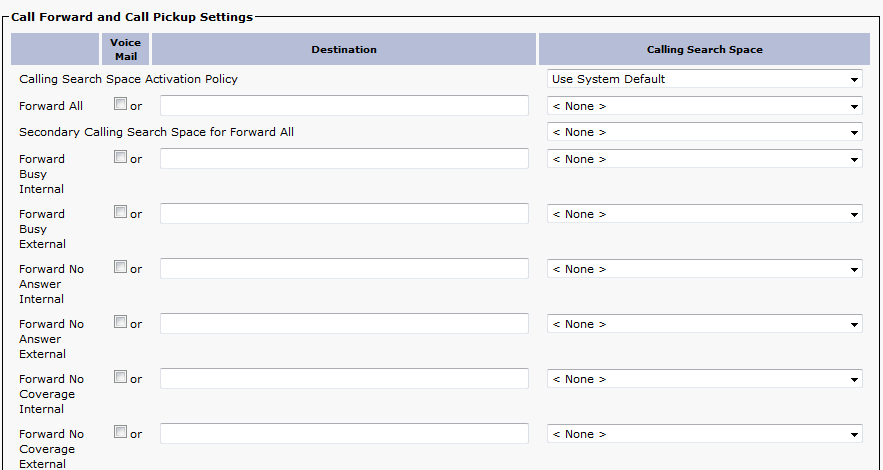




Cisco IP Phone 9971 SIP Configuration (Continued…)

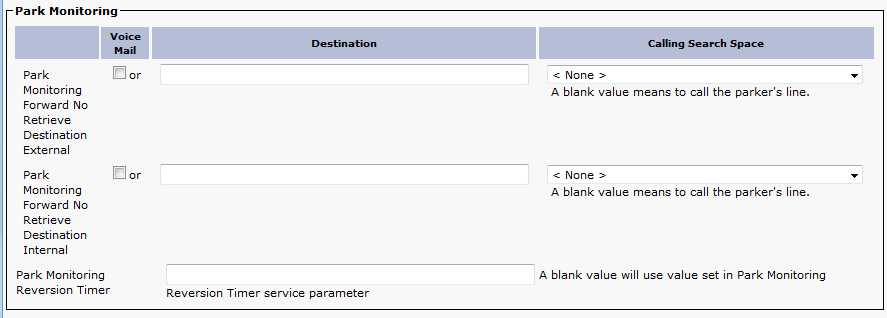


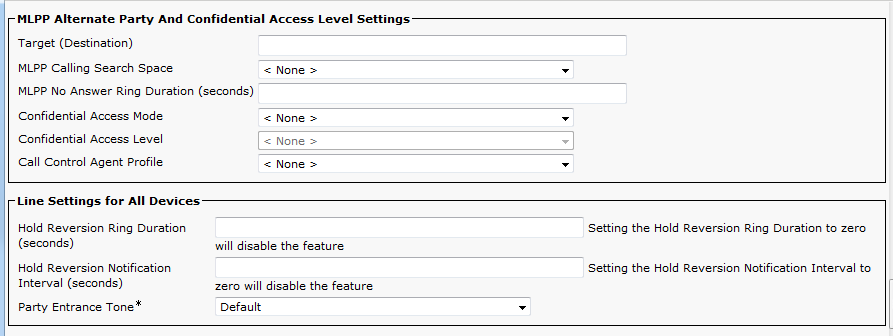




Cisco IP Phone 9971 SIP Configuration (Continued…)







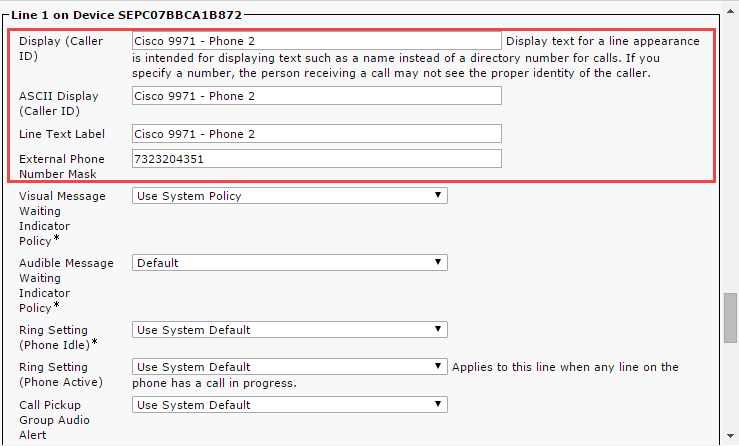
Cisco IP Phone 9971 SIP Configuration (Continued…)

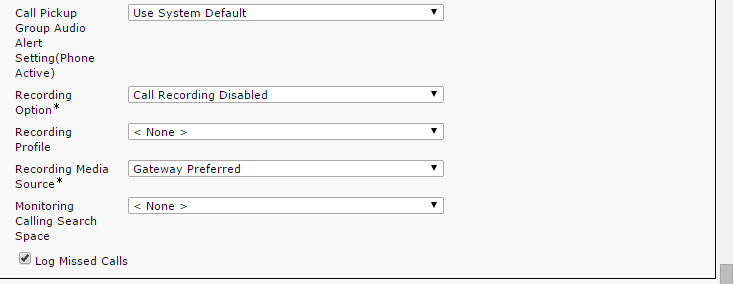
Set Display (caller ID) = Cisco9971-Phone 2. This is used in this example.

Set ASCII Display (caller ID) = Cisco9971-Phone 2. This is used in this example.

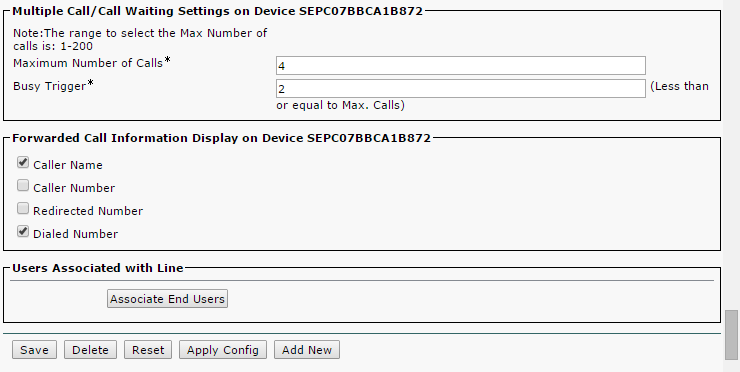
Set Line Text Label = Cisco9971-Phone 2. This is used in this example.

Set External Phone Number Mask = 7323204351. This is used in this example.





Cisco IP Phone 9971 SIP Configuration (Continued…)



## SIP Trunk Security Profile Configuration used by SIP trunk to Cisco UBE

**Navigation:** System 🡪 Security 🡪 SIP Trunk Security Profile

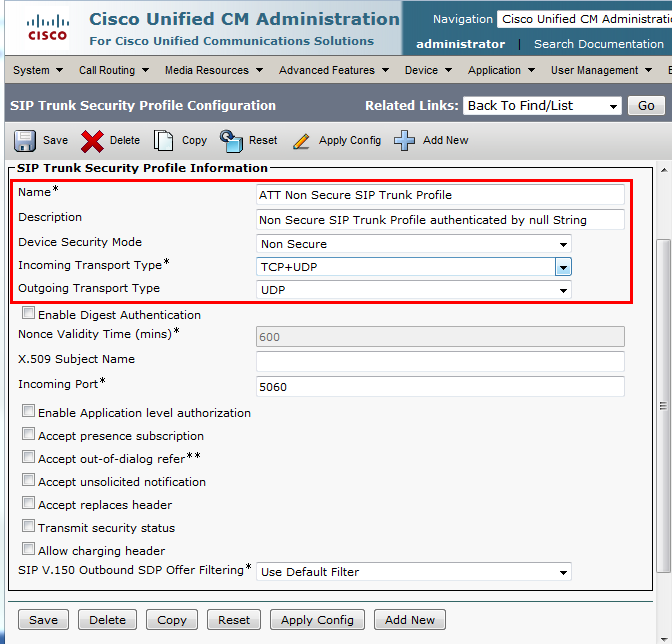
Set Name\* = ATT Non Secure SIP Trunk Profile. This is used in this example.

Set Description = Non Secure SIP Trunk Profile authenticated by null String. This is used in this example.

Set Device Security Mode = Non Secure.

Set Incoming Transport Type\* = TCP+UDP.

Set Outgoing Transport Type = UDP.



## SIP Profile Configuration used by SIP trunk to Cisco UBE

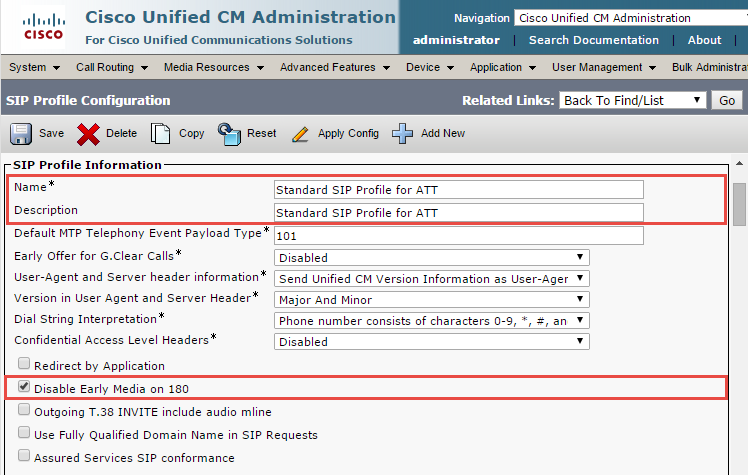
**Navigation:** Device 🡪 Device Settings 🡪 SIP Profile

Set SIP profile Name \* = Standard SIP Profile w/Early Media Disabled. This is used for this example

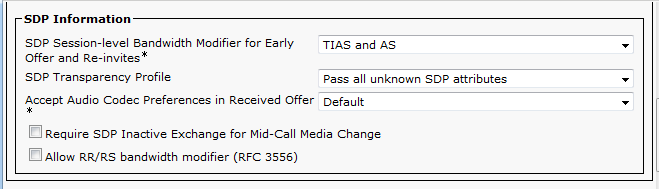
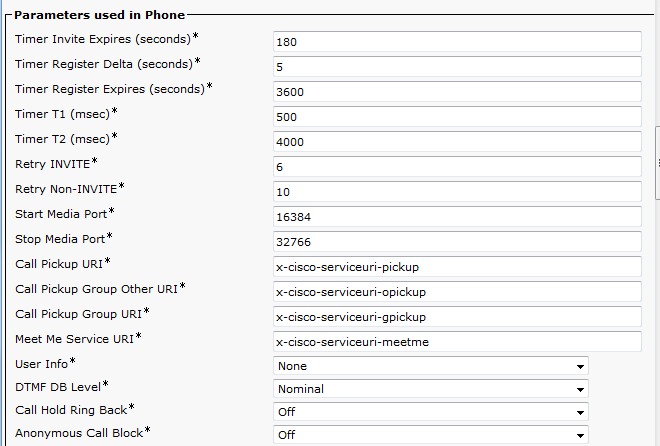
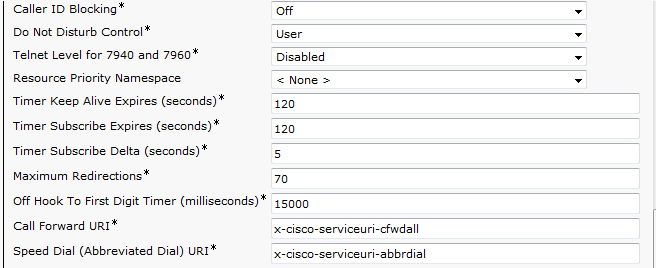
Check Disable Early Media on 180

Set SIP Rel1xx Options\* = Send PRACK if 1xx contains SDP

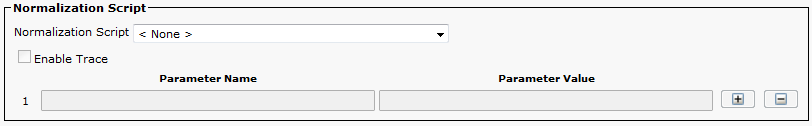
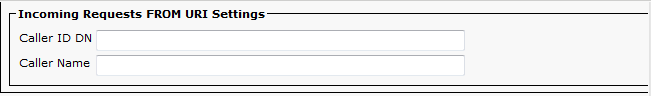
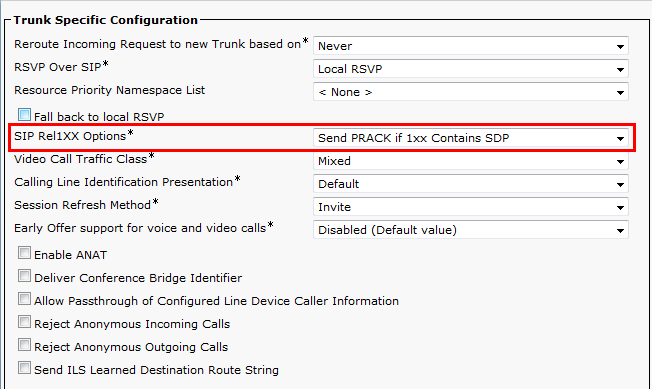
Note\*= Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful early-media cut-through, the Cisco UCM to Cisco UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the Cisco UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”.



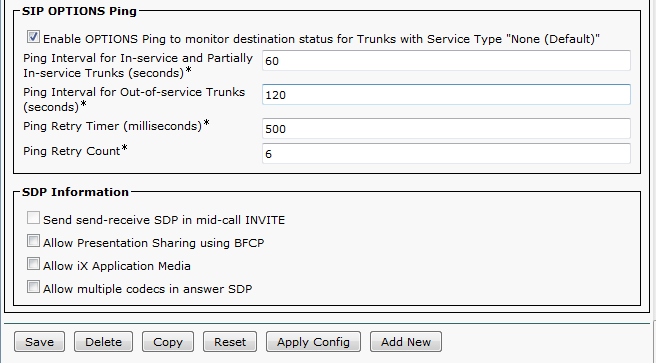
SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)

SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)

SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued…)



## SIP Trunk to Cisco UBE Configuration

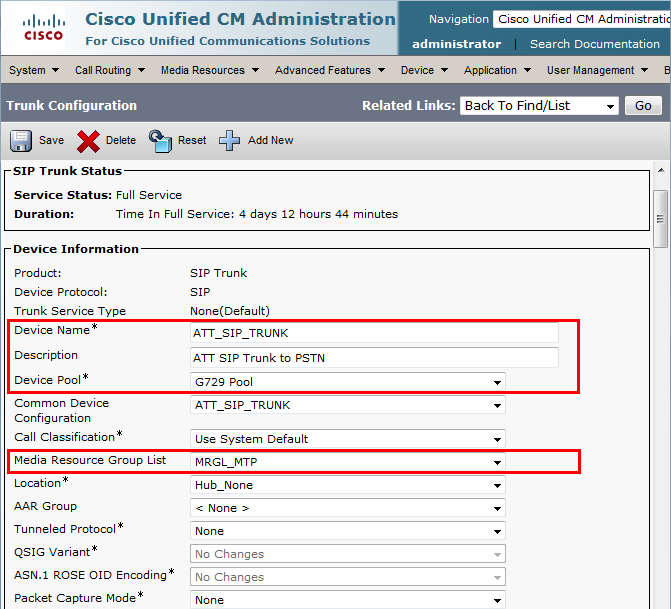
**Navigation:** Device 🡪 Trunk

Set Device Name\* = ATT\_SIP\_TRUNK. This is used for this example

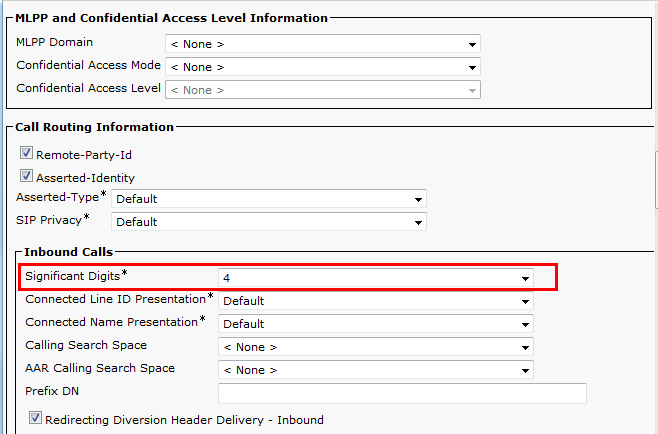
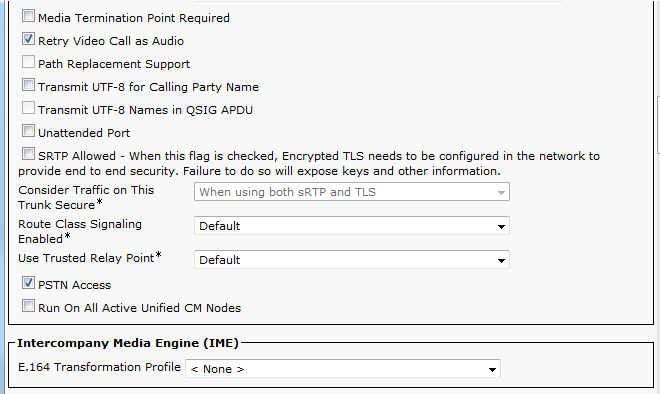
Set Description = ATT SIP Trunk to PSTN. This is used for this example

Set Device Pool\* = G729\_pool. This is used for this example

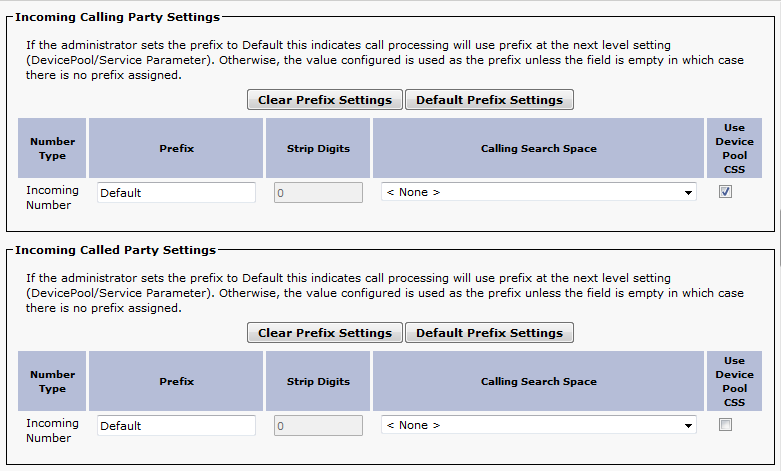
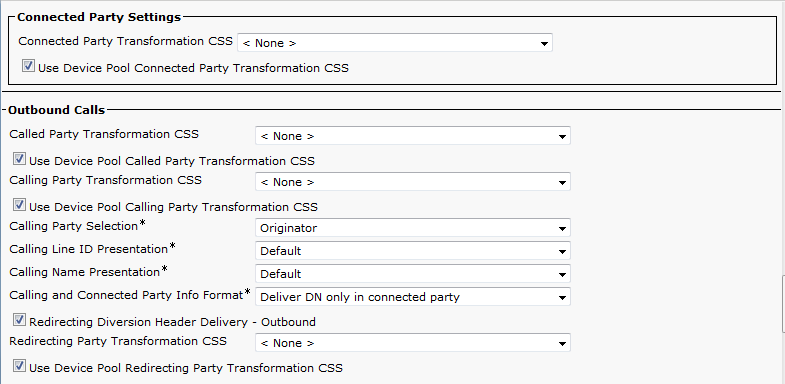
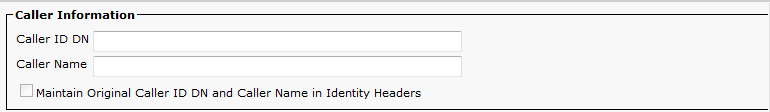
Set Media Resource Group List = MRGL\_MTP.



SIP Trunk to Cisco UBE Configuration (Continued…)

Set Significant Digits\* = 4. This is used in this example.

SIP Trunk to Cisco UBE Configuration (Continued…)

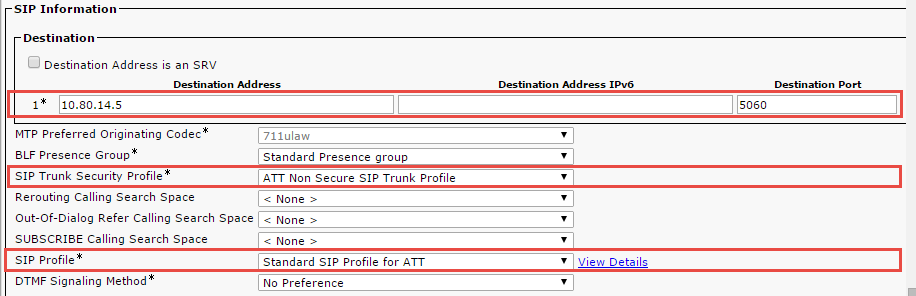
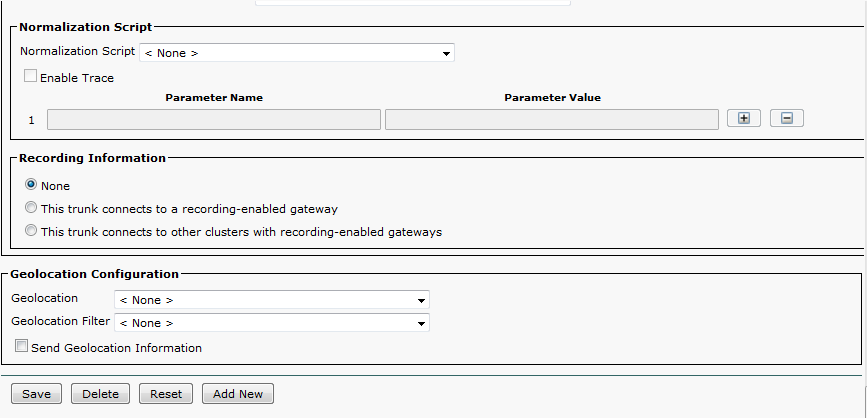
  

SIP Trunk to Cisco UBE Configuration (Continued…)

Set Destination Address = Set IP address of ISR-Cisco UBE.

Set SIP Trunk Security Profile\* = ATT\_Non Secure Sip Trunk Profile.

Set SIP Profile\* = Standard SIP Profile w/Early Media Disabled. This is used in this example.

SIP Trunk to Fax Gateway Configuration.

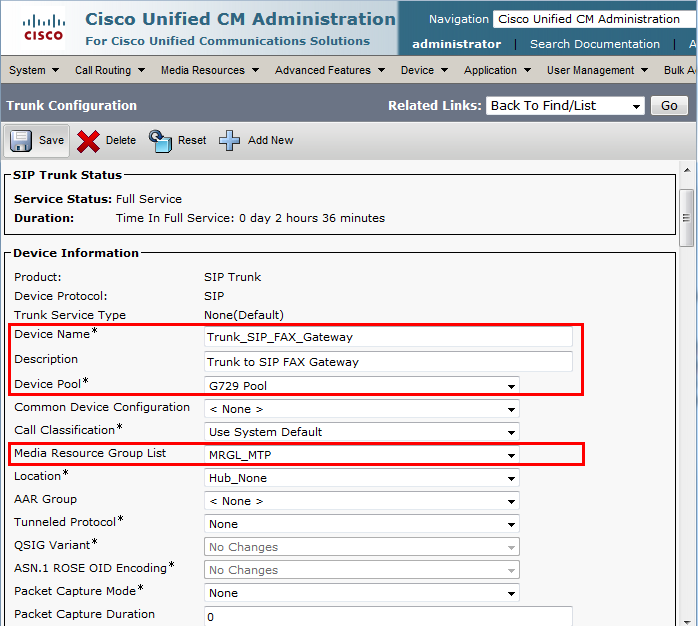
**Navigation:** Device 🡪 Trunk

Set Device Name\* = Trunk\_SIP\_FAX\_Gateway. This is used for this example

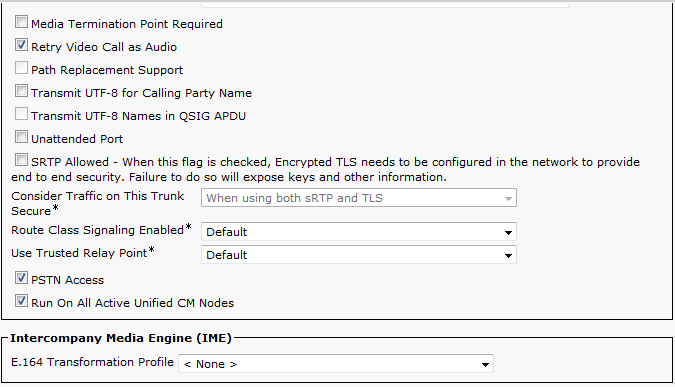
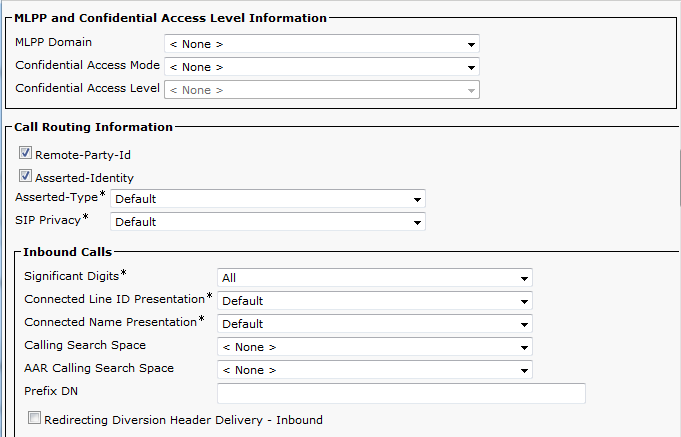
Set Description = Trunk\_SIP\_FAX\_Gateway. This is used for this example

Set Device Pool\* = G729 pool. This is used for this example

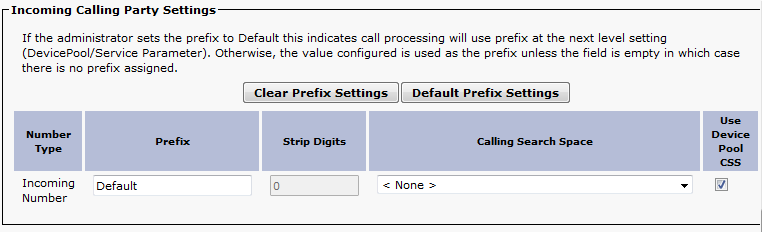
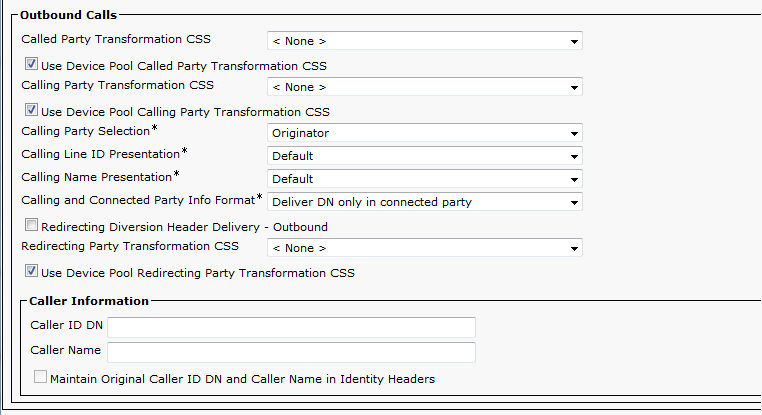
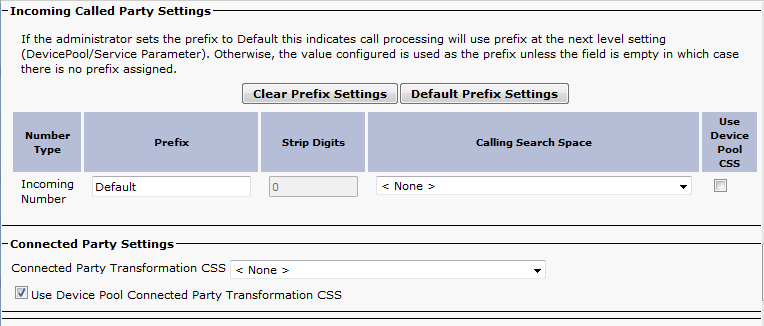
Set Media Resource Group List = MRGL\_MTP.



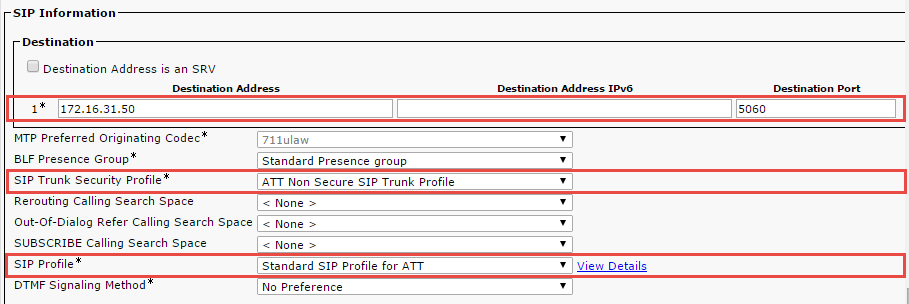
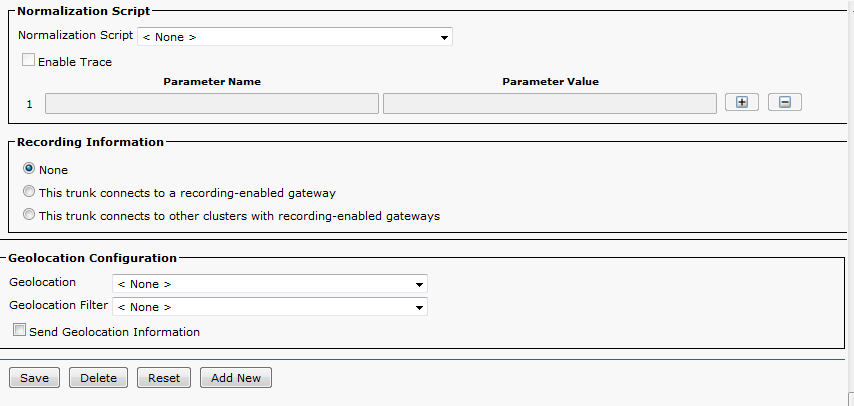
SIP Trunk to Fax Gateway Configuration (Continued…)

SIP Trunk to Fax Gateway Configuration (Continued…)

SIP Trunk to Fax Gateway Configuration (Continued…)

## Route Pattern Configuration

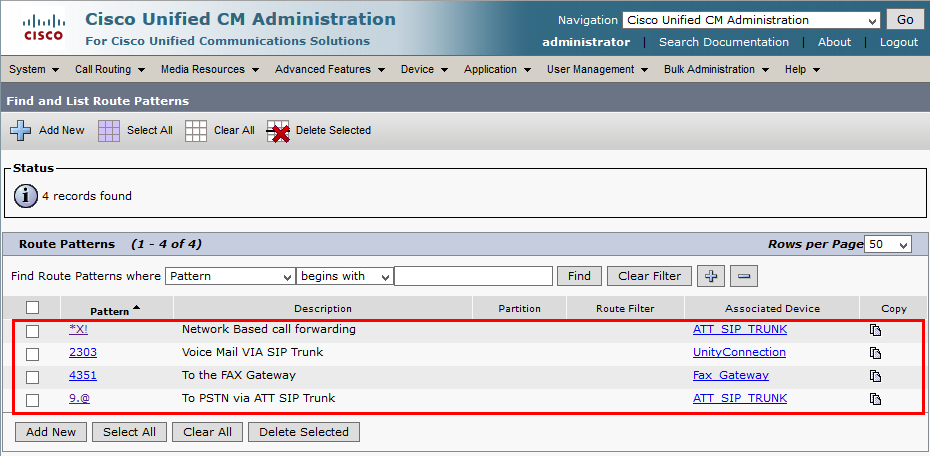
**Navigation:** Call Routing 🡪 Route/Hunt 🡪 Route Pattern

Set Route Pattern\* = 9. @ This is used to route to AT&T via ISR Cisco UBE.

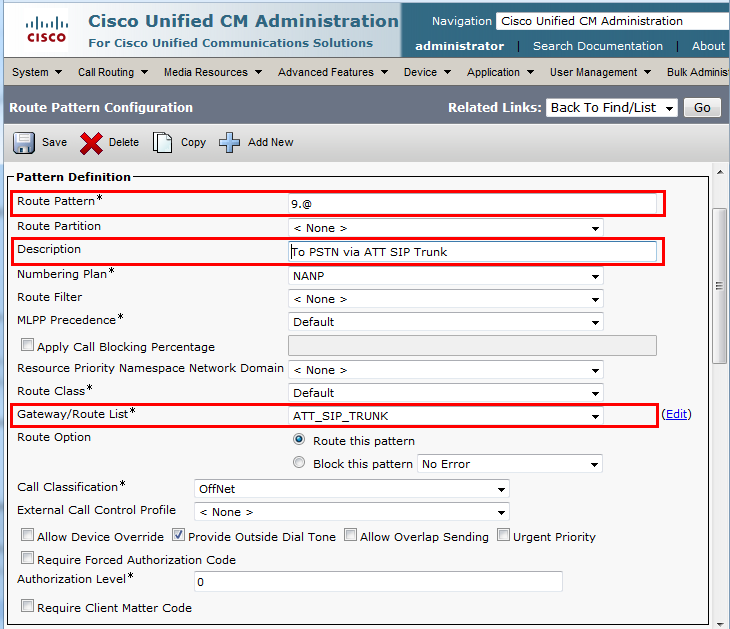
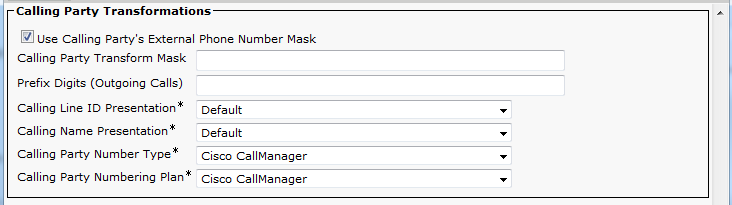
Set Description = To PSTN via ATT SIP Trunk. This text is used to identify this Route Pattern.

Set Gateway/Route List\* = ATT\_SIP\_TRUNK. This is used for this example.

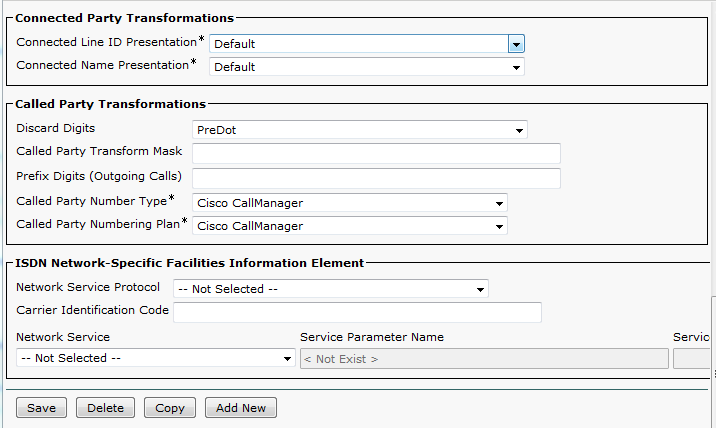
All other values are default



Route Pattern Configuration (Continued…)

Route Pattern Configuration (Continued…)



Route Pattern Configuration (Continued…)

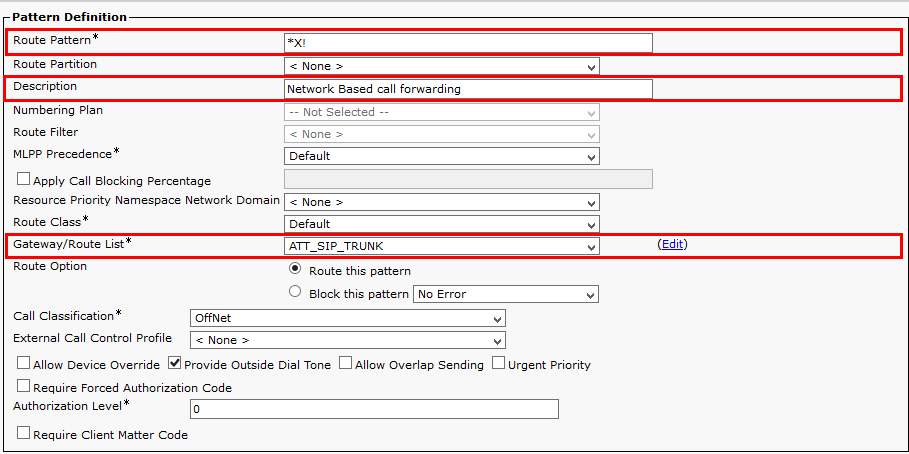
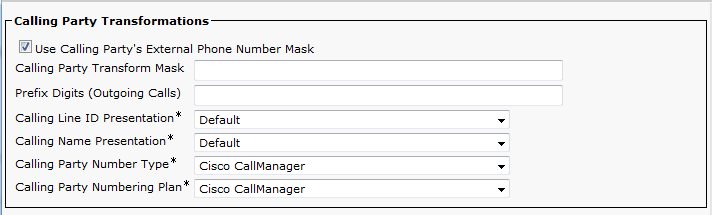
Set Route Pattern\* = \*X! This is used to route to AT&T via ISR Cisco UBE.

Set Description = Network-Based Call Forwarding. This text is used to identify this Route Pattern.

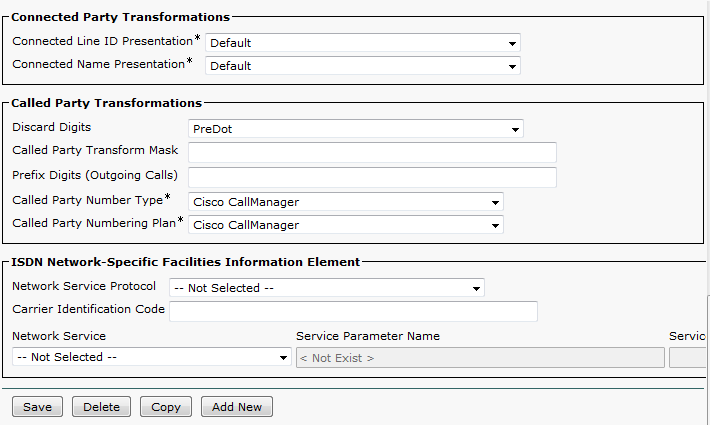
Set Gateway/Route List\* = ATT\_SIP\_TRUNK. This is used for this example.

All other values are default

Note: This Route pattern is used to Activate/De-activate Network Based Call Forwarding Features.

Route Pattern Configuration (Continued…)



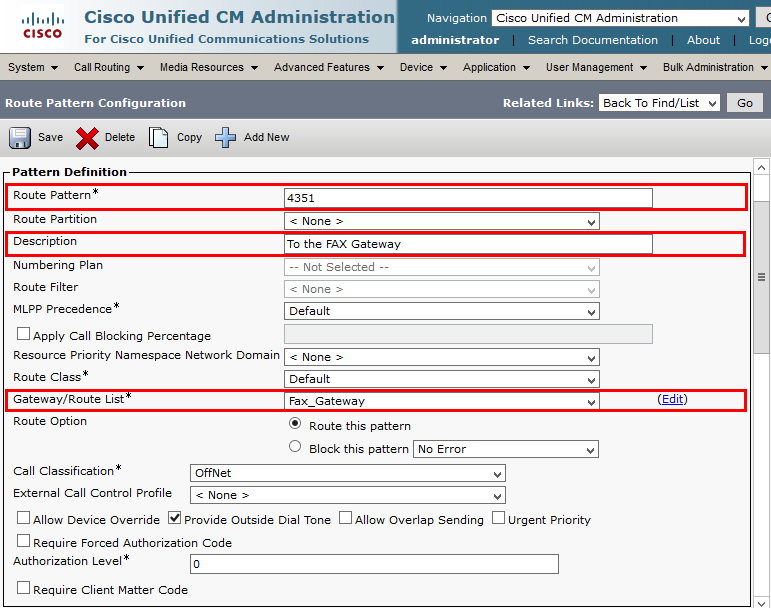
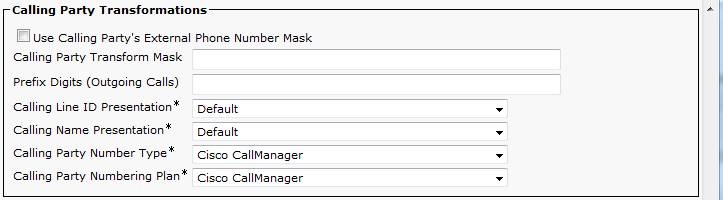
Route Pattern Configuration (Continued…)

Set Route Pattern\* = 4351 this is used to route to Fax Client via Fax Gateway.

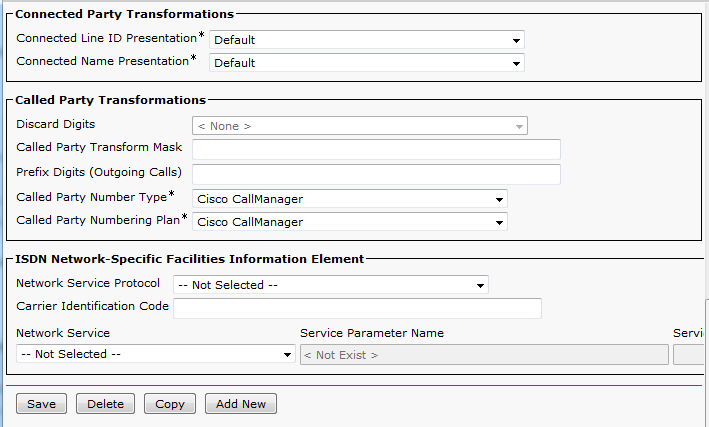
Set Description = To FAX. This text is used to identify this Route Pattern.

Set Gateway/Route List\* = Trunk\_SIP\_FAX\_Gateway. This is used for this example.

All other values are default

Route Pattern Configuration (Continued…)



## Jabber Client Configuration

**Navigation:** Device 🡪 Phone

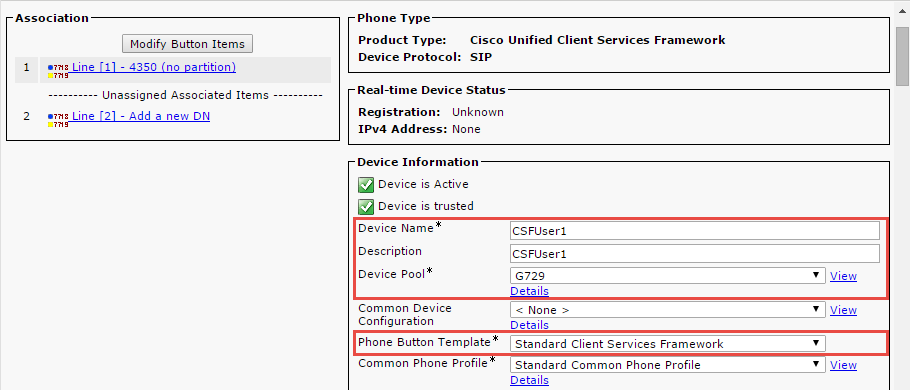
Select Phone Type\* = Cisco Unified Client services framework

Set Device Name\* = CSFUser1. This is used in this example.

Set Description = CSFUser1. This is used in this example.

Select Device Pool = G729. This is used in this example.

Select Phone Button Template\* = Standard Client Services Framework.

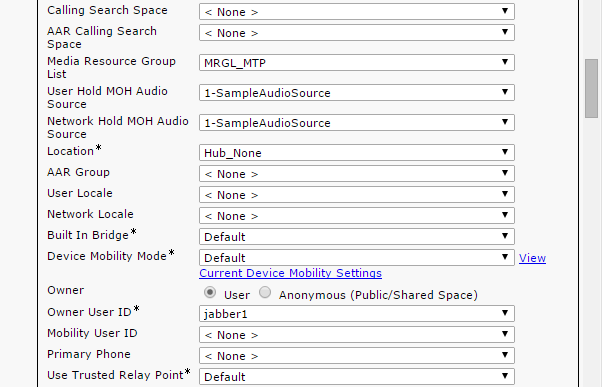


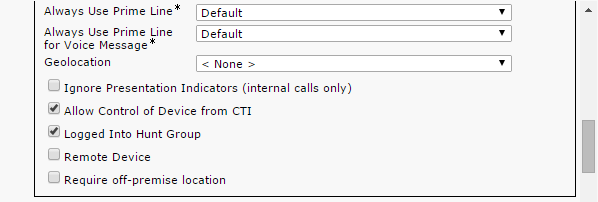
Jabber Client Configuration (Contd…)

Media Resource Group List = MRGL\_MTP

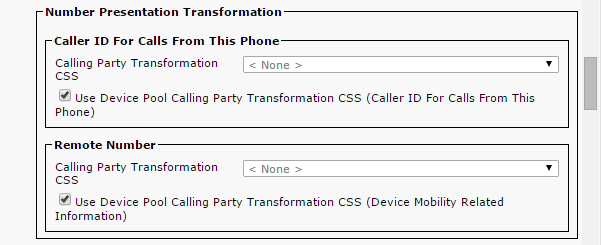
Set Owner check box

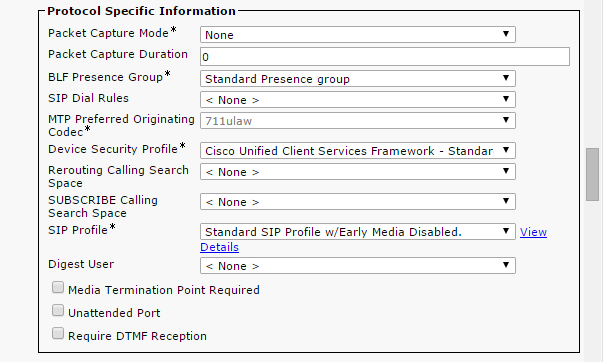
Set Owner user ID\* = jabber1. This is used for this example



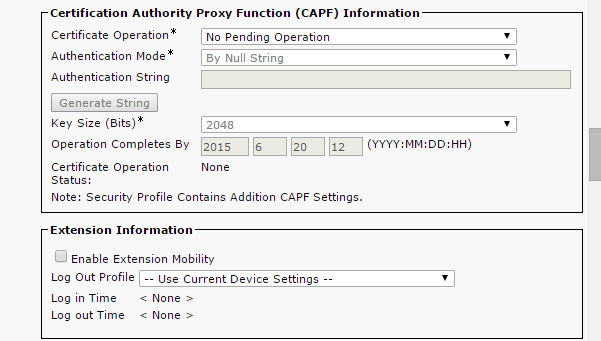


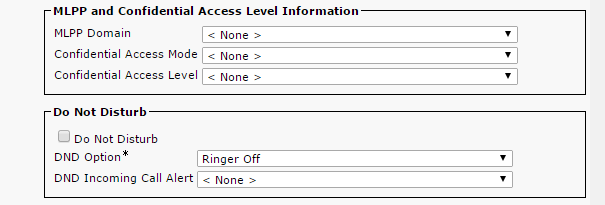
Jabber Client Configuration (Contd…)



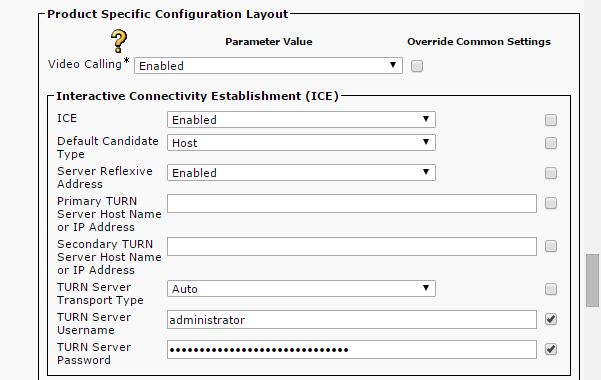


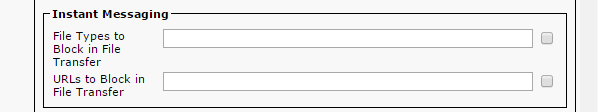
Jabber Client Configuration (Contd…)



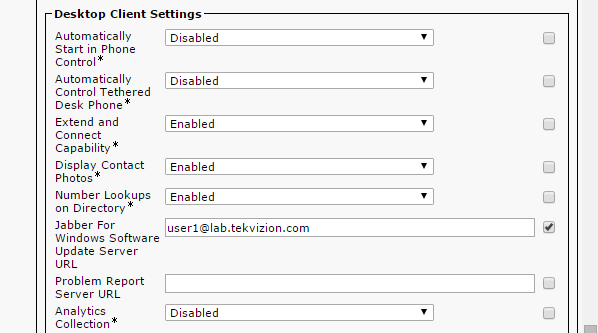


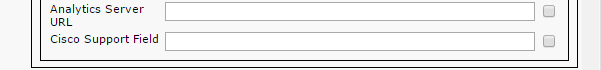
Jabber Client Configuration (Contd…)





Jabber Client Configuration (Contd…)

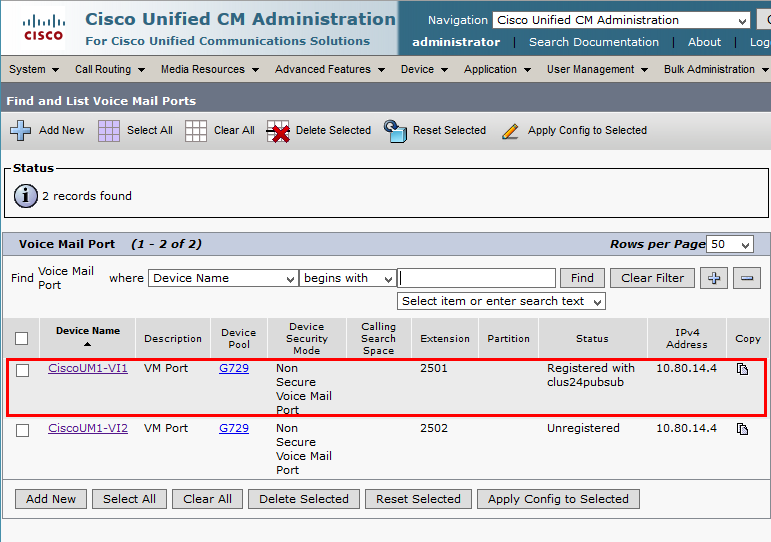






## Voicemail Port Configuration

**Navigation:** Advanced Feature🡪 Voice Mail 🡪 Cisco Voice Mail Port



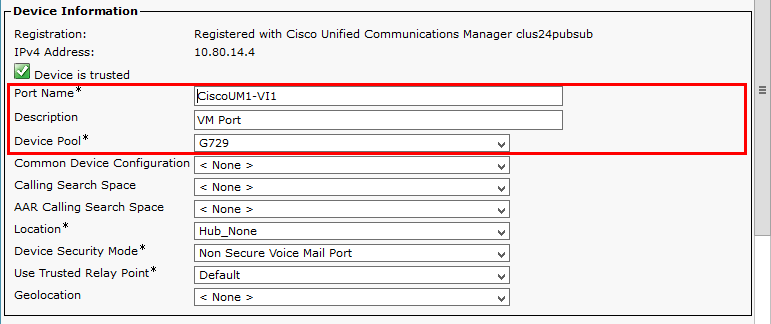
Voicemail Port Configuration (Continued…)

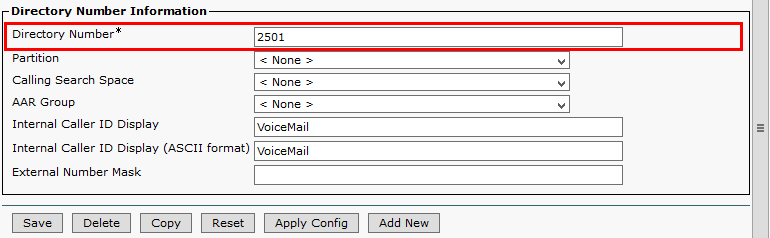
Set Port Name = CiscoUM1-VI1. This is used for this example.

Set Description = VM Port. This is used for this example.

Set Device Pool = G729

Set Directory Number\* = 2501. This is used in this example.



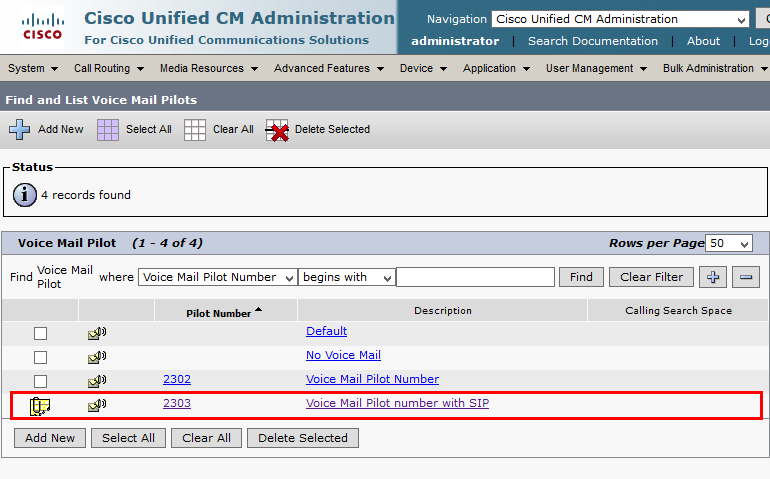
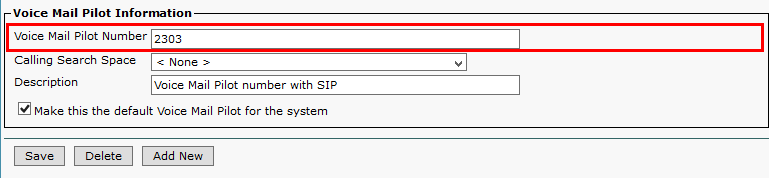


## Voicemail Pilot Configuration

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

Set Voice mail Pilot Number = 2303. This is used for this example

Set Description = VoiceMail Pilot number with SIP

# FAX Gateway Configuration

voice service voip

no ip address trusted authenticate

allow-connections sip to sip

redirect ip2ip

fax protocol pass-through g711ulaw

no fax-relay sg3-to-g3

sip

midcall-signaling passthru

g729 annexb-all

voice class codec 1

codec preference 1 g711ulaw

codec preference 2 g729r8

voice class sip-profiles 1

response ANY sip-header Allow-Header modify "UPDATE," ""

request ANY sip-header Allow-Header modify "UPDATE," ""

response ANY sip-header Allow-Header modify "UPDATE," ""

response ANY sip-header Allow-Header modify "UPDATE," ""

voice-port 0/0/1

ring frequency 50

no echo-cancel enable

no vad

cptone IN

description \*\*telephone analog/fax\*\*

station-id name fax test

station-id number 4351

caller-id enable

dial-peer voice 101 pots

huntstop

service session

destination-pattern 4351

no digit-strip

port 0/0/1

forward-digits all

dial-peer voice 200 voip

description CUCM to Gateway

service session

session protocol sipv2

session transport udp

incoming called-number 4351

voice-class codec 1

voice-class sip profiles 1

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none no vad

dial-peer voice 201 voip

description Gateway to CUCM

service session

destination-pattern [2-9]T

session protocol sipv2

session target ipv4:10.80.14.2

session transport udp

voice-class codec 1

voice-class sip profiles 1

dtmf-relay rtp-nte

no fax-relay sg3-to-g3

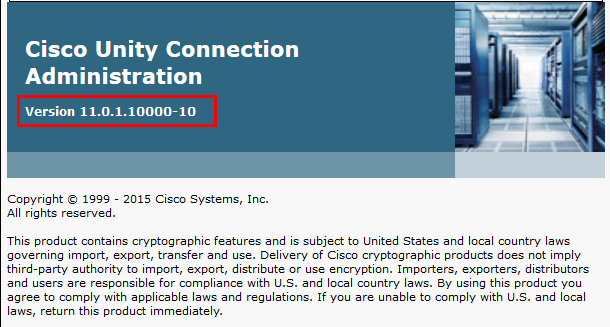
fax rate 14400

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

no vad

# Cisco UCM SIP Integration with Cisco Unity Connection (CUC)

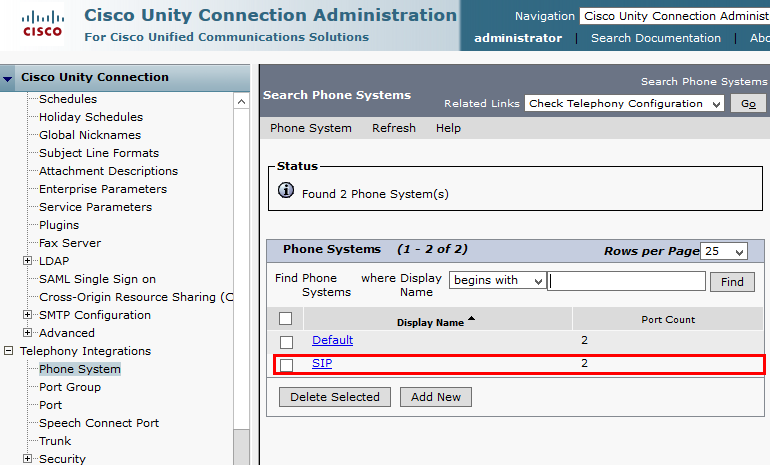
## CUC Version



## CUC Telephony Integration with Cisco UCM

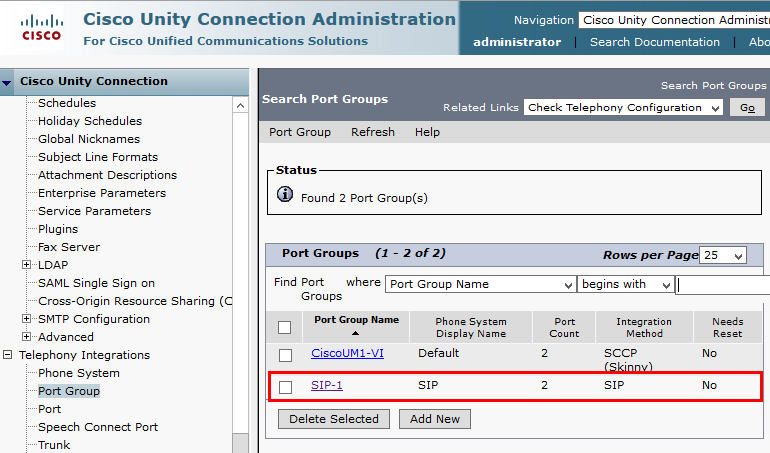
**Navigation:** Telephony Integrations 🡪 Phone system

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



## CUC Port Group

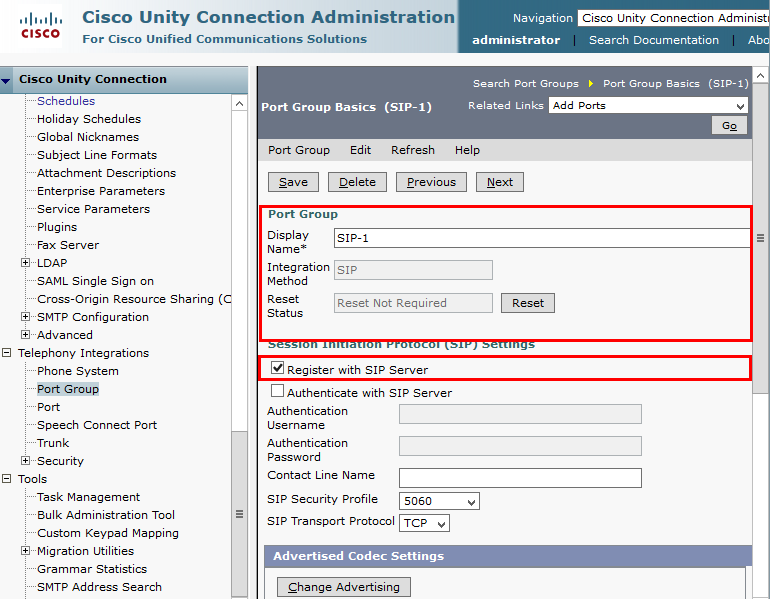
**Navigation:** Telephony Integration 🡪 Port Group



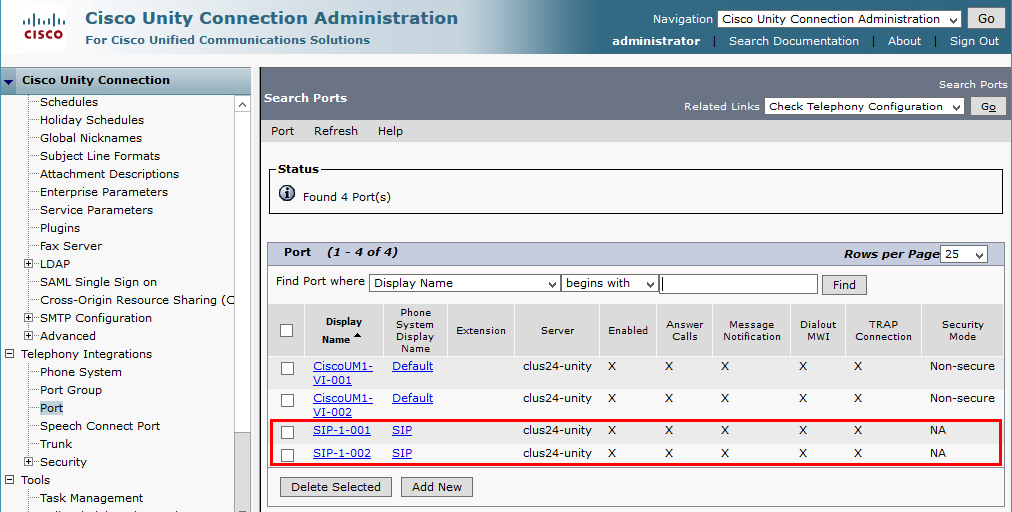
CUC Port Group(continued…)

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

Check Register with SIP Server.



## CUC Port Settings

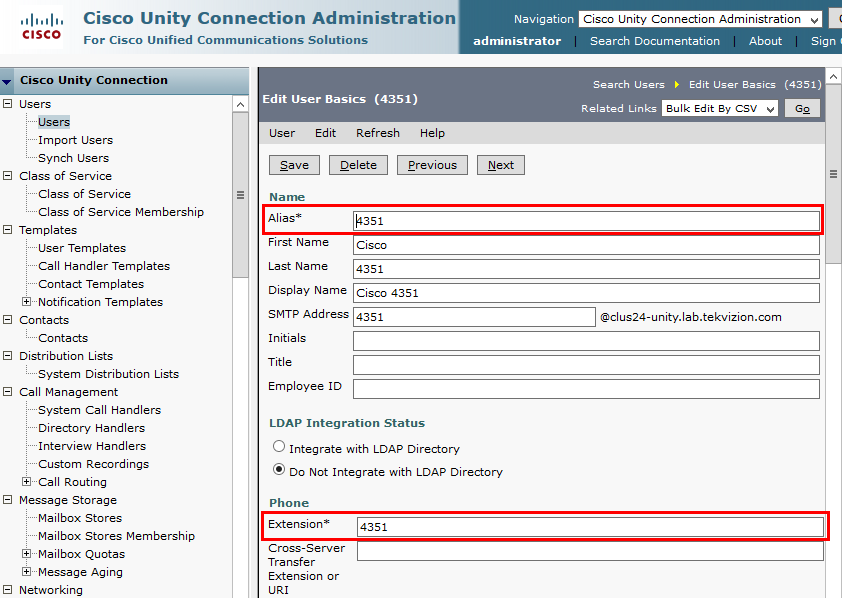


## CUC Sample User Basic Settings

**Navigation:** Cisco Unity connection 🡪 Users 🡪 Users

Set Alias = 4051.This is one of the extension used for this testing.

Set Extension = 4051. This is used for this example.

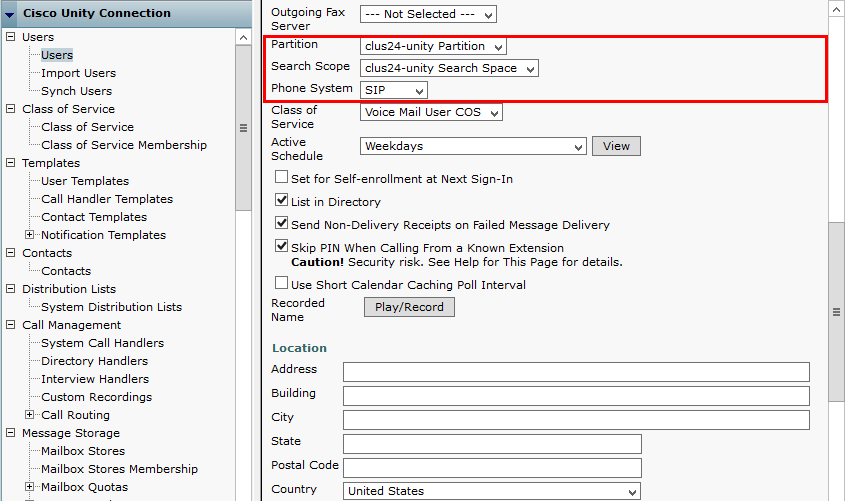


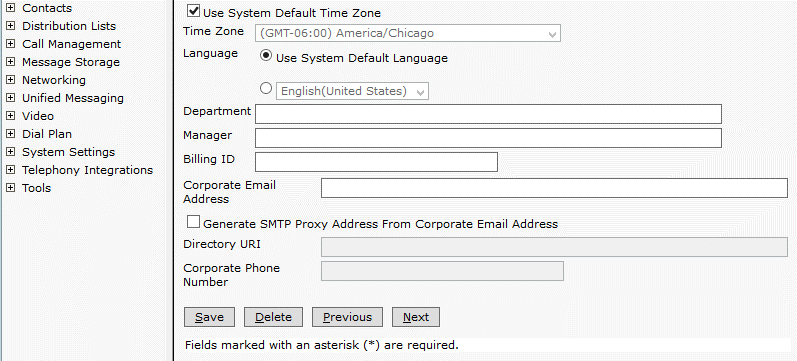
CUC Sample User Basic Settings (Continued…)

Set Partition = clus24-unity partition. This is used for this example.

Select Search Scope = clus24-unity Search Scope.

Select Phone System = SIP.





## Auto Attendant

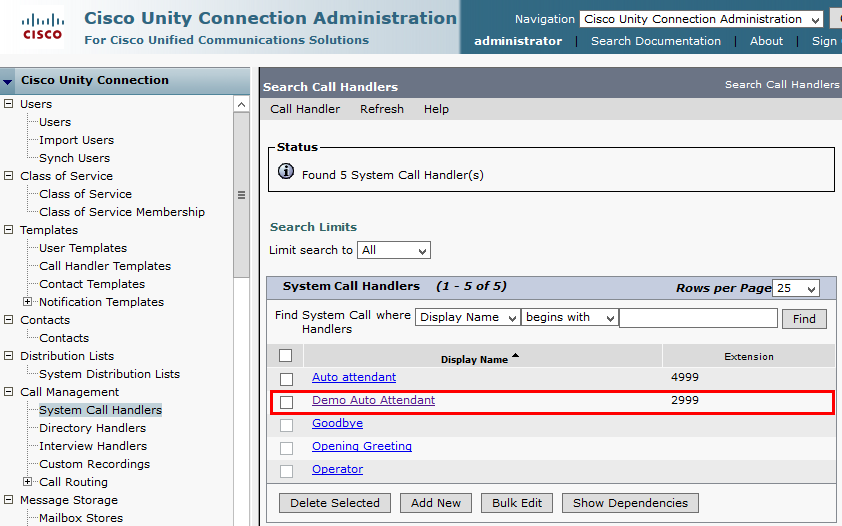
**Navigation:** Call Management 🡪 System Call Handlers

Set Display Name = Demo auto attend. This is used for this example.

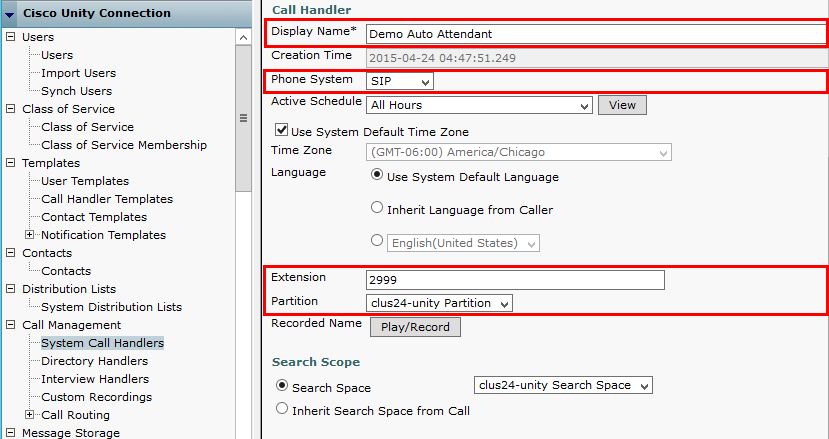
Set Phone System = SIP

Set Extension=2999. This number is used as Auto attendant on this set up.

Set Partition = Clus24-unity Partition. This is used for this example.

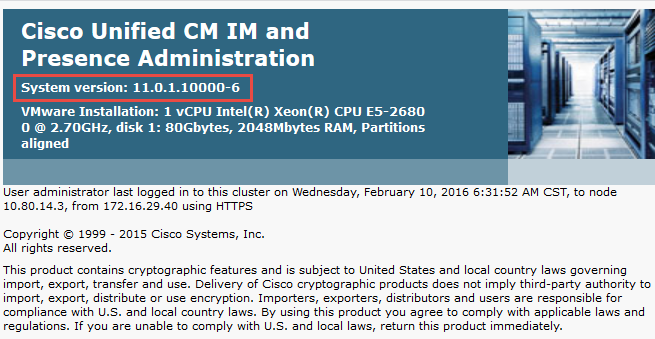


Auto Attendant (Continued…)



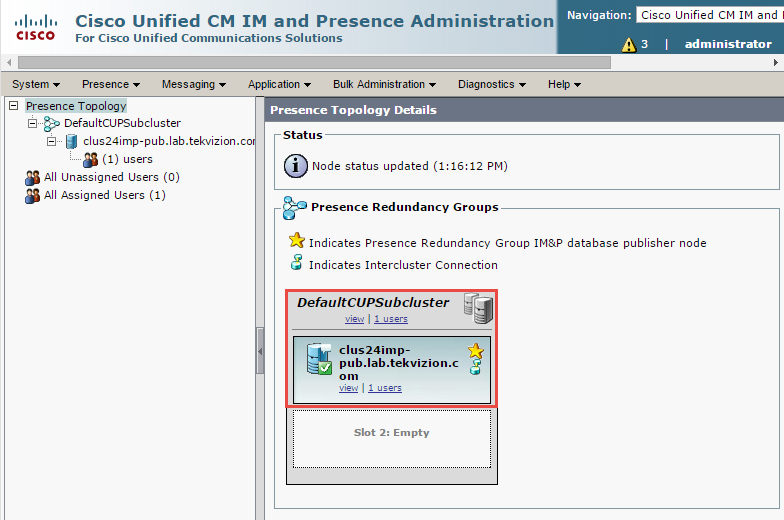
# Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP)

## CUP/IMP Version



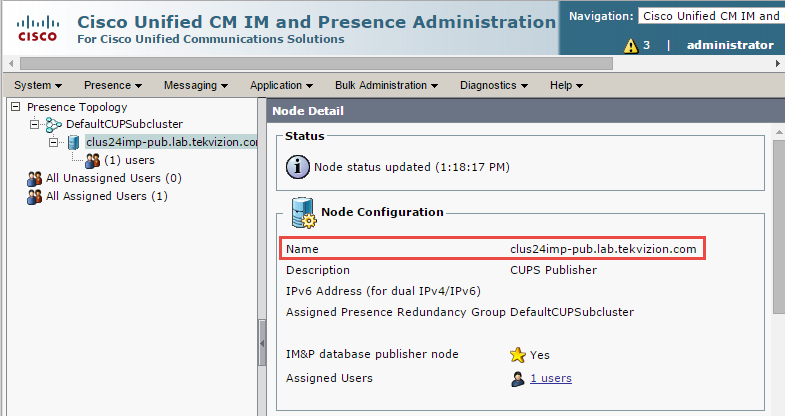
## Presence Topology

**Navigation:** System 🡪 Presence Topology



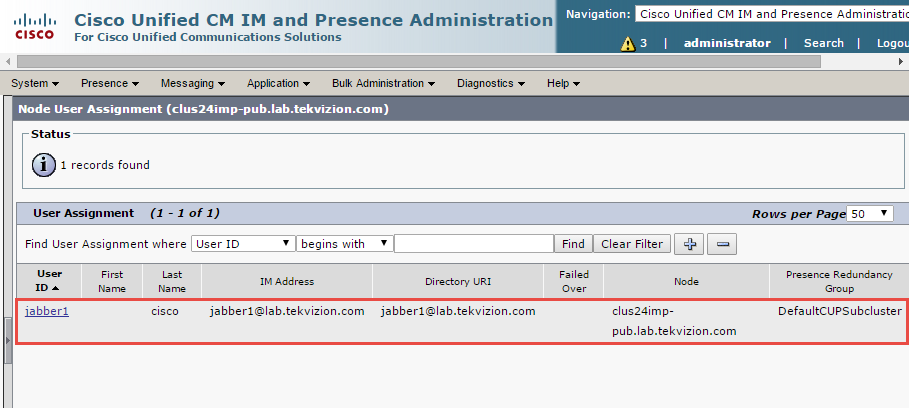
## Node Configuration

**Navigation:** System 🡪 Presence Topology 🡪 Fully Qualified Domain Name



## Users

Navigation: System 🡪 Cluster Topology 🡪 clus24imp.lab.tekvizion.com 🡪 Users



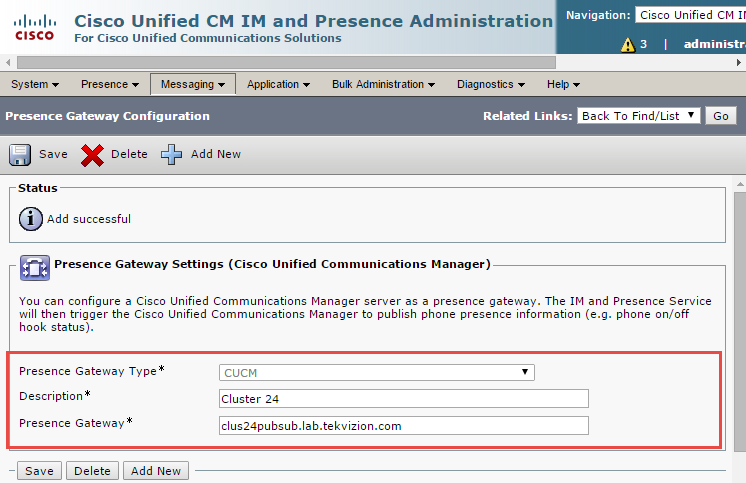
## Presence gateway configuration

**Navigation:** Presence 🡪 Gateways

Set Presence Gateway Type \*= CUCM

Set Description \*= Cluster 24. This is used for this example.

Presence Gateway \* =clus24pubsub.lab.tekvizion.com



# Acronyms

|  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |  |
| --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- | --- |
| |  |  | | --- | --- | | AVPN | AT&T Virtual Private Network | | CODEC | Coder-Decoder (in this document a device used to digitize and undigitize voice signals) | | Cisco UBE | Cisco Unified Border Element | | Cisco UCM | Cisco Unified Communications Manager | | IP | Internet Protocol | | ISR | Integrated Services Router | | MGCP | Media Gateway Control Protocol | | MIS | Managed Internet Services | | PNT | Private Network Transport | | PSTN | Public switched telephone network | | SCCP | Skinny Client Control Protocol | | SIP | Session Initiation Protocol | | SP | Service Provider | | TDM | Time-division multiplexing | |  |
|  |  |
|  |  |
|  |  |
|  |  |
|  |  |

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1. Hide signaling and media peer addresses from endpoints other than gateway. [↑](#footnote-ref-1)
2. If the mode border-element command is not entered, border-element-related commands are not available for Cisco Unified Border Element voice connections on the Cisco 2900 and Cisco 3900 series platforms with a universal feature set. The mode border-element command is not available on any other platforms. [↑](#footnote-ref-2)
3. This command enables Cisco UBE basic IP-to-IP voice communication feature. [↑](#footnote-ref-3)
4. This command allows SIP error messages to pass-through end-to-end without modification through Cisco UBE [↑](#footnote-ref-4)
5. This command enables delay offer-to-early offer conversion of initial SIP INVITE message to calls matched to this dial-peer level. [↑](#footnote-ref-5)
6. This command must be enabled at a global level to maintain integrity of SIP signaling between AT&T network and Cisco Unified Communications Manager (Cisco UCM) across Cisco UBE. [↑](#footnote-ref-6)
7. This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. This command can either be set at a global level, such as in this example, or it can be set at the dial-peer level. [↑](#footnote-ref-7)
8. This command configures the codec preference to be assigned to dial-peers. Alternatively, single code can be configured into individual dial-peers. [↑](#footnote-ref-8)
9. This SIP profile expands the Diversion header number from a 4-digit extension to a full 10-digit DID number in order to obtain interoperability with AT&T’s served users during call-forward scenarios. The six digits in "sip: 732216" are variable and must be replaced with the first 6 digits of the DID's provisioned for the customer site. [↑](#footnote-ref-9)
10. Cisco 6900-series IP phones use ptime value of 20 ms. AT&T networks prefer ptime value of 30 ms. This SIP profile modifies SDP ptime value from 20 to 30 ms and it should be applied to dial-peers where G729 is the preferred codec.  If the customer creates a dial-peer specifically for G711, a sip-profile without modifying the ptime value should be applied. This is because G711 RTP was not defaulting to 20ms. [↑](#footnote-ref-10)
11. This SIP profile is required in order to advertise the ptime=30 attribute in the outgoing SIP INVITE from Cisco UBE to AT&T. Currently RFC’s do not have a standard method to advertise ptime values for each offered codec within a SDP offering with multiple codecs. This SIP profile allows for Cisco UBE to include the ptime attribute with a value of 30ms. [↑](#footnote-ref-11)
12. WAN interface to AT&T [↑](#footnote-ref-12)
13. LAN interface to Cisco UCM [↑](#footnote-ref-13)
14. Cisco UBE LAN interface IPv4 Address [↑](#footnote-ref-14)
15. Dial peer for AT&T facing network [↑](#footnote-ref-15)
16. Session protocol SIPv2 is used for this testing [↑](#footnote-ref-16)
17. Assigns voice class codec 1 settings to dial-peer (codec support and filtering). [↑](#footnote-ref-17)
18. Configures the dynamic SIP asymmetric payload support. [↑](#footnote-ref-18)
19. This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. In this example, the command is set at the dial-peer level, you can also set the command at a global level to affect all dial-peers without necessarily setting the command on each dial-peer. [↑](#footnote-ref-19)
20. This command enables the dial peer to use SIP profile 1 [↑](#footnote-ref-20)
21. Configure the Cisco UBE SIP messaging to use the HSRP virtual address in SIP messaging. Once HSRP is configured under the physical interface and the bind command is issued, calls to the physical IP address will fail. This is because the SIP listening socket is now bound to the virtual IP address but the signaling packets use the physical IP address, and therefore cannot be handled. [↑](#footnote-ref-21)
22. This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call. [↑](#footnote-ref-22)
23. This command enables T38 fax protocol for calls terminating on this dial-peer [↑](#footnote-ref-23)