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 4431 Router with SIP Interface JAN 2016
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Introduction

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

• This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 11.0, Cisco Unity Connection 11.0, Cisco Unified CM IM and Presence 11.0, Cisco Integrated Services Routers (ISR) Version 15.5(3)S1a with connectivity to AT&T’s IP Flex-Reach SIP trunk service. It also covers support and configuration example of Cisco Unity Connection (CUC) messaging integrated with Cisco Unified Communications Manager (Cisco UCM). The deployment model covered in this application note is Cisco Integrated Services Routers (ISR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.

• Testing was performed in accordance to AT&T’s IP Flexible Reach test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, CODEC negotiation, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), CISCO auto-attendant (BACD), fax G.711 and T38 (G3 and SG3 speeds), teleconferencing, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.

• The Cisco Unified Border Element function configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between AT&T SIP network and Cisco Integrated services router. The configurations described in this document details the important commands for successful interoperability. Care must be taken by the network administrator deploying Cisco ISR to ensure these commands are set per each dial-peer required, to interoperate done AT&T SIP network.

• Consult your Cisco representative for the correct IOS image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified Communication Manager with Cisco Unified Border Element components.
Network Topology
Hardware Components

- UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7975 & Cisco 9971 phones
- Cisco ISR4431/K9 (1RU) processor with 1659383K/6147K bytes of memory.
- Processor board ID FTX1850ALVU
  - 4 Gigabit Ethernet interfaces
  - 32768K bytes of non-volatile configuration memory.

Software Requirements

- ISR: ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.5(3)S1a, RELEASE SOFTWARE (fc1)
- System image file is “isr4400-universalk9.03.16.01a.S.155-3.S1a-ext.SPA.bin”.
- Cisco Unity Connection version: System version: 11.0.1.10000-10
- Cisco Unified CM IM and Presence: System version: 11.0.1.10000-6
- Cisco Jabber client version: 11.0.0 Build 65527
- VentaFax client version: 7.6.244.598 I
Features

Features – Supported

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

Network Based Features - Supported

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

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

Features - Not Supported

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

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

Hold/Resume & Music on Hold (MOH)
• Re-invites for hold/resume from PSTN network is potentially dependent on the carrier/network through which the call is traversing.

Ringback Tone on Early Unattended Transfer
• Caller does not hear ringback 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

CISCO_4K_ROUTER2#show version
Cisco IOS XE Software, Version 03.16.01a.S - Extended Support Release
Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.5(3)S1a, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
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Compiled Wed 04-Nov-15 12:50 by mcpre

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ROM: IOS-XE ROMMON

CISCO_4K_ROUTER2 uptime is 2 weeks, 5 days, 8 hours, 17 minutes
Uptime for this control processor is 2 weeks, 5 days, 8 hours, 18 minutes
System returned to ROM by reload
System image file is "bootflash:/isr4400-universalk9.03.16.01a.S.155-3.51a-ext.SPA.bi"

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.

Suite License Information for Module: 'esg'

<table>
<thead>
<tr>
<th>Suite</th>
<th>Suite Current</th>
<th>Type</th>
<th>Suite Next reboot</th>
</tr>
</thead>
</table>

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FoundationSuiteK9  None  None  None
securityk9
appxk9

AdvUCSuiteK9  None  None  None
uck9
cme-srst
cube

Technology Package License Information:

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<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Technology-package</th>
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<th>Next reboot</th>
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</table>
appxk9           | None               | None               | None    | None       | None       |
uck9             | uck9               | Evaluation         | uck9    |            |            |
securityk9       | None               | None               | None    | None       | None       |
ipbase           | ipbasek9           | Permanent          | ipbasek9|            |            |

cisco ISR4431/K9 (1RU) processor with 1659383K/6147K bytes of memory.
Processor board ID FTX1850ALVU
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7057407K bytes of flash memory at bootflash:

Configuration register is 0x2102

CISCO_4K_ROUTER2#show running-config
Building configuration...

Current configuration : 11328 bytes
!
! Last configuration change at 13:15:54 UTC Tue Dec 29 2015 by cisco
!
version 15.5
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CISCO_4K_ROUTER2
!
boot-start-marker
boot system flash isr4400-universalk9.03.16.01a.S.155-3.51a-ext.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
  
 address-family ipv4
 exit-address-family
  
 address-family ipv6
 exit-address-family
  
 enable secret 5 $1$zQRB$CCbfD1aYzk3kPvzAm2KU0
 enable password cisco
  
 aaa new-model
  
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subscriber templating
multilink bundle-name authenticated

cts logging verbose

voice service voip
rtp-port range 16384 32766
address-hiding ¹

¹ Hide signaling and media peer addresses from endpoints other than gateway.
mode border-element
media disable-detailed-stats
allow-connections sip to sip
no supplementary-service sip handle-replaces
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
  header-passing
error-passthru
  asserted-id pai
no update-callerid
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
voice class codec

---

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.
4 This command allows SIP error messages to pass-through end-to-end without modification through Cisco UBE.
5 This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai).
6 This command enables delay offer-to-early offer conversion of initial SIP INVITE message to calls matched to this dial-peer level.
7 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.
8 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.
codec preference 1 g729r8 bytes 30
codec preference 2 g711ulaw
!
voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8 bytes 30
!
voice class codec 3
codec preference 1 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>" 10
request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30" 11
response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
request INVITE sdp-header Audio-Attribute add "a=ptime:30" 12
!
!

9 This command configures the codec preference to be assigned to dial-peers. Alternatively, single code can be configured into individual dial-peers.
10 This SIP profile expands the Diversion header number from a 4-digit extension to a full 10-digit DID number in order to obtain interoperability with AT&T’s served users during call-forward scenarios. The six digits in "sip: 732216" are variable and must be replaced with the first 6 digits of the DID’s provisioned for the customer site.
11 Cisco 6900-series IP phones use ptime value of 20 ms. AT&T networks prefer ptime value of 30 ms. This SIP profile modifies SDP ptime value from 20 to 30 ms and it should be applied to dial-peers where G729 is the preferred codec. If the customer creates a dial-peer specifically for G711, a sip-profile without modifying the ptime value should be applied. This is because G711 RTP was not defaulting to 20ms.
12 This SIP profile is required in order to advertise the ptime=30 attribute in the outgoing SIP INVITE from Cisco UBE to AT&T. Currently RFC’s do not have a standard method to advertise ptime values for each offered codec within a SDP offering with multiple codecs. This SIP profile allows for Cisco UBE to include the ptime attribute with a value of 30ms.
voice translation-rule 1

rule 1 /^.*\(40..\)/ /732320\1/

voice translation-profile NPA

translate calling 1

license udi pid ISR4431/K9 sn FOC18232988

license boot level appxk9

license boot level uck9

username cisco privilege 15 secret 5 $1$AGR7$e7pQx6UI0be3bzRbc0lr81

redundancy

mode none

\[13\] This command used to convert 4 digit to 10 digit in contact header otherwise ATT will send 6xx error response while executing network related feature.
vlan internal allocation policy ascending

interface GigabitEthernet0/0/0
ip address 10.64.4.20 255.255.0.0
media-type rj45
negotiation auto

interface GigabitEthernet0/0/1
no ip address
shutdown
media-type rj45
negotiation auto

interface GigabitEthernet0/0/2 \(^{14}\)

ip address 10.80.22.75 255.255.255.0 \(^{15}\)
media-type rj45
negotiation auto

\(^{14}\) LAN interface to Cisco UCM
\(^{15}\) Cisco UBE LAN interface IPv4 Address
interface GigabitEthernet0/0/3 16
   description Wan Interface
   ip address 192.65.79.58 255.255.255.224
   media-type rj45
   negotiation auto
!
interface GigabitEthernet0
   vrf forwarding Mgmt-intf
   no ip address
   shutdown
   negotiation auto
!
interface Vlan1
   no ip address
   shutdown
!
ip forward-protocol nd
   no ip http server
   no ip http secure-server
   ip route 0.0.0.0 0.0.0.0 192.65.79.33
   ip route 10.80.22.0 255.255.255.0 10.80.22.1
   ip route 172.16.0.0 255.255.0.0 10.80.22.1
   ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 10.64.1.1
!

16 WAN interface to AT&T

Note: Testing was conducted in tekVizion labs
control-plane

dial-peer voice 200 voip

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

no modem passthrough

session protocol sipv2
incoming called-number [1-9]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 GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 800 voip
description " Incoming AT&T to IP-PBX . AT&T facing side "
huntstop
no modem passthrough
session protocol sipv2
incoming called-number [37][13][24]32040..
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nce
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 700 voip
description " Incoming AT&T to IP-PBX - IP-PBX facing side "
huntstop
destination-pattern [37][13][24].......  
no modem passthrough
session protocol sipv2
session target ipv4:10.80.22.2:5060
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 GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nce
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 100 voip 17
description "Outgoing To AT&T"-AT&T facing side
destination-pattern 73236.....
no modem passthrough
session protocol sipv2 18
session target ipv4:207.242.225.210
voice-class codec 1 19
voice-class sip asymmetric payload full 20
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru 21
voice-class sip profiles 1 22
voice-class sip bind control source-interface GigabitEthernet0/0/3 23
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n-te 24

17 Dial peer for AT&T facing network
18 Session protocol SIPv2 is used for this testing
19 Assigns voice class codec 1 settings to dial-peer (codec support and filtering).
20 Configures the dynamic SIP asymmetric payload support.
21 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.
22 This command enables the dial peer to use SIP profile 1
23 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.
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

no modem passthrough

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 GigabitEthernet0/0/3

voice-class sip bind media source-interface GigabitEthernet0/0/3

dtmf-relay rtp-npe

fax rate 14400

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

no vad

!

dial-peer voice 400 voip

---

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

25 This command enables T38 fax protocol for calls terminating on this dial-peer
description " Int'l calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number 011T
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 GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 500 voip
description " N11 Calls to AT&T - AT&T facing side "
destination-pattern .11
no modem passthrough
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 600 voip
description " N11 Calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number .11
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 GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nte
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 122 voip
description "OPERATOR TESTING"
destination-pattern 0
no modem passthrough
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nite
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 141 voip
description "Network Feature"
translation-profile outgoing NPA
destination-pattern *..
no modem passthrough
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 2151 voip
description "Outgoing To AT&T"-AT&T facing side
destination-pattern 7323204607
no modem passthrough
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n-te
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 214 voip
description "Outgoing To AT&T"-AT&T facing side
destination-pattern [2-9]T
no modem passthrough
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n-te
fax-relay ecm disable
fax-relay sg3-to-g3
fax rate disable
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
!
gateway
timer receive-rtp 1200
!
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0
session-timeout 90
exec-timeout 960 0
password tekV1z10n
no activation-character
logging synchronous
transport preferred ssh
transport input all
stopbits 1
line vty 1 4
exec-timeout 960 0
password tekV1z10n
logging synchronous
transport input all
!
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 Unified CM Administration](image)

**Cisco UCM Region Configuration**

**Navigation Path:** System → Region Information → Region
Device Pool Configuration

**Navigation Path:** System → Device Pool
“G729_pool” 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...)
Note: Testing was conducted in tekVizion labs
Device Pool Configuration (continued...)

- **Incoming Called Party Settings**
  - If the administrator sets the prefix to Default; this indicates call processing will use prefix at the next level setting (Device Pool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

- **Phone Settings**
  - **Caller ID For Calls From This Phone**
    - Calling Party Transformation CSS: <None>
  - **Connected Party Settings**
    - Connected Party Transformation CSS: <None>
  - **Redirecting Party Settings**
    - Redirecting Party Transformation CSS: <None>
Annunciator Configuration

**Navigation:** Media Resource → Annunciator

Set Name* = ANN_2.
Set Description = ANN_clus32pubsub. This is used for this example
Set Device Pool* = G729_pool.
Conference Bridge Configuration

**Navigation**: Media Resources → Conference Bridge

Set Conference Bridge Type* = Cisco Conference Bridge Software.
Set Host Server = clus32pubsub. This is used for this example.
Set Conference Bridge Name* = CFB_2.
Set Description = CFB_clus32pubsub. This is used in this example.
Set Device Pool* = G729_pool.
Media Termination Point Configuration

**Navigation:** Media Resource → Media Termination Point

Set Media Termination Point Name* = MTP_2
Set Description = MTP_clus32pubsub. This is used for this example
Set Device pool* = G729 Pool

<table>
<thead>
<tr>
<th>Media Termination Point Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Registration:</strong> Registered with Cisco Unified Communications Manager clus32pubsub</td>
</tr>
<tr>
<td><strong>IPv4 Address:</strong> 10.80.22.2</td>
</tr>
<tr>
<td><strong>Media Termination Point Type</strong>*</td>
</tr>
<tr>
<td><strong>Host Server</strong>*</td>
</tr>
<tr>
<td><strong>Media Termination Point Name</strong>*</td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td><strong>Device Pool</strong>*</td>
</tr>
<tr>
<td><strong>Trusted Relay Point</strong></td>
</tr>
</tbody>
</table>
Music on Hold Server Configuration

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

Set Music on Hold Server Name* = MOH_2.
Set Description = MOH_clus32pubsub. This is used for this example.
Set Device Pool* = G729_pool.
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* = clus32pubsub--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* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.
Select Service* = Cisco CallManager (Active).
Select Duplex Streaming Enabled * = True
Media Resource Group Configuration

**Navigation Path:** Media Resources → Media Resources group

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

Set Name* = MRG_MTP - This is used for this example.  
Set Description = MRG_MTP - This text is used to define this Media Resource Group List.  
Set all Resources in the selected Media Resources Box.

**Note:** Testing was conducted in tekVizion labs
Set Name = MRGL_MTP.
Set selected Media Resource Groups = MRG_MTP.

UC Service Configuration

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

Note: Testing was conducted in tekVizion labs
**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* = 10.80.22.2 (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* = 10.80.22.3 (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...)
Note: Testing was conducted in tekVizion labs
End User Configuration

**Navigation:** User Management ➔ End User

Set User ID* = jabber – This is used in this example.
Set Password = Password for profile.
Set Directory URI = jabber@lab.tekvizion.com.

![End User Configuration Diagram]

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EDCS# xxx Rev #
Page 54 of 154
Note: Testing was conducted in tekVizion labs
End User Configuration (continued...)
Note: Testing was conducted in tekVizion labs
End User Configuration (continued...)

![Device Information](image)

![Extension Mobility](image)

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

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

**CAPF Information**
- Associated CAPF Profiles

**Permissions Information**
- Groups:
  - Admin-3rd Party API
  - Application Client Users
  - Standard Audit Users
  - Standard CARE Admin Users
  - Standard CCM Admin Users

- Roles:
  - Standard AXL API Access
  - Standard Admin (Rep Tool) Admin
  - Standard Audit Log Administration
  - Standard CCM Admin Users
  - Standard CCM End Users

**Conference Now Information**
- Enable End User to Host Conference Now
- Meeting Number
- Attendees Access Code

Save  Delete  Add New
Cisco IP Phone 7975 SCCP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = Cisco7975_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 Softkey Template = Standard User. This is used in this example.
Cisco IP Phone 7975 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 7975 SCCP Configuration (Continued...)
### 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**
  - 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**
- **Unattended Port**
- **Require DTMF Reception**
- **RFC2833 Disabled**

### Certification Authority Proxy Function (CAPF) Information

- **Certificate Operation**
  - No Pending Operation
- **Authentication Mode**
  - By Null String
- **Authentication String**
- **Generate String**
- **Key Order**
  - RSA Only
- **RSA Key Size (Bits)**
  - 2048
- **EC Key Size (Bits)**
  - None
- **Operation Completes By**
  - 2016-01-18 12:00 (YYYY-MM-DD:HH)
- **Certificate Operation Status**
  - None
- **Note:** Security Profile Contains Additional CAPF Settings.

### Expansion Module Information

- **Module 1**
  - None
- **Module 1 Load Name**
- **Module 2**
  - None
- **Module 2 Load Name**
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 Procurement: 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: ******************
Cisco IP Phone 7975 SCCP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Line Select</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Web Access</td>
<td>Disabled</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>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>Enable Power Save Plus</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>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>
</tbody>
</table>
Cisco IP Phone 7975 SCCP Configuration (Continued...)
Cisco IP Phone 7975 SCCP Configuration (Continued…)

Set Directory Number* = 4086. This is used in this example.
Set Description = 7323204086. This is used in this example.
Set Alerting Name = 7323204086. This is used in this example.
Set ASCII Alerting Name = 7323204086. This is used in this example.
Cisco IP Phone 7965 SCCP Configuration (Continued…)

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

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

<table>
<thead>
<tr>
<th>AAR Settings</th>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Call Forward and Call Pickup Settings

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

No Answer Ring Duration (seconds) < None >

Call Pickup Group < None >
Cisco IP Phone 7965 SCCP Configuration (Continued...)

- **Park Monitoring**
  - **Voice Mail**
    - Park Monitoring: [ ] or [ ]
    - Forward: [ ] or [ ]
    - Retrieve Destination: [ ] or [ ]
    - External: [ ] or [ ]

- **MLPP Alternate Party And Confidential Access Level Settings**
  - **Target (Destination)**
  - **MLPP Calling Search Space**
  - **MLPP No Answer Ring Duration (seconds)**
  - **Confidential Access Mode**
  - **Confidential Access Level**
  - **Call Control Agent Profile**

- **Line Settings for All Devices**
  - Hold Reversion Ring Duration (seconds)
  - Hold Reversion Notification Interval (seconds)
  - Party Entrance Tone

- **Line 1 on Device SEP0008E3031F5D4**
  - **Display (Caller ID)**
  - **ASCII Display (Caller ID)**
  - **Line Text Label**
  - **External Phone Number Mask**
  - **Visual Message Waiting Indicator Policy**
  - **Audible Message Waiting Indicator Policy**
  - **Ring Setting (Phone Idle)**
  - **Ring Setting (Phone Active)**
  - **Call Pickup Group**
  - **Audio Alert Setting (Phone Idle)**

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

<table>
<thead>
<tr>
<th>Call Pickup Group</th>
<th>Use System Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Audio Alert Setting (Phone Active)</td>
<td></td>
</tr>
<tr>
<td>Recording Option*</td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Recording Media Source*</td>
<td>Gateway Preferred</td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Log Missed Calls</td>
<td>✓</td>
</tr>
</tbody>
</table>

**Multiple Call/Call Waiting Settings on Device SEP00083031F5D4**

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

<table>
<thead>
<tr>
<th>Maximum Number of Calls*</th>
<th>1</th>
</tr>
</thead>
<tbody>
<tr>
<td>Busy Tripper*</td>
<td>1</td>
</tr>
<tr>
<td>(Less than or equal to Max. Cells)</td>
<td></td>
</tr>
</tbody>
</table>

**Forwarded Call Information Display on Device SEP00083031F5D4**

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

**Users Associated with Line**

- Add New

Save Delete Reset Apply Config Add New
Cisco IP Phone 9971 SIP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = 7323204085. this text is used to identify this Phone.
Set Device Pool* = G729 Pool. 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...)

Note: Testing was conducted in tekVizion labs
### Device Mobility Mode

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Owner</td>
<td>User</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>Jabber</td>
</tr>
<tr>
<td>BLF Audible Alert</td>
<td>Default</td>
</tr>
<tr>
<td>Setting (Phone Busy)</td>
<td>Default</td>
</tr>
<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>
<tr>
<td>Feature Control Policy</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### 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)**: Checked

#### Remote Number

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

---

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

### Protocol Specific Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</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</td>
<td>71ilaw</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Cisco 9971 - Standard SIP Non-Secure Profile</td>
</tr>
</tbody>
</table>

### Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</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>RSA Key Size (Bits)</td>
<td>2048</td>
</tr>
<tr>
<td>Operation Completes By</td>
<td>2016 1 21 12 (YYYY:MM:DD:HH)</td>
</tr>
<tr>
<td>Certificate Operation</td>
<td>None</td>
</tr>
</tbody>
</table>

### 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>
</tbody>
</table>

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

External Data Locations Information (Leave blank to use default)

- Information
- Directory
- Messages
- Services
- Authentication Server
- Proxy Server
- Idle
- Idle Timer (seconds)
- Secure Authentication URL
- Secure Directory URL
- Secure Idle URL
- Secure Information URL
- Secure Messages URL
- Secure Services URL

Extension Information

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

MLPP and Confidential Access Level Information

- MLPP Domain: None
- MLPP Indication: Default
- MLPP Association: 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: **************
Cisco IP Phone 9971 SIP Configuration (Continued...)
**Cisco IP Phone 9971 SIP Configuration (Continued...)**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allow EnergyWise Overrides</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port</td>
<td>Disabled</td>
</tr>
<tr>
<td>Logging Display</td>
<td>Disabled</td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Tone</td>
<td>Disabled</td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td>100</td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td>50</td>
</tr>
<tr>
<td>Recording Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td>Enabled</td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
</tr>
<tr>
<td>Remote Log</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

**Log Profile**
- Default
- Support
- Telephony

**Advertise G.722 and G.711 Codes**
- Use System Default

**Wideband Headset UT Control**
- Enabled

**Wideband Headset**
- Enabled

**Peer Firmware Sharing**
- Enabled

**Cisco Discovery Protocol (CDP): Switch Port**
- Enabled

**Cisco Discovery Protocol (CDP): PC Port**
- Enabled

**Link Layer Discovery Protocol - Media Endpoint**
- Enabled

**Link Layer Discovery Protocol (LLDP-MED): Switch Port**
- Enabled
Cisco IP Phone 9971 SIP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td>Unknown</td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td>User Controlled</td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td>Disabled</td>
</tr>
<tr>
<td>FIPS Mode</td>
<td>Normal</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td>Disabled</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>
<tr>
<td>Hide Video By Default</td>
<td>Disabled</td>
</tr>
<tr>
<td>Background Image</td>
<td></td>
</tr>
<tr>
<td>Simplified New Call ID</td>
<td>Disabled</td>
</tr>
</tbody>
</table>
Cisco IP Phone 9971 SIP Configuration (Continued...)
Set Directory Number* = 4084. This is used in this example.
Set Description = 7323204084. This is used in this example.
Set Alerting Name = Cisco 9971 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 9971 Phone. This is used in this example.
### Directory URIs

<table>
<thead>
<tr>
<th>Primary</th>
<th>URI</th>
<th>Partition</th>
<th>Advertise Globally via ILS</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td><a href="mailto:jabber@lab.tekvizion.com">jabber@lab.tekvizion.com</a></td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>

**Add Row**

---

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

### 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>✓ 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></td>
<td></td>
</tr>
<tr>
<td>Forward All</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>&lt; None &gt;</td>
<td>✓</td>
</tr>
</tbody>
</table>
Cisco IP Phone 9971 SIP Configuration (Continued...)

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>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Retrieve Destination External</td>
<td>or</td>
<td></td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Reversion Timer</td>
<td></td>
<td>Reversion Timer service parameter</td>
</tr>
</tbody>
</table>
Cisco IP Phone 9971 SIP Configuration (Continued...)

---

MLPP Alternate Party And Confidential Access Level Settings

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

Line Settings for All Devices:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hold Reversion Ring Duration (seconds)</td>
<td>Setting the Hold Reversion Ring Duration to zero will disable the feature</td>
</tr>
<tr>
<td>Hold Reversion Notification Interval (seconds)</td>
<td>Setting the Hold Reversion Notification Interval to zero will disable the feature</td>
</tr>
<tr>
<td>Party Entrance Tone*</td>
<td>Default</td>
</tr>
</tbody>
</table>

---

Line 1 on Device SEPC07EBCA181DD

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Update Shared Device Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display (Caller ID)</td>
<td>79223204085</td>
<td></td>
</tr>
<tr>
<td>ASCII Display (Caller ID)</td>
<td>79223204085</td>
<td></td>
</tr>
<tr>
<td>Line Text Label</td>
<td>79223204085</td>
<td></td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td>79223204085</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>
</tbody>
</table>

---

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

Note: Testing was conducted in tekVizion labs
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”.

[Image of SIP Profile Configuration]

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

- **SDP Session-level Bandwidth Modifier for Early Offer and Re-invites**
  - TIAS and AS
- **SDP Transparency Profile**
- **Accept Audio Codec Preferences in Received Offer**
  - Pass all unknown SDP attributes
- **Require SDP Inactive Exchange for Mid-Call Media Change**
  - Default
- **Allow RS/RS bandwidth modifier (RFC 3555)**

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)*</td>
<td>1800</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>Media Port Ranges</td>
<td></td>
</tr>
<tr>
<td>Common Port Range for Audio and Video</td>
<td></td>
</tr>
<tr>
<td>Separate Port Ranges for Audio and Video</td>
<td></td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>32766</td>
</tr>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</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-ogpickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-serviceuri-ogpickup</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>None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)*</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections*</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)*</td>
<td>15000</td>
</tr>
</tbody>
</table>
SIP Profile Configuration used by SIP trunk to Cisco UBE (Continued...)

- **Call Forward URI**
  - x-cisco-serviceuri-cfwdall

- **Speed Dial (Abbreviated Dial) URI**
  - x-cisco-serviceuri-abbrevial

- **Normalization Script**
  - Parameter Name | Parameter Value
    - Enable Trace | |

- **Incoming Requests FROM URI Settings**
  - Caller ID DN
  - Caller Name

- **Trunk Specific Configuration**
  - Reroute Incoming Request to new Trunk based on
    - Never
  - Resource Priority Namespace List
    - < None >
  - SIP Re12XX Options
    - Send PRACK if 1xx Contains SDP
  - Video Call Traffic Class
    - Mixed
  - Calling Line Identification Presentation
    - Default
  - Session Refresh Method
    - Invite
  - Early Offer support for voice and video calls
    - Disabled (Default value)
  - Enable ANAT
  - Deliver Conference Bridge Identifier
  - Allow Passthrough of Configured Line Device Caller Information
  - Reject Anonymous Incoming Calls
  - Reject Anonymous Outgoing Calls
  - Send LIS Learned Destination Route String

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

- **SIP OPTIONS Ping**
  - Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)".
  - Ping Interval for In-service and Partially In-service Trunks (seconds): 60
  - Ping Interval for Out-of-service Trunks (seconds): 120
  - Ping Retry Timer (milliseconds): 500
  - Ping Retry Count: 6

- **SDP Information**
  - Send send-receive SDP in mid-call INVITE
  - Allow Presentation Sharing using BFCP
  - Allow IX Application Media
  - Allow multiple codecs in answer SDP

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...)

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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

<table>
<thead>
<tr>
<th>Connected Party Transformation CSS</th>
<th>Use Device Pool Connected Party Transformation CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; None &gt;</td>
<td>✔</td>
</tr>
</tbody>
</table>

### 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**
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* = ATT_SIP_Profile. This is used in this example.

---

Normalization Script

Recording Information

Geolocation Configuration

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Note: Testing was conducted in tekVizion labs
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.

---

**Device Information**

- **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 DID Encoding** = No Changes
- **Packet Capture Mode** = None
- **Packet Capture Duration** = 0
Note: Testing was conducted in tekVizion labs
SIP Trunk to Fax Gateway Configuration (Continued...)

- **Incoming Calling Party Settings**
  
  If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting [DevicePool/Service Parameter]. Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

  ![Incoming Calling Party Settings Table](image)

- **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.

  ![Incoming Called Party Settings Table](image)

- **Connected Party Settings**
  
  Connected Party Transformation CSS < None >

  ![Use Device Pool Connected Party Transformation CSS](image)
**Outbound Calls**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
<td>True</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>True</td>
</tr>
<tr>
<td>Calling Party Selection</td>
<td>Originator</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling and Connected Party Info Format</td>
<td>Deliver DN only in connected party</td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Outbound</td>
<td>True</td>
</tr>
<tr>
<td>Redirecting Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Redirecting Party Transformation CSS</td>
<td>True</td>
</tr>
</tbody>
</table>

**Caller Information**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
<tr>
<td>Maintain Original Caller ID DN and Caller Name in Identity Headers</td>
<td>True</td>
</tr>
</tbody>
</table>

SIP Trunk to Fax Gateway Configuration (Continued...)
### SIP Information

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>172.16.31.60</td>
<td>5060</td>
</tr>
</tbody>
</table>

- **MTP 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-Dialing Refer Calling Search Space**: < None >
- **SUBSCRIBE Calling Search Space**: < None >
- **SIP Profile**: Standard SIP Profile w/Early Media Disabled
- **DTMF Signaling Method**: No Preference

### Normalization Script

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

### Recording Information

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

### Geolocation Configuration

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

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Note: Testing was conducted in tekVizion labs
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.

![Cisco Unified CM Administration](image)
Route Pattern Configuration (Continued...)

Pattern Definition

Route Pattern: 9[.]
Route Partition: <None>
Description: To PSTN via ATT SIP Trunk

Gateway/Route List: ATT_SIP_TRUNK

Routing Option:
- Route this pattern
- Block this pattern: No Error

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

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Note: Testing was conducted in tekVizion labs
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* = 9.*X! This is used to route to AT&T via ISR Cisco UBE.
Set Description = Network-Based Call Forwarding. This text is used to identify this Route Pattern.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
All other values are default

Note: This Route pattern is used to Activate/De-activate Network Based Call Forwarding Features.
Route Pattern Configuration (Continued...)

- **Connected Party Transformations**
  - Connected Line ID Presentation: Default
  - Connected Name Presentation: Default

- **Called Party Transformations**
  - Discard Digits: PreDot
  - Called Party Transform Mask
  - Prefix Digits (Outgoing Calls)
  - Called Party Number Type: Cisco CallManager
  - Called Party Numbering Plan: Cisco CallManager

- **ISDN Network-Specific Facilities Information Element**
  - Network Service Protocol: -- Not Selected --
  - Carrier Identification Code
  - Network Service: -- Not Selected --
  - Service Parameter Name: < Not Exist >

[Buttons: Save, Delete, Copy, Add New]
Route Pattern Configuration (Continued...)

Set Route Pattern* = 4084 this is used to route to Fax Client via Fax Gateway.
Set Description = To FAX. This text is used to identify this Route Pattern.
Set Gateway/Route List* = Trunk_SIP_FAX_Gateway. This is used for this example.
All other values are default
Route Pattern Configuration (Continued...)
### Jabber Client Configuration

**Navigation:** Device → Phone
Select Phone Type* = Cisco Unified Client services framework
Set Device Name* = CSFUser1. This is used in this example.
Set Description = jabberclient. This is used in this example.
Select Device Pool = G729 Pool. 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* = jabber. This is used for this example

Jabber Client Configuration (Contd...)
### Remote Number

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>Device Mobility Related (Information)</td>
</tr>
</tbody>
</table>

### Protocol Specific Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</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 Codec</td>
<td>711_low</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Cisco Unified Client Services Framework - Standard</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 w/Early Media Disabled</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>Feature</th>
<th>Value</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>Generate String</td>
<td></td>
</tr>
<tr>
<td>Key Order</td>
<td>RSA Only</td>
</tr>
<tr>
<td>RSA Key Size (Bits)</td>
<td>2048</td>
</tr>
<tr>
<td>EC Key Size (Bits)</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Operation Completes By</td>
<td>2016 1 21 12 (YYYY:MM:DD:HH)</td>
</tr>
<tr>
<td>Certificate Operation None</td>
<td></td>
</tr>
<tr>
<td>Status:</td>
<td></td>
</tr>
<tr>
<td>Note: Security Profile Contains Additional CAPF Settings</td>
<td></td>
</tr>
</tbody>
</table>

### Extension Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Extension Mobility</td>
<td></td>
</tr>
<tr>
<td>Log Out Profile</td>
<td>-- Use Current Device Settings --</td>
</tr>
<tr>
<td>Log In Time</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Log Out Time</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
Jabber Client Configuration (Contd...)

![Diagram of Jabber Client Configuration](image)

Note: Testing was conducted in tekVizion labs
Jabber Client Configuration (Contd...)

![Desktop Client Settings](image)

![Analytics](image)

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Note: Testing was conducted in tekVizion labs
Voicemail Port Configuration

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

![Voicemail Port Configuration Diagram]
Voicemail Port Configuration (Continued...)

Set Port Name = CiscoUM1-V11. This is used for this example.
Set Description = VoiceMail. This is used for this example.
Set Device Pool = G729 Pool
Set Directory Number* = 2295. This is used in this example.
Message Waiting Numbers Configurations

**Navigation:** Advanced Features → Voice Mail → Message Waiting

Set Message Waiting Number* = 2298
Set Message Waiting Indicator* = On
Set Message Waiting Number* = 2399
Set Message Waiting Indicator* = Off

---

**Note:** Testing was conducted in tekVizion labs.
Voicemail Pilot Configuration

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

Set Voice mail Pilot Number = 2300. This is used for this example
Set Description = VoiceMail Pilot-Default
FAX Gateway Configuration

cme.in.tekvizion.com#sh version
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
Cisco 2851 (revision 1.0) with 249856K/12288K bytes of memory.

Processor board ID FHK1137F4LY

2 Gigabit Ethernet interfaces

62 Serial interfaces

2 terminal lines

2 Channelized E1/PRI ports

4 Voice FXS interfaces

2 cisco service engine(s)

DRAM configuration is 64 bits wide with parity enabled.

239K bytes of non-volatile configuration memory.

62720K bytes of ATA CompactFlash (Read/Write)

License Info:

License UDI:

--------------------------------------------------------
Device#   PID       SN
--------------------------------------------------------
*0       CISCO2851  FHK1137F4LY

Note: Testing was conducted in tekVizion labs
Configuration register is 0x2102

cme.in.tekvizion.com#sh running-config
Building configuration...

Current configuration : 11391 bytes
!
! Last configuration change at 15:08:21 IST Sun Jan 10 2016 by cisco
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname cme.in.tekvizion.com
!
boot-start-marker
boot-end-marker
!
!
enable password tekViz10n
!
aaa new-model
!
!
aaa authentication login local_auth local

!

!

!

!

!

aaa session-id common

clock timezone IST 5 30

network-clock-participate wic 2

network-clock-participate wic 3

!

dot11 syslog

ip source-route

!

!

ip cef

!

!

!

ip host Clus1-862-Pub 172.16.26.2

no ipv6 cef

multilink bundle-name authenticated

!

!

!

!
isdn switch-type primary-qsig

!

voice rtp send-receive

!

voice service pots

!

voice service voip

no ip address trusted authenticate

allow-connections sip to sip

no supplementary-service sip handle-replaces

redirect ip2ip

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

sip

g729 annexb-all

!

voice class codec 1

codec preference 1 g729r8

codec preference 2 g711ulaw

!

voice class codec 2

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-card 0
!
!
!
!
!
!
crypto pki token default removal timeout 0
!
!
!
!
!
llicense udi pid CISCO2851 sn FHK1137F4LY
username cisco password 0 tekV1z10n
!
!
controller E1 0/2/0
   pri-group timeslots 1-31 service mgcp
!
controller E1 0/3/0
   clock source internal
   pri-group timeslots 1-31
!
ip tftp source-interface GigabitEthernet0/0
!
!
!
!
!
!
interface GigabitEthernet0/0
ip address 172.16.31.50 255.255.255.0
duplex auto
speed auto
!
interface Service-Engine0/0
no ip address
shutdown
!
interface GigabitEthernet0/1
no ip address
ip nat outside
ip virtual-reassembly in
shutdown
duplex auto
speed auto
!
interface Serial0/2/0:15
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn timer T310 120000
isdn protocol-emulate network
isdn incoming-voice voice
isdn map address .* plan isdn type national
isdn bind-l3 ccm-manager
isdn send-alerting
isdn sending-complete
no cdp enable
!
interface Serial0/3/0:15
no ip address
encapsulation hdlc
isdn switch-type primary-qsig
isdn timer T310 120000
isdn protocol-emulate network
isdn incoming-voice voice
no cdp enable
!
interface Service-Engine1/0
no ip address
shutdown
!
ip forward-protocol nd
!
ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.16.31.1
!
access-list 1 permit 172.16.31.0 0.0.0.255
!
snmp-server community public RO
snmp-server location Chennai
!
!
!
!
control-plane
!
!
!
voice-port 0/0/0
no vad
shutdown
!
voice-port 0/0/1
no vad
shutdown
!
voice-port 0/3/0:15
!
voice-port 0/2/0:15

!

voice-port 0/1/0

no vad

shutdown

!

voice-port 0/1/1

cptone IN

station-id number 7323204084

caller-id enable

!

no mgcp timer receive-rtcp

mgcp behavior rsip-range tgcp-only

mgcp behavior comedia-role none

mgcp behavior comedia-check-media-src disable

mgcp behavior comedia-sdp-force disable

!

mgcp profile default

!

!

!

!

!

!

!

!

!

dial-peer voice 777 pots

huntstop

service session
destination-pattern 4084
no digit-strip
port 0/1/1
forward-digits all
!
dial-peer voice 9224 voip
description CUCM to Gateway
service session
session protocol sipv2
session transport udp
incoming called-number 4084
voice-class codec 3
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 92240 voip
description Gateway to CUCM
service session
destination-pattern 9T
session protocol sipv2
session target ipv4:10.80.22.2
session transport udp
voice-class codec 3
dtmf-relay rtp-nce
fax-relay sg3-to-g3
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!

!
!

! gateway
!
timer receive-rtp 1200
!
!

sip-ua
credentials username +19728522671 password 7 11584B5643475D realm bsrhelas.lab.tekvizion.com
no remote-party-id
retry register 5
timers connection aging 30
timers update 1000
no timers hold
timers register 1000
!
!
telephony-service
max-ephones 50
max-dn 60
ip source-address 172.16.31.50 port 2000
service phone sshAccess 0
cnf-file perphone
max-conferences 8 gain -6
web admin system name Administrator password tekV1z10n
transfer-system full-consult
create cnf-files version-stamp 7960 Nov 22 2013 19:05:58
  
ephone-dn 1
  
ephone-dn 2
  
ephone-dn 5
  
ephone-dn 6 dual-line
  
ephone-dn 10
  

ephone 1
mac-address 001D.A21A.3577
type 7961
button 1:1
!
!
!
ephone 2
mac-address 001D.A21A.291D
type 7961
button 1:2
!
!
!
ephone 5
mac-address 0008.21DF.CEC5
type 7940
ssh userid cisco password cisco
button 1:5
!
!
!
ephone 6
!
!
!
Note: Testing was conducted in tekVizion labs

dephone 10
mac-address 0C27.2431.5FB9
button 1:10
!
!
!
!
banner login ^CC

==================================================
WELCOME to CISCO CME 8.6
==================================================

Cisco IOS Software (C2800NM-IPVOICEK9-M)
Version 15.1(4)M5, RELEASE SOFTWARE (fc1)
Cisco 2851 with 249856K/12288K bytes of memory
Processor board ID FHK1137F4LY
2 Gigabit Ethernet interfaces
2 terminal lines
2 Voice FXS interfaces
2 cisco service engine(s)
239K bytes of non-volatile configuration memory
62720K bytes of ATA CompactFlash (Read/Write)

Warning: Access is restricted.
All user activity is logged!
For support, contact 'kkumaraguru@tekvizion.com'
^C
!
line con 0
line aux 0
line 66
no activation-character
no exec
transport preferred none
transport input all
transport output pad telnet rlogin lapb-ta mop udp v120 ssh
line 194
no activation-character
no exec
transport preferred none
transport input all
transport output all
line vty 0 4
session-timeout 180
exec-timeout 0 0
password tekV1z10n
login authentication local_auth
transport input all
!
scheduler allocate 20000 1000
ntp server 103.6.16.254
Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)
CUC Telephony Integration with Cisco UCM

**Navigation:** Telephony Integrations → Phone system

Set Phone System Name* = cucmUM1. This is used for this example
CUC Port Group

**Navigation:** Telephony Integration → Port Group
### CUC Port Group (continued...)

<table>
<thead>
<tr>
<th>Port Group Name</th>
<th>Phone System Display Name</th>
<th>Port Count</th>
<th>Integration Method</th>
<th>Needs Reset</th>
</tr>
</thead>
<tbody>
<tr>
<td>CiscoUM1-1</td>
<td>CiscoUM1</td>
<td>2</td>
<td>SCCP (Skinny)</td>
<td>No</td>
</tr>
<tr>
<td>SIP_VM-1</td>
<td>SIP_VM</td>
<td>2</td>
<td>SIP</td>
<td>No</td>
</tr>
</tbody>
</table>
Set Display Name* = cucmUM-1. This is used in this example.
Check Enable Message waiting indicators.
Set MWI on Extension = 2298. This is used in this example.
Set MWI off Extension= 299. This is used in this example.

CUC Port Settings
CUC Sample User Basic Settings

**Navigation:** Cisco Unity connection → Users → Users
Set Alias = 4084. This is one of the extension used for this testing.
Set Extension = 4084. This is used for this example.

CUC Sample User Basic Settings (Continued...)

Set Partition = clus32unity partition. This is used for this example.
Select Search Scope = clus32unity Search Scope.
Select Phone System = cucmUM1.

CUC Sample User Basic Settings (Continued...)

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Note: Testing was conducted in tekVizion labs
<table>
<thead>
<tr>
<th>Distribution Lists</th>
<th>Call Management</th>
<th>Message Storage</th>
<th>Networking</th>
<th>Unified Messaging</th>
<th>Video</th>
<th>Dial Plan</th>
<th>System Settings</th>
<th>Telephony Integrations</th>
<th>Tools</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use System Default Time Zone</td>
<td>Time Zone: (GMT-06:00) America/Chicago</td>
<td>Language: Use System Default Language</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Department</td>
<td>Manager</td>
<td>Billing ID</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Directory URI</td>
<td>Corporate Phone Number</td>
<td></td>
</tr>
</tbody>
</table>

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

**Navigation:** Call Management → System Call Handlers

Set Display Name = Demo auto attend. This is used for this example.
Set Phone System = CUCM
Set Extension=2999. This number is used as Auto attendant on this set up.
Set Partition = Clus32unity Partition. This is used for this example.

![Image of Cisco Unity Connection Administration interface](image)

<table>
<thead>
<tr>
<th>Display Name</th>
<th>Extension</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Attendant</td>
<td>2555</td>
</tr>
<tr>
<td>Goodbye</td>
<td></td>
</tr>
<tr>
<td>Opening Greeting</td>
<td></td>
</tr>
<tr>
<td>Operator</td>
<td>0</td>
</tr>
</tbody>
</table>
Auto Attendant (Continued...)

[Diagram of Cisco Auto Attendant configuration page with highlighted fields: Display Name, Creation Time, Phone System, Call Handler, Active Schedule, Time Zone, Language, Extension, Partition, Recorded Name, Search Scope, and Search Space.

Note: Testing was conducted in tekVizion labs]
Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP)

CUP/IMP Version

![Cisco Unified CM IM and Presence Administration Interface]

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

User administrator last logged in to this cluster on Wednesday, January 6, 2016 5:09:09 AM CST, to node 10.80.22.3, from 172.16.29.115 using HTTPS

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A summary of U.S. laws governing Cisco cryptographic products may be found at our Export Compliance Product Report web site.

For information about Cisco Unified CM IM and Presence please visit our IM and Presence Documentation web site.

Note: Testing was conducted in tekVizion labs
Presence Topology

**Navigation:** System → Presence Topology

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Note: Testing was conducted in tekVizion labs
Node Configuration

Navigation: System → Cluster Topology → Fully Qualified Domain Name
Users

Navigation: System → Cluster Topology → clus32imp.lab.tekvizion.com → Users

<table>
<thead>
<tr>
<th>User ID</th>
<th>First Name</th>
<th>Last Name</th>
<th>IM Address</th>
<th>Directory URI</th>
<th>Failed Over</th>
<th>Node</th>
<th>Presence Redundancy Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>jabber</td>
<td>cisco</td>
<td></td>
<td><a href="mailto:jabber@lab.tekvizion.com">jabber@lab.tekvizion.com</a></td>
<td><a href="mailto:jabber@lab.tekvizion.com">jabber@lab.tekvizion.com</a></td>
<td></td>
<td>clus32imp.lab.tekvizion.com</td>
<td>DefaultCUPSSubcluster</td>
</tr>
</tbody>
</table>
Presence gateway configuration

**Navigation:** Presence → Gateways

Set Presence Gateway Type *= CUCM
Set Description *= Cluster 32 9.1.2. This is used for this example.
Presence Gateway *= clus23pubsub.lab.tekvizion.com
## Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</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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