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

FEB 2016
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EDCS# xxx Rev #
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Note: Testing was conducted in tekVizion labs
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 inbounds 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
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

- 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 Jabber client version: 11.0.0 Build 65527
- VentaFax client version: 7.3.233.582
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 endpoints. 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

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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:

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'

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<thead>
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<th>Suite</th>
<th>Suite Current</th>
<th>Type</th>
<th>Suite Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>FoundationSuiteK9_npe</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>securityk9_npe</td>
<td></td>
<td></td>
<td></td>
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<tr>
<td>datak9</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>AdvUCSuiteK9</td>
<td>None</td>
<td>None</td>
<td>None</td>
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<tr>
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<tr>
<td>cme-srst</td>
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<td></td>
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<tr>
<td>cube</td>
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</tbody>
</table>

Technology Package License Information for Module:'c2900'

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<thead>
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<th>Technology</th>
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<tbody>
<tr>
<td></td>
<td>Current</td>
<td>Type</td>
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<tr>
<td>ipbase</td>
<td>ipbasek9</td>
<td>Permanent</td>
<td>ipbasek9</td>
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<tr>
<td>security</td>
<td>None</td>
<td>None</td>
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<tr>
<td>uc</td>
<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
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</tbody>
</table>
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 Pe0NhIw5tXZpE.k5VhTSoGpcuVeRyrer9kEPz20Z6
!
no aaa new-model
ethernet lmi ce
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!
no ip domain lookup
ip cef
no ipv6 cef
multilink bundle-name authenticated
!
Note: Testing was conducted in tekVizion labs.

!
!  
!  
!  
cts logging verbose
  
!

voice-card 0
dspfarm
dsp services dspfarm
  
!

voice service voip
  
no ip address trusted authenticate
  
address-hiding¹
  
mode border-element²
  
allow-connections sip to sip³
  
redirect ip2ip
  
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
  
no fax-relay sg3-to-g3
  
sip
  
header-passing

---
¹ Hide signaling and media peer addresses from endpoints other than gateway.
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.
³ This command enables Cisco UBE basic IP-to-IP voice communication feature.
error-passthru
early-offer forced
midcall-signaling passthru
privacy-policy passthru
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

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>"

---

4 This command allows SIP error messages to pass-through end-to-end without modification through Cisco UBE
5 This command enables delay offer-to-early offer conversion of initial SIP INVITE message to calls matched to this dial-peer level.
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.
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.
8 This command configures the codec preference to be assigned to dial-peers. Alternatively, single code can be configured into individual dial-peers.
request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
request INVITE sdp-header Audio-Attribute add "a=ptime:30"

!
!
!
!
!
!
license udi pid CISCO2921/K9 sn FTX1746AJCB

hw-module pvdm 0/0
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interface GigabitEthernet0/0
  description WAN to ATT
  ip address 192.65.79.59 255.255.255.224
  duplex auto
  speed auto

interface GigabitEthernet0/1
  description LAN Interface
  ip address 10.80.14.5 255.255.0.0
  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

---

12 WAN interface to AT&T
13 LAN interface to Cisco UCM
14 Cisco UBE LAN interface IPv4 Address
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

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-nre

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


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

destination-pattern [2-9]T

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

---

15 Dial peer for AT&T facing network
16 Session protocol SIPv2 is used for this testing
17 Assigns voice class codec 1 settings to dial-peer (codec support and filtering).
18 Configures the dynamic SIP asymmetric payload support.
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles

dtmf-relay rtp-n-te
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 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
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0

---

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.

20 This command enables the dial peer to use SIP profile 1

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.

22 This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call.

23 This command enables T38 fax protocol for calls terminating on this dial-peer
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 rtpnte

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-n-te
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-n-te
no fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!

Note: Testing was conducted in tekVizion labs
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-npe
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
!

Note: Testing was conducted in tekVizion labs
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 udp tn 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 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

Note: Testing was conducted in tekVizion labs

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Note: Testing was conducted in tekVizion labs
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 Mobility Related Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Mobility Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Geolocation Configuration

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation Filter</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Call Routing Information

#### Incoming Calling Party Settings

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

#### Incoming Called Party Settings

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
Device Pool Configuration (continued...)

<table>
<thead>
<tr>
<th>Phone Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID For Calls From This Phone</td>
</tr>
<tr>
<td>Calling Party Transformation CSS: &lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Connected Party Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Party Transformation CSS: &lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Redirecting Party Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Redirecting Party Transformation CSS: &lt; None &gt;</td>
</tr>
</tbody>
</table>

- Save
- Delete
- Copy
- Reset
- Apply Config
- Add New
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_cluster24pubsub. This is used for this example
Set Device pool* = G729

![Media Termination Point Configuration](image-url)
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

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default Network Hold MOH Audio Source ID</td>
<td>1</td>
</tr>
<tr>
<td>Default User Hold MOH Audio Source ID</td>
<td>1</td>
</tr>
<tr>
<td>Duplex Streaming Enabled</td>
<td>True</td>
</tr>
<tr>
<td>Media Exchange Interface Capability Timer</td>
<td>True</td>
</tr>
<tr>
<td>Send Multicast MOH in H.245 OLC Message</td>
<td>True</td>
</tr>
<tr>
<td>Media Exchange Stop Streaming Timer</td>
<td>12</td>
</tr>
<tr>
<td>Open Video Channel Response Timer for SIP Interop</td>
<td>500</td>
</tr>
<tr>
<td>Port Received Timer After Call Connection</td>
<td>500</td>
</tr>
<tr>
<td>Media Resource Allocation Timer</td>
<td>12</td>
</tr>
<tr>
<td>MTP and Transcoder Resource Throttling Percentage</td>
<td>95</td>
</tr>
</tbody>
</table>

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.
End User Configuration (continued...)

Note: Testing was conducted in tekVizion labs
End User Configuration (continued...)

- **Service Settings**
  - Check Home Cluster
  - Enable User for Unified CM IM and Presence (Configure IM and Presence in the associated UC Service Profile)
  - Include meeting information in presence (Requires Exchange Presence Gateway to be configured on CUCM IM and Presence server)

---

- **Device Information**
  - Controlled Devices
    - CSFUser1
    - SEP00083031F2A8

---

- **Extension Mobility**
  - Available Profiles
  - Controlled Profiles

---

- **Directory Number Associations**
  - Primary Extension

---

Note: Testing was conducted in tekVizion labs
End User Configuration (continued...)

- Mobility Information
  - Enable Mobility
  - Enable Mobile Voice Access
  - Maximum Wait Time for Desk Pickup: 10000
  - Remote Destination Limit*: 4
  - Remote Destination Profiles: View Details

- Mutilevel Precedence and Preemption Authorization
  - MLPP User Identification Number
  - MLPP Password
  - Confirm MLPP Password
  - MLPP Precedence Authorization Level: Default

- CAF Information
  - Associated CAF Profiles: View Details

- Permissions Information
  - Groups: Standard Audit Users, Standard CAR Admin Users, Standard CCM Admin Users, Standard CCM End Users, Standard CCM Gateway Administration: Add to Access Control Group, Remove from Access Control Group

Save | Delete | Add New
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...)

- Remote Device
- Protected Device
- Hot line Device
- Require off-premise location

**Number Presentation Transformation**

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

- **Remote Number**
  - Calling Party Transformation CSS
  - Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

**Protocol Specific Information**

- Packet Capture Mode
- Packet Capture Duration
- BU Presence Group
- Device Security Profile
- SUBSCRIBE Calling Search Space
- Unattended Port
- Requires DTMF Reception
- RFC2933 Disabled

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

- Certificate Operation
- Authentication Mode
- Authentication String
  - Generate String
- Key Size (Bits)
- Operation Completes By
- Certificate Operation Status
- Note: Security Profile Contains Additional CAPF Settings

**Expansion Module Information**

- Module 1
- Module 1 Load Name
- Module 2
- Module 2 Load Name
Cisco IP Phone 7965 SCCP Configuration (Continued...)

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

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

### Extension Information

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

### MLPP and Confidential Access Level Information

- MLPP Domain: < None >
- MLPP Indication: Default
- MLPP Preemption: Default
- Confidential Access Mode: < None >
- Confidential Access Level: < None >

### Do Not Disturb

- Do Not Disturb: [ ]
- DND Option: Use Common Phone Profile Setting
- DND Incoming Call Alert: < None >

### Secure Shell Information

<table>
<thead>
<tr>
<th>Secure Shell User</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell Password</td>
</tr>
</tbody>
</table>
Note: Testing was conducted in tekVizion labs
Note:

Testing was conducted in tekVizion labs
<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>5</td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codec</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discovery Protocol - Media Endpoint</td>
<td></td>
</tr>
<tr>
<td>Discover (LLDP-MED): Switch Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
<td></td>
</tr>
<tr>
<td>LLDP Assoc ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td>Unknown</td>
</tr>
<tr>
<td>Wireless Headset Hookswitch Control</td>
<td>Disabled</td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td>User Controlled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td>Normal</td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
<td>0-Silent</td>
</tr>
</tbody>
</table>
Cisco IP Phone 7965 SCCP Configuration (Continued…)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Headset</td>
<td>Default</td>
</tr>
<tr>
<td>Sidetone Level</td>
<td>Default</td>
</tr>
<tr>
<td>Headset Send Gain</td>
<td>Default</td>
</tr>
<tr>
<td>HTTPS Server</td>
<td>http and https Enabled</td>
</tr>
<tr>
<td>Handset/Headset Monitor</td>
<td>Enabled</td>
</tr>
<tr>
<td>Headset Recording</td>
<td>Disabled</td>
</tr>
<tr>
<td>Enblc Dialog</td>
<td>Enabled</td>
</tr>
<tr>
<td>Switch Port Remote Configuration</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td>Disabled</td>
</tr>
<tr>
<td>SSR Access</td>
<td>Disabled</td>
</tr>
<tr>
<td>LOGIN Access</td>
<td>Enabled</td>
</tr>
<tr>
<td>FIPS Mode</td>
<td>Disabled</td>
</tr>
<tr>
<td>80-bit SRTCP</td>
<td>Disabled</td>
</tr>
<tr>
<td>Customer Support Use</td>
<td></td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
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...)

### Directory URIs

<table>
<thead>
<tr>
<th>Primary</th>
<th>URI</th>
<th>Partition</th>
<th>Advertise Globally via ILS</th>
<th>Remove</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Add Row

### PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number: < None >

### AAR Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
<th>Forward: or</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Retain this destination in the call forwarding history

### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Calling Search Space Activation Policy</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Forward All: or</td>
<td></td>
<td>Use System Default</td>
</tr>
<tr>
<td></td>
<td>Secondary Calling Search Space for Forward All:</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>Forward Busy Internal: or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>Forward Busy External: or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>Forward No Answer Internal: or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>Forward No Answer External: or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>Forward No Coverage Internal: or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td></td>
<td>Forward No Coverage External: or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
Cisco IP Phone 7965 SCCP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Forward on CTI Failure</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward Unregistered Internal</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>No Answer Ring Duration (seconds)</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Park Monitoring**

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>A blank value means to call the parker's line.</td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>A blank value means to call the parker's line.</td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>A blank value will use value set in Park Monitoring</td>
</tr>
</tbody>
</table>

**MLPP Alternate Party And Confidential Access Level Settings**

<table>
<thead>
<tr>
<th>Target (Destination)</th>
<th>MLPP Calling Search Space</th>
<th>MLPP No Answer Ring Duration (seconds)</th>
<th>Confidential Access Mode</th>
<th>Confidential Access Level</th>
<th>Call Control Agent Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Line Settings for All Devices**

| Hold Reversion Ring Duration (seconds) | Setting the Hold Reversion Ring Duration to zero will disable the feature |
| Hold Reversion Notification Interval (seconds) | Setting the Hold Reversion Notification Interval to zero will disable the feature |
| Party Entrance Tone | Default |
Cisco IP Phone 7965 SCCP Configuration (Continued...)
### Line 1 on Device SEP00083031F5D4

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
<th>Notes</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Display (Caller ID)</strong></td>
<td>Cisco7965 - Phone 1</td>
<td>Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.</td>
</tr>
<tr>
<td><strong>ASCII Display (Caller ID)</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td><strong>Line Text Label</strong></td>
<td>Cisco7965 - Phone 1</td>
<td></td>
</tr>
<tr>
<td><strong>External Phone Number Mask</strong></td>
<td>7323204351</td>
<td></td>
</tr>
<tr>
<td><strong>Visual Message Waiting Indicator Policy</strong></td>
<td>Use System Policy</td>
<td></td>
</tr>
<tr>
<td><strong>Audible Message Waiting Indicator Policy</strong></td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Idle)</strong></td>
<td>Use System Default</td>
<td>Applies to this line when any line on the phone has a call in progress.</td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Active)</strong></td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td><strong>Call Pickup Group Audio Alert Setting (Phone Idle)</strong></td>
<td>Use System Default</td>
<td></td>
</tr>
</tbody>
</table>

### Call Pickup Group Audio Alert Setting (Phone Active)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Recording Option</strong></td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td><strong>Recording Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Recording Media Source</strong></td>
<td>Gateway Preferred</td>
</tr>
<tr>
<td><strong>Monitoring Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- Log Missed Calls

---

Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7965 SCCP Configuration (Continued…)

- **Multiple Call/Call Waiting Settings on Device SEP44ADD9D56F39**
  
  **Note:** The range to select the Max Number of calls is: 1-200

  **Maximum Number of Calls**
  
  [4]

  **Busy Trigger**
  
  [2] (Less than or equal to Max. Calls)

- **Forwarded Call Information Display on Device SEP44ADD9D56F39**
  
  - [ ] Caller Name
  - [ ] Caller Number
  - [ ] Redirected Number
  - [x] Dialed Number

- **Users Associated with Line**
  
  [ ] Associate End Users

  - [Save] [Delete] [Reset] [Apply Config] [Add New]
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...)

**Protocol Specific Information**

- Packet Capture Mode: None
- Packet Capture Duration: 0
- BLF Presence Group: Standard Presence Group
- **Device Security Profile**: Cisco 7975 - Standard SCCP Non-Secure Profile
- SUBSCRIBE Calling Search Space: < None >

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

- Certificate Operation: No Pending Operation
- Authentication Mode: By Null String
- Authentication String: Generate String
- Key Size (Bits): 2048
- Operation Completes By: 2015-03-27 12:00
- Certificate Operation Status: None
- Note: Security Profile Contains Addition CAPF Settings.

**Expansion Module Information**

- Module 1 Load Name: < None >
- Module 2 Load Name: < None >
- Module 3 Load Name: < None >
Cisco IP Phone 7975 SCCP Configuration (Continued…)

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

<table>
<thead>
<tr>
<th>Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory</td>
<td></td>
</tr>
<tr>
<td>Messages</td>
<td></td>
</tr>
<tr>
<td>Services</td>
<td></td>
</tr>
<tr>
<td>Authentication Server</td>
<td></td>
</tr>
<tr>
<td>Proxy Server</td>
<td></td>
</tr>
<tr>
<td>Idle</td>
<td></td>
</tr>
<tr>
<td>Idle Timer (seconds)</td>
<td></td>
</tr>
<tr>
<td>Secure Authentication URL</td>
<td></td>
</tr>
<tr>
<td>Secure Directory URL</td>
<td></td>
</tr>
<tr>
<td>Secure Idle URL</td>
<td></td>
</tr>
<tr>
<td>Secure Information URL</td>
<td></td>
</tr>
<tr>
<td>Secure Messages URL</td>
<td></td>
</tr>
<tr>
<td>Secure Services URL</td>
<td></td>
</tr>
</tbody>
</table>

### Extension Information

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

### MLPP and Confidential Access Level Information

| MLPP Domain | `<None>` |
| MLPP Indication* | Default |
| MLPP Preemption* | Default |
| Confidential Access Mode | `<None>` |
| Confidential Access Level | `<None>` |

### Do Not Disturb

- **Do Not Disturb**
- **DND Option** — Use Common Phone Profile Setting
- **DND Incoming Call Alert** — `<None>`
### Secure Shell Information

- **Secure Shell User**: administrator
- **Secure Shell Password**: **************

### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>PC Port *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Back USB Port*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Side USB Port*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Camera*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Console Access*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Video Capabilities*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Enable/Disable USB Classes</td>
<td>Mass Storage</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Human Interface Device</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Audio Class</td>
<td></td>
</tr>
<tr>
<td>SDIO *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Bluetooth *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Wifi *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Bluetooth Profiles*</td>
<td>Handsfree</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Human Interface Device</td>
<td></td>
</tr>
<tr>
<td>Settings Access*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Web Access*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Show All Calls on Primary Line*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Sunday</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Monday</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Tuesday</td>
<td></td>
</tr>
<tr>
<td>Setting</td>
<td>Setting Value</td>
<td></td>
</tr>
<tr>
<td>------------------------------</td>
<td>--------------------------------</td>
<td></td>
</tr>
<tr>
<td>Display On Time</td>
<td>07:30</td>
<td></td>
</tr>
<tr>
<td>Display On Duration</td>
<td>10:30</td>
<td></td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td>01:00</td>
<td></td>
</tr>
<tr>
<td>HTTPS Server*</td>
<td>http and https Enabled</td>
<td></td>
</tr>
<tr>
<td>Enable Power Save Plus</td>
<td>Sunday, Monday, Tuesday</td>
<td></td>
</tr>
<tr>
<td>Phone On Time</td>
<td>00:00</td>
<td></td>
</tr>
<tr>
<td>Phone Off Time</td>
<td>24:00</td>
<td></td>
</tr>
<tr>
<td>Phone Off Idle Timeout*</td>
<td>60</td>
<td></td>
</tr>
<tr>
<td>Enable Audible Alert</td>
<td></td>
<td></td>
</tr>
<tr>
<td>EnergyWise Domain</td>
<td></td>
<td></td>
</tr>
<tr>
<td>EnergyWise Endpoint Secret</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Allow EnergyWise Overrides</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Span to PC Port*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Logging Display*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Recording Tone*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Local Volume*</td>
<td>100</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Remote Volume*</td>
<td>50</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>RTCP*</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Remote Log*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Log Profile</td>
<td>Default, Preset, Telephony</td>
<td></td>
</tr>
</tbody>
</table>
Cisco IP Phone 7975 SCCP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Advertise G.722 and iSAC Codecs*</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Wideband Handset UI Control*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Handset*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP); Switch Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP); PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint</td>
<td>Disabled</td>
</tr>
<tr>
<td>Discover (LLDP-MED); Switch Port*</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP); PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority*</td>
<td>Unknown</td>
</tr>
<tr>
<td>802.1x Authentication*</td>
<td>User Controlled</td>
</tr>
<tr>
<td>FIPS Mode*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>Normal</td>
</tr>
<tr>
<td>Switch Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Power Negotiation*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Restrict Data Rates*</td>
<td>Disabled</td>
</tr>
<tr>
<td>SSH Access*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Incoming Call Toast Timer*</td>
<td>5</td>
</tr>
<tr>
<td>Provide Dial Tone from Release Button*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
### Cisco IP Phone 7975 SCCP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hide Video By Default*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Background Image</td>
<td></td>
</tr>
<tr>
<td>Simplified New Call UI*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Enable VXC VPN for MAC</td>
<td></td>
</tr>
<tr>
<td>VXC VPN Option*</td>
<td>Dual Tunnel</td>
</tr>
<tr>
<td>VXC Challenge*</td>
<td>Challenge</td>
</tr>
<tr>
<td>VXC-M Servers</td>
<td></td>
</tr>
<tr>
<td>Revert to All Calls*</td>
<td>Disabled</td>
</tr>
<tr>
<td>RTCP for Video*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Record Call Log from Shared Line*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Show Remote Private Calls*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Record Call Log For Remote Private Calls*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Show Call History for Selected Line Only*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Actionable</td>
<td>Disabled</td>
</tr>
<tr>
<td>Incoming Call Alert*</td>
<td></td>
</tr>
<tr>
<td>DF bit*</td>
<td>0</td>
</tr>
<tr>
<td>Default Line Filter</td>
<td></td>
</tr>
<tr>
<td>Separate Audio and Video Mute*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Softkey Control*</td>
<td>Feature Control Policy</td>
</tr>
<tr>
<td>Start Video Port</td>
<td></td>
</tr>
<tr>
<td>Stop Video Port</td>
<td></td>
</tr>
<tr>
<td>Lowest Alerting Line State Priority*</td>
<td>Disabled</td>
</tr>
<tr>
<td>TLS Resumption Timer*</td>
<td>3600</td>
</tr>
<tr>
<td>Audio EQ*</td>
<td>Default : Default</td>
</tr>
</tbody>
</table>

---

Note: Testing was conducted in tekVizion labs
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...)

### Directory URIs

<table>
<thead>
<tr>
<th>Primary</th>
<th>URI</th>
<th>Partition</th>
<th>Advertise Globally via ILS</th>
<th>Remove</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

[Add Row]

### PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

Advertised Failover Number: <None>

### AAR Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Retain this destination in the call forwarding history</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space Activation Policy</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Forward All</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>or</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>
Cisco IP Phone 7975 SCCP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Forward on CTI Failure</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward Unregistered Internal</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

No Answer Ring Duration (seconds)

Call Pickup Group | < None > |

---

**Park Monitoring**

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>A blank value means to call the parker's line.</td>
</tr>
<tr>
<td>Forward No Retrieve Destination External</td>
<td>&lt; None &gt;</td>
<td>A blank value means to call the parker's line.</td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>A blank value will use value set in Park Monitoring</td>
</tr>
</tbody>
</table>

Reversion Timer: Reversion Timer service parameter

---

**MLPP Alternate Party And Confidential Access Level Settings**

Target (Destination)

MLPP Calling Search Space | < None > |

MLPP No Answer Ring Duration (seconds)

Confidential Access Mode | < None > |

Confidential Access Level | < None > |

Call Control Agent Profile | < None > |

---

**Line Settings for All Devices**

Hold Reversion Ring Duration (seconds) | Setting the Hold Reversion Ring Duration to zero will disable the feature |

Hold Reversion Notification Interval (seconds) | Setting the Hold Reversion Notification Interval to zero will disable the feature |

Party Entrance Tone | Default |
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...)

<table>
<thead>
<tr>
<th>Display (Caller ID)</th>
<th>Cisco7975 - Phone 1</th>
<th>Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASCII Display (Caller ID)</td>
<td>Cisco7975 - Phone 1</td>
<td></td>
</tr>
<tr>
<td>Line Text Label</td>
<td>Cisco7975 - Phone 1</td>
<td></td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td>73232024350</td>
<td></td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy*</td>
<td>Use System Policy</td>
<td></td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)*</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
<td>Applies to this line when any line on the phone has a call in progress.</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Recording Option*</td>
<td>Call Recording Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Recording Media Source*</td>
<td>Gateway Preferred</td>
<td></td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Log Missed Calls</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
### Multiple Call/Call Waiting Settings on Device SEP00083031F2A8

Note: The range to select the Max Number of calls is: 1-200

<table>
<thead>
<tr>
<th>Maximum Number of Calls *</th>
<th>4</th>
</tr>
</thead>
<tbody>
<tr>
<td>Busy Trigger *</td>
<td>2</td>
</tr>
</tbody>
</table>

### Forwards Call Information Display on Device SEP00083031F2A8

- [x] Caller Name
- [ ] Caller Number
- [ ] Redirected Number
- [x] Dialed Number

### Users Associated with Line

- [ ] Associate End Users

---

**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...)
<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>28</td>
<td>Alerting Calls</td>
</tr>
<tr>
<td>29</td>
<td>Queue Status</td>
</tr>
<tr>
<td>30</td>
<td>Privacy</td>
</tr>
<tr>
<td>31</td>
<td>None</td>
</tr>
</tbody>
</table>

### Built In Bridge
- Default

### Privacy
- Default

### Device Mobility Mode
- Default

#### Current Device Mobility Settings
- User: Anonymous (Public/Shared Space)

#### Phone Load Name
- Default

#### Use Trusted Relay Point
- Default

#### BLF Audible Alert Setting (Phone Idle)
- Default

#### BLF Audible Alert Setting (Phone Busy)
- Default

#### Always Use Prime Line
- Default

#### Always Use Prime Line For Voice Message
- Default

#### Geolocation
- None

#### Number Presentation Transformation

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

#### Remote Number
- Calling Party Transformation CSS
- Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

---

Cisco IP Phone 9971 SIP Configuration (Continued...)
### Cisco IP Phone 9971 SIP Configuration (Continued...)

#### Protocol Specific Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode*</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>BLF Presence Group*</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>SIP Dial Rules</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MTP Preferred Originating Codecs*</td>
<td>71ulaw</td>
</tr>
<tr>
<td><strong>Device Security Profile</strong></td>
<td>Cisco 9971 - Standard SIP Non-Secure Profile</td>
</tr>
<tr>
<td>Rerouting Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Profile*</td>
<td>Standard SIP Profile for AT&amp;T View Details</td>
</tr>
<tr>
<td>Digest User</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td></td>
</tr>
<tr>
<td>Unattended Port</td>
<td></td>
</tr>
<tr>
<td>Require DTMF Reception</td>
<td></td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Operation*</td>
<td>No Pending Operation</td>
</tr>
<tr>
<td>Authentication Mode*</td>
<td>By Null String</td>
</tr>
<tr>
<td>Authentication String</td>
<td></td>
</tr>
<tr>
<td>Key Size (Bits)*</td>
<td>2048</td>
</tr>
<tr>
<td>Operation Completes By</td>
<td>2015 06 20 12 (YYYY:MM:DD:HH)</td>
</tr>
<tr>
<td>Certificate Operation Status:</td>
<td>None</td>
</tr>
</tbody>
</table>

Note: Security Profile contains additional CAPF settings.

#### Expansion Module Information

<table>
<thead>
<tr>
<th>Module</th>
<th>Load Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>2</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>3</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>4</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
### External Data Locations Information (Leave blank to use default)

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Extension Information

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

### MLPP and Confidential Access Level Information

- **MLPP Domain**: < None >
- **MLPP Indication**: Default
- **MLPP Preemption**: Default
- **Confidential Access Mode**: < None >
- **Confidential Access Level**: < None >

### Do Not Disturb

- **Do Not Disturb**
- **DND Option**: Use Common Phone Profile Setting
- **DND Incoming Call Alert**: < None >

---

Cisco IP Phone 9971 SIP Configuration (Continued...)
### Secure Shell Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell User</td>
<td>administrator</td>
</tr>
<tr>
<td>Secure Shell Password</td>
<td>***********</td>
</tr>
</tbody>
</table>

### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>PC Port *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Back USB Port *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Side USB Port *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Camera *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Console Access *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Video Capabilities *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Enable/Disable USB Classes</td>
<td>Mass Storage, Human Interface Device, Audio Class</td>
<td></td>
</tr>
<tr>
<td>SDIO *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Bluetooth *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Wifi *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Bluetooth Profiles *</td>
<td>Handfree, Human Interface Device</td>
<td></td>
</tr>
<tr>
<td>Settings Access *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Web Access *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Show All Calls on Primary Line *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Sunday, Monday, Tuesday</td>
<td></td>
</tr>
</tbody>
</table>

---

Cisco IP Phone 9971 SIP Configuration (Continued...)

Note: Testing was conducted in tekVizion labs
Cisco IP Phone 9971 SIP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display On Time</td>
<td>07:30</td>
</tr>
<tr>
<td>Display On Duration</td>
<td>10:30</td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td>01:00</td>
</tr>
<tr>
<td>HTTPS Server x</td>
<td>http and https Enabled</td>
</tr>
<tr>
<td>Enable Power Save Plus</td>
<td>Sunday, Monday, Tuesday</td>
</tr>
<tr>
<td>Phone On Time</td>
<td>00:00</td>
</tr>
<tr>
<td>Phone Off Time</td>
<td>24:00</td>
</tr>
<tr>
<td>Phone Off Idle Timeout x</td>
<td>60</td>
</tr>
<tr>
<td>Enable Audible Alert</td>
<td></td>
</tr>
<tr>
<td>EnergyWise Domain</td>
<td></td>
</tr>
<tr>
<td>EnergyWise Endpoint Security Secret</td>
<td></td>
</tr>
<tr>
<td>Allow EnergyWise Overrides</td>
<td></td>
</tr>
<tr>
<td>Open to PC Port x</td>
<td>Disabled</td>
</tr>
<tr>
<td>Logging Display x</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Tone x</td>
<td>Disabled</td>
</tr>
<tr>
<td>Recording Tone Local Volume x</td>
<td>100</td>
</tr>
<tr>
<td>Recording Tone Remote Volume x</td>
<td>50</td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call x</td>
<td>Enabled</td>
</tr>
<tr>
<td>RTCP x</td>
<td>Enabled</td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
</tr>
<tr>
<td>Remote Log x</td>
<td>Disabled</td>
</tr>
<tr>
<td>Log Profile</td>
<td>Default</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Advertise G.722 and ISAC Codecs x</td>
<td>Use System Default ▼</td>
</tr>
<tr>
<td>Wideband Headset UI Control x</td>
<td>Enabled ▼</td>
</tr>
<tr>
<td>Wideband Headset x</td>
<td>Enabled ▼</td>
</tr>
<tr>
<td>Peer Firmware Sharing x</td>
<td>Enabled ▼</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port x</td>
<td>Enabled ▼</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port x</td>
<td>Enabled ▼</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port x</td>
<td>Enabled ▼</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port x</td>
<td>Enabled ▼</td>
</tr>
</tbody>
</table>

**Cisco IP Phone 9971 SIP Configuration (Continued...)**

Note: Testing was conducted in tekVizion labs
<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hide Video By Default*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Background Image</td>
<td></td>
</tr>
<tr>
<td>Simplified New Call UI*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Enable VXC VPN for MAC</td>
<td></td>
</tr>
<tr>
<td>VXC VPN Option*</td>
<td>Dual Tunnel</td>
</tr>
<tr>
<td>VXC Challenge*</td>
<td>Challenge</td>
</tr>
<tr>
<td>VXC-M Servers</td>
<td></td>
</tr>
<tr>
<td>Revert to All Calls*</td>
<td>Disabled</td>
</tr>
<tr>
<td>RTCP for Video*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Record Call Log from Shared Line*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Show Remote Private Calls*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Record Call Log For Remote Private Calls*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Show Call History for Selected Line Only*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Actionable Incoming Call Alert*</td>
<td>Disabled</td>
</tr>
<tr>
<td>DF bit*</td>
<td>0</td>
</tr>
<tr>
<td>Default Line Filter</td>
<td></td>
</tr>
<tr>
<td>Separate Audio and Video Mute*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Softkey Control*</td>
<td>Feature Control Policy</td>
</tr>
<tr>
<td>Start Video Port</td>
<td></td>
</tr>
<tr>
<td>Stop Video Port</td>
<td></td>
</tr>
<tr>
<td>Lowest Alerting Line State Priority*</td>
<td>Disabled</td>
</tr>
<tr>
<td>TLS Resumption Timer*</td>
<td>3600</td>
</tr>
<tr>
<td>Audio EQ*</td>
<td>Default : Default</td>
</tr>
</tbody>
</table>

Cisco 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...)

<table>
<thead>
<tr>
<th>Directory URIs</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
</tr>
<tr>
<td>---------</td>
</tr>
<tr>
<td></td>
</tr>
</tbody>
</table>

- PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing

- AAR Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Retain this destination in the call forwarding history</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Calling Search Space Activation Policy</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward All</td>
<td>Use System Default</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
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.

### Line 1 on Device SEPC07B8CA18872

<table>
<thead>
<tr>
<th><strong>Display (Caller ID)</strong></th>
<th>Cisco 9971 - Phone 2</th>
<th><strong>Display text for a line appearance</strong> is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ASCII Display (Caller ID)</strong></td>
<td>Cisco 9971 - Phone 2</td>
<td></td>
</tr>
<tr>
<td><strong>Line Text Label</strong></td>
<td>Cisco 9971 - Phone 2</td>
<td></td>
</tr>
<tr>
<td><strong>External Phone Number Mask</strong></td>
<td>7323204351</td>
<td></td>
</tr>
<tr>
<td><strong>Visual Message Waiting Indicator Policy</strong></td>
<td>Use System Policy</td>
<td></td>
</tr>
<tr>
<td><strong>Audible Message Waiting Indicator Policy</strong></td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Idle)</strong></td>
<td>Use System Default</td>
<td>Applies to this line when any line on the phone has a call in progress.</td>
</tr>
<tr>
<td><strong>Ring Setting (Phone Active)</strong></td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td><strong>Call Pickup Group Audio Alert</strong></td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td><strong>Recording Option</strong></td>
<td>Call Recording Disabled</td>
<td></td>
</tr>
<tr>
<td><strong>Recording Profile</strong></td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td><strong>Recording Media Source</strong></td>
<td>Gateway Preferred</td>
<td></td>
</tr>
<tr>
<td><strong>Monitoring Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

**Cisco IP Phone 9971 SIP Configuration (Continued...)**
### Multiple Call/Call Waiting Settings on Device SEPC07BBA1B872

**Note:** The range to select the Max Number of calls is: 1-200

- **Maximum Number of Calls**: 4
- **Busy Trigger**: 2

(Less than or equal to Max. Calls)

### Forwarded Call Information Display on Device SEPC07BBA1B872

- **Caller Name**
- **Caller Number**
- **Redirected Number**
- **Dialed Number**

### Users Associated with Line

<table>
<thead>
<tr>
<th>Associate End Users</th>
</tr>
</thead>
<tbody>
<tr>
<td>Save</td>
</tr>
</tbody>
</table>

---

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...)

Note: Testing was conducted in tekVizion labs
### SDP Information

- **SDP Session-level Bandwidth Modifier for Early Offer and Re-invites**
  - TIAS and AS
- **SDP Transparency Profile**
  - Pass all unknown SDP attributes
- **Accept Audio Codec Preferences in Received Offer**
  - Default
- **Require SDP Inactive Exchange for Mid-Call Media Change**
- **Allow RR/RS bandwidth modifier (RFC 3556)**

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32765</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-gpickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-cfwdial</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrdial</td>
</tr>
</tbody>
</table>
SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued...)

- Conference Join Enabled
- RFC 2543 Hold
- Semi Attended Transfer
- Enable YAD
- Stutter Message Waiting
- MLPP User Authorization

### Normalization Script

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td></td>
</tr>
</tbody>
</table>

### Incoming Requests FROM URI Settings

- Caller ID DN
- Caller Name

Note: Testing was conducted in tekVizion labs
SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued...)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reroute Incoming Request to new Trunk based on</td>
<td>Never</td>
</tr>
<tr>
<td>RSVP Over SIP</td>
<td>Local RSVP</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Fall back to local RSVP</td>
<td></td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK if 1xx Contains SDP</td>
</tr>
<tr>
<td>Video Call Traffic Class</td>
<td>Mixed</td>
</tr>
<tr>
<td>Calling Line Identification Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Session Refresh Method</td>
<td>Invite</td>
</tr>
<tr>
<td>Early Offer support for voice and video calls</td>
<td>Disabled (Default value)</td>
</tr>
<tr>
<td>Enable ANAT</td>
<td></td>
</tr>
<tr>
<td>Deliver Conference Bridge Identifier</td>
<td></td>
</tr>
<tr>
<td>Allow Passthrough of Configured Line Device Caller Information</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Incoming Calls</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Outgoing Calls</td>
<td></td>
</tr>
<tr>
<td>Send ILS Learned Destination Route String</td>
<td></td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
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/Eary 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.

```
Product: SIP Trunk
Device Protocol: SIP
Trunk Service Type: None (Default)
Device Name*: Trunk_SIP_FAX_Gateway
Description: Trunk to SIP FAX Gateway
Device Pool*: G729 Pool
Common Device Configuration: < None >
Call Classification*: Use System Default
Media Resource Group List: MRGL_MTP
Location*: Hub_None
AAR Group: < None >
Tunneled Protocol*: None
QSIG Variant*: No Changes
ASN.1 ROSE OID Encoding*: No Changes
Packet Capture Mode*: None
Packet Capture Duration: 0
```
SIP Trunk to Fax Gateway Configuration (Continued...)
Note: Testing was conducted in tekVizion labs
- Incoming Calling Party Settings

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

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

- Incoming Called Party Settings

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

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

- Connected Party Settings

Connected Party Transformation CSS: < None >

Use Device Pool Connected Party Transformation CSS

- Outbound Calls

Called Party Transformation CSS: < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS: < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection: Originator

Calling Line ID Presentation: Default

Calling Name Presentation: Default

Calling and Connected Party Info Format: Deliver DN only in connected party

Redirecting Diversion Header Delivery: Outbound

Redirecting Party Transformation CSS: < None >

Use Device Pool Redirecting Party Transformation CSS

- Caller Information

Caller ID DN
Caller Name

Maintain Original Caller ID DN and Caller Name in Identity Headers

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EDCS# xxx Rev #
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Note: Testing was conducted in tekVizion labs
SIP Trunk to Fax Gateway Configuration (Continued...)

**SIP Information**

- **Destination Address**: 172.16.31.50, Destination Address IPv6, Destination Port 5060
- **MTI Preferred Originating Codec**: 711ulaw
- **BLF Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: ATT Non Secure SIP Trunk Profile
- **Rerouting Calling Search Space**: <None>
- **Out-Of-Dial Refer Calling Search Space**: <None>
- **SUBSCRIBE Calling Search Space**: <None>
- **SIP Profile**: Standard SIP Profile for ATT
- **DTMF Signaling Method**: No Preference

**Normalization Script**

- **Normalization Script**: <None>
- **Enable Trace**

**Recording Information**

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

**Geolocation Configuration**

- **Geolocation**: <None>
- **Geolocation Filter**: <None>
- **Send Geolocation Information**

**Buttons**

- Save, Delete, Reset, Add New
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

**Route Pattern**: 9.1000008

**Route Partition**: <None>

**Description**: To PSTN via ATT SIP Trunk

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Numbering Plan</td>
<td>NAP</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>NLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td>No</td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Route Class</td>
<td>Default</td>
</tr>
</tbody>
</table>

**Gateway/Route List**: ATT_SIP_TRUNK

**Call Classification**: OffNet

**External Call Control Profile**: <None>

**Route Option**: Route this pattern

### Calling Party Transformations

**Use Calling Party's External Phone Number Mask**

**Calling Party Transform Mask**

**Prefix Digits (Outgoing Calls)**

**Calling Line ID Presentation**: Default

**Calling Name Presentation**: Default

**Calling Party Number Type**: Cisco CallManager

**Calling Party Numbering Plan**: Cisco CallManager
Route Pattern Configuration (Continued...)

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>PreDot</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Ccall)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Protocol</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Carrier Identification Code</td>
<td></td>
</tr>
<tr>
<td>Network Service</td>
<td>Service Parameter Name</td>
</tr>
<tr>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
</tr>
</tbody>
</table>

[Form Fields and Buttons]
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.

---

**Calling Party Transformations**

- Use Calling Party’s External Phone Number Mask
- Calling Party Transform Mask
- Prefix Digits (Outgoing Calls)
- Calling Line ID Presentation* = Default
- Calling Name Presentation* = Default
- Calling Party Number Type* = Cisco CellManager
- Calling Party Numbering Plan* = Cisco CellManager

---
Route Pattern Configuration (Continued...)

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
</tr>
<tr>
<td>Called Party Number Type</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Protocol</td>
</tr>
<tr>
<td>Carrier Identification Code</td>
</tr>
<tr>
<td>Network Service</td>
</tr>
<tr>
<td>-- Not Selected --</td>
</tr>
</tbody>
</table>

Save  Delete  Copy  Add New
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...)
### Connected Party Transformations

<table>
<thead>
<tr>
<th>Item</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

### Called Party Transformations

<table>
<thead>
<tr>
<th>Item</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

### ISDN Network-Specific Facilities Information Element

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>--- Not Selected ---</th>
</tr>
</thead>
<tbody>
<tr>
<td>Carrier Identification Code</td>
<td></td>
</tr>
<tr>
<td>Network Service</td>
<td>--- Not Selected ---</td>
</tr>
<tr>
<td>Service Parameter Name</td>
<td>&lt; Not Exist &gt;</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted in tekVizion labs
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

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Built In Bridge*</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Mode*</td>
<td>Default</td>
</tr>
<tr>
<td>Owner User ID*</td>
<td>jabber1</td>
</tr>
<tr>
<td>Mobility User ID</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Primary Phone</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Always Use Prime Line*</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message*</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- Ignore Presentation Indicators (internal calls only)
- Allow Control of Device from CTI
- Logged Into Hunt Group
- Remote Device
- Require off-premise location

Jabber Client Configuration (Contd...)
### Number Presentation Transformation

**Caller ID For Calls From This Phone**
- **Calling Party Transformation CSS**
  - < None >
- **Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)**

**Remote Number**
- **Calling Party Transformation CSS**
  - < None >
- **Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)**

### Protocol Specific Information

- **Packet Capture Mode***
  - None
- **Packet Capture Duration**
  - 0
- **BLF Presence Group***
  - Standard Presence group
- **SIP Dial Rules**
  - < None >
- **MTP Preferred Originating Codec***
  - 711ulaw
- **Device Security Profile***
  - Cisco Unified Client Services Framework - Standard
- **Rerouting Calling Search Space**
  - < None >
- **SUBSCRIBE Calling Search Space**
  - < None >
- **SIP Profile***
  - Standard SIP Profile w/Early Media Disabled.
  - View Details
- **Digest User**
  - < None >
- **Media Termination Point Required**
- **Unattended Port**
- **Require DTMF Reception**
Jabber Client Configuration (Contd...)

### Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Certificate Operation*</th>
<th>No Pending Operation</th>
</tr>
</thead>
<tbody>
<tr>
<td>Authentication Mode*</td>
<td>By Null String</td>
</tr>
<tr>
<td>Authentication String</td>
<td></td>
</tr>
<tr>
<td>Generate String</td>
<td></td>
</tr>
<tr>
<td>Key Size (Bits)*</td>
<td>2048</td>
</tr>
<tr>
<td>Operation Completes By</td>
<td>2015 06 20 12 (YYYY:MM:DD:HH)</td>
</tr>
<tr>
<td>Certificate Operation Status</td>
<td>None</td>
</tr>
</tbody>
</table>

Note: Security Profile Contains Addition CAPF Settings.

### Extension Information

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

### MLPP and Confidential Access Level Information

<table>
<thead>
<tr>
<th>MLPP Domain</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Confidential Access Mode</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Do Not Disturb

- Do Not Disturb
- DND Option*: Ringer Off
- DND Incoming Call Alert: < None >

Jabber Client Configuration (Contd...)

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EDCS# xxx Rev #
Page 121 of 146
Note: Testing was conducted in tekVizion labs
### Jabber Client Configuration (Contd...)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Video Calling</strong></td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td><strong>Interactive Connectivity Establishment (ICE)</strong></td>
<td></td>
<td></td>
</tr>
<tr>
<td>ICE</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Default Candidate Type</td>
<td>Host</td>
<td></td>
</tr>
<tr>
<td>Server Reflexive Address</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Primary TURN Server Host Name or IP Address</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Secondary TURN Server Host Name or IP Address</td>
<td></td>
<td></td>
</tr>
<tr>
<td>TURN Server Transport Type</td>
<td>Auto</td>
<td></td>
</tr>
<tr>
<td>TURN Server Username</td>
<td>administrator</td>
<td></td>
</tr>
<tr>
<td>TURN Server Password</td>
<td>************</td>
<td></td>
</tr>
</tbody>
</table>

**Instant Messaging**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>File Types to Block in File Transfer</td>
<td></td>
</tr>
<tr>
<td>URLs to Block in File Transfer</td>
<td></td>
</tr>
</tbody>
</table>
Voicemail Port Configuration

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

![Voicemail Port Configuration](image)

**Note:** Testing was conducted in tekVizion labs.
Voicemail Port Configuration (Continued...)

Set Port Name = CiscoUM1-V11. 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.

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Port Name</td>
<td>CiscoUM1-V11</td>
</tr>
<tr>
<td>Description</td>
<td>VM Port</td>
</tr>
<tr>
<td>Device Pool</td>
<td>G729</td>
</tr>
<tr>
<td>Directory Number*</td>
<td>2501</td>
</tr>
<tr>
<td>Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Internal Cell ID Display</td>
<td>VoiceMail</td>
</tr>
<tr>
<td>Internal Cell ID Display (ASCII format)</td>
<td>VoiceMail</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs.
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-nre
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-nre
no fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none no vad
no vad
Cisco UCM SIP Integration with Cisco Unity Connection (CUC)

CUC Version

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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

![Cisco Unity Connection Administration Interface](image)

- **Status:** Found 2 Port Group(s)
- **Port Groups (1 - 2 of 2):**
  - **SIP-1**
    - **Port Group Name:** SIP-1
    - **Phone System Display Name:** SIP
    - **Port Count:** 2
    - **Integration Method:** SIP
    - **Needs Reset:** No
CUC Port Group (continued...)

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

Check Register with SIP Server.
CUC Port Settings

![Image of Cisco Unity Connection Administration interface showing 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.

![Cisco Unity Connection Administration](image)
Auto Attendant (Continued…)

![Auto Attendant Configuration](image-url)
Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP)

CUP/IMP Version

![Cisco Unified CM IM and Presence Administration](image)

System version: 11.0.1.10000-6
VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 2048Mbytes RAM, Partitions aligned

User administrator last logged in to this cluster on Wednesday, February 10, 2016 6:31:52 AM CST, to node 10.80.14.3, from 172.16.29.40 using HTTPS

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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

![Node User Assignment (clus24imp-pub.lab.tekvizion.com)](image-url)
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

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