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.10.5.2 and Cisco UBE v.10.0.2 on an ASR Router with SIP Interface

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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) 10.5.2, Cisco Unity Connection 10.5.2, Cisco Unified CM IM and Presence 10.5.2, Cisco Aggregation Services Routers (ASR) Version 15.4(3) S1 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 Aggregation Services Routers (ASR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service. Network Topology between Cisco ASR and AT&T Flexible reach SIP is deployed using IPV6.

- 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 Aggregation Services 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 ASR 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.
Hardware Components

- Cisco UCS-C240 VMWare server running ESXi 5.5
- Cisco IP Phones. This solution was tested with Cisco 7965, Cisco 7975 and Cisco 9971 phones
- Cisco Aggregation Services Router - Cisco ASR1001 (1RU) processor (revision 1RU) with 1089559K/6147K bytes of memory.
- Processor board ID SSI17360FV1
  4 Gigabit Ethernet interfaces
  32768K bytes of non-volatile configuration memory
  4194304K bytes of physical memory
  7741439K bytes of eUSB flash at bootflash

Software Requirements

- Cisco ASR: Cisco IOS Software, ASR1000 Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.4(3)S1, RELEASE SOFTWARE (fc3)
- System image file is "bootflash:/asr1001-universalk9.03.13.01.S.154-3.31-ext.bin"
- Cisco Unity Connection version: System version: 10.5.2.10000-5
- Cisco Unified CM IM and Presence: System version: 10.5.2.10000-9
- Cisco Jabber client version: 10.5.0 Build 37889
- VentaFax client version: 7.4.237.590
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 T.38 and G.711
- 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 911 calls were terminated to a voicemail platform in lab environment within AT&T for test

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 Ringing and Simultaneous Ringing features. Provisioning is self-explanatory. AT&T representative can be contacted for any help required in this regard.

Features - Not Supported

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

Fax
- The maximum fax rate achieved using (G3 or SG3) is only 14400 kbps.
- G711Passthrough test is achieved using “fax protocol pass-through g711ulaw”.
- Fax protocol T38 has been tested.

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 executed with limitations on AT&T network
- IPTC - Hold & Resume
- IPTC - PBX-Based Attended Transfer
- IPTC - PBX-Based Three-way Call Conference
Configuration Considerations

• To enable conference using G729 on AT&T IP Flexible Reach and Cisco UCM SIP trunk, it is required to configure a conference bridge (CFB) resource to initiate a three-way conference between end-points. See configuration section for details.

• Forwarded calls from Cisco UCM user to PSTN (out to AT&T’s IP Flexible Reach service), AT&T serviced areas require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption has been made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using Cisco UBE translation profile. This is because the Cisco UCM uses 4-digit extensions on Cisco UCM IP phones and it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP profile in Cisco UBE. See configuration section for details.

• Upon receiving inbound calls, AT&T SIP network will always have the first choice codec presented in the initial SIP INVITE (unless the end-device does not support the listed preferred codec), and processes calls accordingly. Customers wishing to place/receive G.711-only calls must configure separate voice class codec on Cisco UBE with G.711 as the first choice.

• SIP Profiles may also be employed to advertise desired RTP payload packet size.

• “voice-class sip privacy id” needs to configure in Cisco UBE dial peer to make call From a CPE Phone to PSTN phone.

• 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 SIP UPDATE message. This causes the Cisco UBE to timeout and hence 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.

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ASR Configuration

ATT-ASR-IPv6-IPFR#show version

Cisco IOS XE Software, Version 03.13.01.S - Extended Support Release

Cisco IOS Software, ASR1000 Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.4(3)S1, RELEASE SOFTWARE (fc3)

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

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ROM: IOS-XE ROMMON

ATT-ASR-IPv6-IPFR uptime is 1 week, 6 days, 16 hours, 16 minutes

Uptime for this control processor is 1 week, 6 days, 16 hours, 17 minutes

System returned to ROM by reload
System image file is "bootflash:/asr1001-universalk9.03.13.01.S.154-3.S1-ext.bin"

Last reload reason: Watchdog

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License Level: advipservices
License Type: Permanent
Next reload license Level: advipservices

cisco ASR1001 (1RU) processor (revision 1RU) with 1089559K/6147K bytes of memory.
Processor board ID SSI17360FV1
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7741439K bytes of eUSB flash at bootflash:

Configuration register is 0x2102

ATT-ASR-IPv6-IPFR#sh running-config
Building configuration...

Current configuration : 9357 bytes
!
! Last configuration change at 21:43:19 UTC Wed Jul 1 2015 by cisco
!

version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname ATT-ASR-IPv6-IPFR
!
boot-start-marker
boot system flash bootflash:asr1001-universalk9.2014-12-23_06.19_raghs.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
no logging console
no logging monitor
enable secret 4 Pe0NhiWw5IXZpE.k5VhTSCoGPcuVeRyrer9kEPz20Z6
!
no aaa new-model
!
!
no ip domain lookup

!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
subscriber templating
!
!
!
!
!
!
!
!
multilink bundle-name authenticated
!
!
!
!
!
!
!
!
voice service voip
no ip address trusted authenticate
address-hiding\(^1\)
mode border-element\(^2\)

\(^{1}\) Hide signaling and media peer addresses from endpoints other than gateway.
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
early-offer forced
no silent-discard untrusted
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
voice class codec

codec preference 1 g729r8 bytes 30

---

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.

9 This command configures the codec preference to be assigned to dial-peers. Alternatively, single code can be configured into individual dial-peers.
codec preference 2 g711ulaw
!
voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8 bytes 30
!
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
!
!
!
!
voice iec syslog
!
!
!
!
voice translation-rule 1\13

\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.
rule 1 /\^\+(1\)\(7\.........\)\//\^2/
rule 2 /\^\+\(435\)\//732320\1/

!

voice translation-profile AddPlusOne
translate called 1

!

voice translation-profile NPA
translate calling 1

!

!

!

!

!

license udi pid ASR1001 sn JAE17430GQ5
license boot level advipservices
spanning-tree extend system-id

!

username cisco privilege 15 password 0 cisco

!

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.
! cdp run!
! interface GigabitEthernet0/0/0
    description Wan Interface
    no ip address
    negotiation auto
    ipv6 address 2620:96:C000:4::10/64
    ipv6 enable
    cdp enable

interface GigabitEthernet0/0/1
    description Lan Interface
    ip address 10.80.14.5 255.255.255.0
    negotiation auto

---
14 Cisco UBE WAN interface to AT&T
15 Cisco UBE WAN interface IPv6 Address
16 Cisco UBE LAN interface to Cisco UCM
17 Cisco UBE LAN interface IPv4 Address

Note: Testing was conducted in tekVizion labs
cdp enable
!
interface GigabitEthernet0/0/2
  no ip address
  shutdown
  negotiation auto
!
interface GigabitEthernet0/0/3
  no ip address
  shutdown
  negotiation auto
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
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
!
ipv6 route ::/0 2620:96:C000::1

control-plane

dial-peer voice 300 voip
description " Int'l calls to AT&T - AT&T facing side "
destination-pattern 011T
session protocol sipv2
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 GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
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 800 voip
description "Incoming AT&T to IP-PBX . AT&T facing side"
huntstop
session protocol sipv2
incoming called-number [27][37][27]......
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 bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
fax rate 14400
fax nsf 000000
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
session protocol sipv2
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 GigabitEthernet0/0/0\(^{18}\)
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n te
fax nsf 000000
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 122 voip
description "OPERATOR TESTING"
destination-pattern 0
session protocol sipv2
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru

\(^{18}\) 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. By enabling these commands, Cisco UBE Wan Interface is bound.
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nge
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"
destination-pattern *..
session protocol sipv2
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 GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nge
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
! 
dial-peer voice 2151 voip
description " Incoming AT&T to IP-PBX - IP-PBX facing side "
destination-pattern 7322162...
session protocol sipv2
session target ipv4:10.80.14.2:5060
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 bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
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 700 voip

description " Incoming AT&T to IP-PBX - IP-PBX facing side "
huntstop
destination-pattern 7323680...
session protocol sipv2
session target ipv4:10.80.14.2:5060
voice-class codec 1

19 Dial-peer facing AT&T Network
20 Session protocol SIPv2 is used for this testing.
voice-class sip asymmetric payload full\textsuperscript{22}
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru\textsuperscript{23}
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/1\textsuperscript{24}
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nce\textsuperscript{25}
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none\textsuperscript{26}
no vad
!
dial-peer voice 400 voip
description " Int'l calls to AT&T - IP-PBX facing side "
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

\textsuperscript{21} Assigns voice class codec 1 settings to dial-peer (codec support and filtering).
\textsuperscript{22} Configures the dynamic SIP asymmetric payload support.
\textsuperscript{23} 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.
\textsuperscript{24} 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. The Cisco UBE LAN interface is bind using these commands.
\textsuperscript{25} This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call.
\textsuperscript{26} This command enables T38 fax protocol for calls terminating on this dial-peer.
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
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 500 voip
description " N11 Calls to AT&T - AT&T facing side "
destination-pattern .11
session protocol sipv2
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 GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-npe
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
description " N11 Calls to AT&T - IP-PBX facing side "
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 bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
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 2152 voip
description " Incoming AT&T to IP-PBX - AT&T facing side "
session protocol sipv2
incoming called-number 7322162...
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 bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
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
protocol mode dual-stack
!
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
login local
!
!
end
Cisco UCM Configuration

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

Cisco UCM Version

![Cisco Unified CM Administration](image-url)
Cisco UCM Audio Codec Preference List

Navigation Path: System → Region Information → Audio codec preference list

Cisco UCM 10.5.2 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

---

### Region Information

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
<th>Maximum Session Bit Rate for Immersive Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G729 Preferred codec list</td>
<td>64 kbps (G.722, G.711)</td>
<td>384 kbps</td>
<td>2147483647 kbps</td>
</tr>
<tr>
<td>G729</td>
<td>G729 Preferred codec list</td>
<td>64 kbps (G.722, G.711)</td>
<td>384 kbps</td>
<td>2147483647 kbps</td>
</tr>
</tbody>
</table>

**Note:** Regions not displayed

### Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
<th>Maximum Session Bit Rate for Immersive Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>G711</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>G729</td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

---

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

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

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

- **Geolocation Configuration**
  - Geolocation: < None >
  - Geolocation Filter: < None >

- **Call Routing Information**
  - **Incoming Calling Party Settings**
    - If the administrator sets the prefix to Default, this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.
    - **Clear Prefix Settings**
    - **Default Prefix Settings**
    - **Number Type**
      - National Number: Default
      - International Number: Default
      - Unknown Number: Default
      - Subscriber Number: Default
    - **Prefix**
    - **Strip Digits**
    - **Calling Search Space**
      - National Number: < None >
      - International Number: < None >
      - Unknown Number: < None >
      - Subscriber Number: < None >
Device Pool Configuration (Contd.)

![Incoming Called Party Settings Diagram]

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

### Clear Prefix Settings

<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>

### Phone Settings

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

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

- **Redirecting Party Settings**
  - **Redirecting Party Transformation CSS**: < None >

![Save, Delete, Copy, Reset, Apply Config, Add New Buttons]

Note: Testing was conducted in tekVizion labs
Annunciator Configuration

**Navigation:** Media Resource → Annunciator

Set Name* = ANN_2.
Set Description = ANN_clus24pubsub. This is used for this example
Set Device Pool* = G729

![Cisco Unified CM Administration](image)

**Status**

- Status: Ready

**Annunciator Information**

- **Registration:** Registered with Cisco Unified Communications Manager clus24pubsub.lab.tekvizion.com
- **IPv4 Address:** 10.80.14.2
- **Device is trusted**
- **Server:** clus24pubsub.lab.tekvizion.com
- **Name:** ANN_2
- **Description:** ANN_clus24pubsub
- **Device Pool:** G729
- **Location:** Hub_None
- **Use Trusted Relay Point:** Off
Conference Bridge Configuration

**Navigation:** Media Resources → Conference Bridge

Set Conference Bridge Type* = Cisco Conference Bridge Software.
Set Host Server = clus24pubsub.lab.tekvizion.com. This is used for this example.
Set Conference Bridge Name* = CFB_2.
Set Description = CFB_clus24pubsub. This is used in this example.
Set Device Pool* = G729
Media Termination Point Configuration

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

Set Media Termination Point Name* = MTP_2
Set Description = MTP_clus24pubsub. This is used for this example
Set Device pool* = G729
Music on Hold Server Configuration

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

Set Music on Hold Server Name* = MOH_2.
Set Description = MOH_clus24pubsub. This is used for this example.
Set Device Pool* = G729

---

### Music On Hold (MOH) Server Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
</table>

### Device Information

- **Registration:** Registered with Cisco Unified Communications Manager clus24pubsub.lab.tekvizion.com
- **IPv4 Address:** 10.80.14.2
- **Device is trusted**
- **Host Server:** clus24pubsub.lab.tekvizion.com
- **Music On Hold Server Name**: MOH_2
- **Description**: MOH_clus24pubsub
- **Device Pool**: G729
- **Location**: Hub_None
- **Maximum Half Duplex Streams**: 250
- **Maximum Multi-cast Connections**: 250000
- **Fixed Audio Source Device**: 
- **Use Trusted Relay Point**: Off
- **Run Flag**: Yes

### Multi-cast Audio Source Information

- **Enable Multi-cast Audio Sources on this MOH Server**: 
- **Base Multi-cast IP Address**: 0.0.0.0
- **Base Multi-cast Port Number**: 0 (Even numbers only)
- **Increment Multi-cast on**: Port Number

### Selected Multi-cast Audio Sources

There are no Music On Hold Audio Sources selected for Multi-casting. Click Configure Audio Sources in the top right corner of the page to select Multi-cast Audio Sources.
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 mu-law are configured in parameter Supported MOH Codecs.
Select Server* = clus24pubsub.lab.tekvizion.com (Active). This is used in this example.
Select Service* = Cisco IP Voice Media Streaming App (Active).

![Parameter Configuration](Image)

- **Select Server and Service**
  - Server*: clus24pubsub.lab.tekvizion.com → CUCM Voice/Voice
  - Service*: Cisco IP Voice Media Streaming App (Active)

- **Clusterwide Parameters (Parameters that apply to all servers)**
  - **Supported MOH Codecs**
    - 711 mulaw
    - 711 alaw
    - 729 Annex A
  - **MOH Fixed Audio Quality Level**
    - Medium Quality
  - **IP DSCP to Cisco Unified Communications Manager**
    - CS3(precedence 3) DSCP (011000)
  - **Multicast MOH IP DSCP**
    - EF DSCP (101110)
  - **MTP DTMF Duration**
    - 100
  - **MTP DTMF Power (volume)**
    - 9

There are hidden parameters in this group. Click on Advanced button to see hidden parameters.
Music on Hold Service (Duplex Streaming) Parameter Settings

Navigation: System → Service Parameter

Select Server* = clus24pubsub.lab.tekvizion.com (Active). This is used in this example.
Select Service* = Cisco Call Manager (Active).
Select Duplex Streaming Enabled * = True
Media Resource Group Configuration

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

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

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

Navigation Path: Media Resources → Media Resource Group List

Set Name = MRGL_MTP.
Set selected Media Resource Groups = MRG_MTP.
**UC Service Configuration**

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

---

**Cisco Unified CM Administration**

**Navigation:** Cisco Unified CM Administration → administrator → Search Documentation → About → UC Service

---

**Find and List UC Services**

- **Add New**
- **Select All**
- **Clear All**
- **Delete Selected**

**Status:**

- 3 records found

---

**UC Service (1 - 3 of 3)**

<table>
<thead>
<tr>
<th>Name</th>
<th>UC Service Type</th>
<th>Product Type</th>
<th>Host/IPv Address</th>
<th>Port</th>
<th>Protocol</th>
</tr>
</thead>
<tbody>
<tr>
<td>CTI_SRV</td>
<td>CTI</td>
<td>CTI</td>
<td>10.80.14.2</td>
<td>2748</td>
<td>TCP</td>
</tr>
<tr>
<td>IM_SRV</td>
<td>IM and Presence</td>
<td>Unified CM (IM and Presence)</td>
<td>10.80.14.3</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Unity Conn</td>
<td>Voicemail</td>
<td>Unity</td>
<td>10.80.14.4</td>
<td>443</td>
<td>HTTP</td>
</tr>
</tbody>
</table>

---

**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 Client. 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)
UC Service Configuration (Contd.)

Select UC Service Type: = Voicemail
Set Name* = Unity Connection. This is used in this example.
Set Description = Voicemail. This is used in this example.
Set Host Name/IP Address* = 10.80.14.4 (Cisco Unity Connection Administration)
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.)

#### Conferencing Profile

<table>
<thead>
<tr>
<th></th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Secondary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Server Certificate Verification</td>
<td>Any</td>
</tr>
</tbody>
</table>

**Credentials source for web conference service** Not set

#### Directory Profile

<table>
<thead>
<tr>
<th></th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Secondary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

- **Use UDS for Contact Resolution**
- **Use Logged On User Credential**

**Username**

administrator

**Password**

**********

**Search Base 1**

**Search Base 2**

**Search Base 3**

- **Recursive Search on All Search Bases**

**Search Timeout (seconds)**

5

**Base Filter (Only used for Advance Directory)**

**Predictive Search Filter (Only used for Advance Directory)**

#### IM and Presence Profile

<table>
<thead>
<tr>
<th></th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td>IMP_SRV</td>
</tr>
<tr>
<td>Secondary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

#### CTI Profile

<table>
<thead>
<tr>
<th></th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td>CTI_SRV</td>
</tr>
<tr>
<td>Secondary</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Tertiary</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>
Service Profile Configuration (Contd.)

```
<p>| | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Primary</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
<tr>
<td>Secondary</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
<tr>
<td>Tertiary</td>
<td>&lt;None&gt;</td>
<td></td>
</tr>
</tbody>
</table>
```

*- indicates required item.
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 (Contd.)

![User Information](image)

- **User Status**: Enabled Local User
- **User ID**: jabber1
- **Password**: ***************
- **Confirm Password**: ***************
- **Self-Service User ID**: 
- **PIN**: 
- **Confirm PIN**: 
- **Last Name**: cisco
- **Directory URI**: jabber1@lab.tekvizion.com
- **Manager User ID**: 
- **Department**: 
- **User Locale**: < None >
- **Associated PC**: 
- **Digest Credentials**: 
- **Confirm Digest Credentials**: 
- **User Profile**: Use System Default (Standard (Factory Default)) User
End User Configuration (Contd.)

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

- **Device Information**
  - **Controlled Devices**
    - CSFUser1
    - SEP00083031F2A8
  - **Device Association**
    - Line Appearance Association for Presence

- **Extension Mobility**
  - **Available Profiles**
  - **Controlled Profiles**
  - **Default Profile**
    - Not Selected
  - **BLF Presence Group**
    - Standard Presence Group
  - **SUBSCRIBE Calling Search Space**
    - None
  - **Allow Control of Device from CTI**
  - **Enable Extension Mobility Cross Cluster**

- **Directory Number Associations**
  - **Primary Extension**
    - None
End User Configuration (Contd.)
Cisco IP Phone 7965 SCCP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = SCCP_Chen_7965. This text is used to identify this Phone.
Set Device Pool* = G729. 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 (Contd.)

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 (Contd.)

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

<table>
<thead>
<tr>
<th>External Data Locations Information (Leave blank to use default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>-------------</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Extension Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Extension Mobility</td>
</tr>
<tr>
<td>Log In Time</td>
</tr>
<tr>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Do Not Disturb</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>----------------</td>
</tr>
<tr>
<td>Use Common Phone Profile Setting</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Secure Shell Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell User</td>
</tr>
<tr>
<td>administrator</td>
</tr>
</tbody>
</table>

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

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>Forwarding</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Delay</td>
<td></td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td>Enabled</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>Video Capabilities</td>
<td>Disabled</td>
<td></td>
</tr>
<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>Display On Time</td>
<td>07:30</td>
<td></td>
</tr>
</tbody>
</table>

Display On Duration 10:30
Display Idle Timeout 01:00
Enable Power Save Plus Sunday Monday Tuesday
Phone On Time 00:00
Phone Off Time 24:00
Phone Off Idle Timeout 60
Enable Audible Alert
EnergyWise Domain
EnergyWise Endpoint Security Secret
Allow EnergyWise Overides
Span to PC Port Disabled
Logging Display PC Controlled
Load Server

Recording Tone Disabled
Recording Tone Local Volume 100
Recording Tone Remote Volume 50
Recording Tone Duration
Display On When Incoming Call Disabled
RTCP Disabled

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

Set Directory Number* = 0461. This is used in this example.
Set Description = Cisco 7965 Phone. 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 (Contd.)

### Directory Number Settings

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Mail Profile</td>
<td>&lt; None &gt;</td>
<td>(Choose &lt;None&gt; to use system default)</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>BLF Presence Group</td>
<td>Standard Presence group</td>
<td></td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Auto Answer</td>
<td>Auto Answer Off</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Calls</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Enterprise Alternate Number

- Add Enterprise Alternate Number

### +E.164 Alternate Number

- Add +E.164 Alternate Number

### Directory URIs

<table>
<thead>
<tr>
<th>Primary</th>
<th>URI</th>
<th>Partition</th>
<th>Advertise Globally via 115</th>
<th>Remove</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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

Advertised Failover Number: < None >

### AAR Settings

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Voice Mail</td>
<td></td>
<td></td>
</tr>
<tr>
<td>AAR Destination Mask</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>AAR</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Retain this destination in the</td>
<td></td>
<td></td>
</tr>
<tr>
<td>call forwarding history</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Cisco IP Phone 7965 SCCP Configuration (Contd.)

### 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><strong>Calling Search Space Activation Policy</strong></td>
<td></td>
<td>Use System Default</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></td>
<td>&lt; None &gt;</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>

**Forward on CTI Failure**

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward Unregistered Internal</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**No Answer Ring Duration (seconds)**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<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 Forward No Retrieve Destination External</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Park Monitoring Forward No Retrieve Destination Internal</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Reversion Timer**

<p>| | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring Reversion Timer service parameter</td>
<td>A blank value will use value set in Park Monitoring</td>
</tr>
</tbody>
</table>

---

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

Set Display (Caller ID) = Cisco 7965-Phone. This is used in this example.
Set ASCII Display (Caller ID) = Cisco 7965-Phone. This is used in this example.
Set Line Text Label = Cisco 7965-Phone. This is used in this example.
Set External Phone Number Mask = 7323680461. This is used in this example.
Cisco IP Phone 7965 SCCP Configuration (Contd.)

- Call Pickup Group: Use System Default
- Audio Alert Setting (Phone Active):
  - Recording Option: Call Recording Disabled
  - Recording Profile: < None >
  - Recording Media Source: Gateway Preferred
  - Monitoring Calling Search Space: < None >
  - Log Missed Calls

**Multiple Call/Call Waiting Settings on Device SEP08CC68E9FAB3**

- 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 SEP08CC68E9FAB3**

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

**Users Associated with Line**

- Associate End Users

- Save  Delete  Reset  Apply Config  Add New

**Notes:**
- * indicates required item.
- ** indicates changes to Line or Directory Number settings require restart.
Cisco IP Phone 7975 SIP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = SIP_Chen_ 7975. This text is used to identify this Phone.
Set Device Pool* = G729. This is used in this example.
Set Phone Button Template* = Standard 7975 SIP. This is used in this example.
Set Soft key Template = Standard User. 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

Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7975 SIP Configuration (Contd.)

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

```
<table>
<thead>
<tr>
<th>Malicious Call Identification</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>26</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>27</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>28</td>
<td>Network Local</td>
</tr>
<tr>
<td>29</td>
<td>Default</td>
</tr>
<tr>
<td>30</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>31</td>
<td>Privacy</td>
</tr>
<tr>
<td>32</td>
<td>Default</td>
</tr>
<tr>
<td>33</td>
<td>Device Mobility Mode*</td>
</tr>
<tr>
<td>34</td>
<td>Default</td>
</tr>
<tr>
<td>35</td>
<td>Default</td>
</tr>
<tr>
<td>36</td>
<td>None</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Owner</th>
<th>User</th>
<th>Anonymous (Public/Shared Space)</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Phone Load Name</th>
<th>Single Button Barge</th>
<th>Join Across Lines</th>
<th>Use Trusted Relay Point*</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Default</td>
<td>Default</td>
<td>Default</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>BLF Audible Alert Setting (Phone Idle)*</th>
<th>BLF Audible Alert Setting (Phone Busy)*</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Always Use Prime Line*</th>
<th>Always Use Prime Line for Voice Message*</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Geo location</th>
</tr>
</thead>
<tbody>
<tr>
<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
- Protected Device
- Hot line Device
- Require off-premise location
```
Cisco IP Phone 7975 SIP Configuration (Contd.)

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
BLF Presence Group
SIP Dial Rules
MTP Preferred
Originating Codec
Device Security Profile
Re-routing Calling Search Space
SUBSCRIBE Calling Search Space
SIP Profile
Digest User

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.
### Cisco IP Phone 7975 SIP Configuration (Contd.)

<table>
<thead>
<tr>
<th>Expansion Module Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module 1</td>
</tr>
<tr>
<td>Module 1 Load Name</td>
</tr>
<tr>
<td>Module 2</td>
</tr>
<tr>
<td>Module 2 Load Name</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>External Data Locations Information (Leave blank to use default)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Information</td>
</tr>
<tr>
<td>Directory</td>
</tr>
<tr>
<td>Messages</td>
</tr>
<tr>
<td>Services</td>
</tr>
<tr>
<td>Authentication Server</td>
</tr>
<tr>
<td>Proxy Server</td>
</tr>
<tr>
<td>Idle</td>
</tr>
<tr>
<td>Idle Timer (seconds)</td>
</tr>
<tr>
<td>Secure Authentication URL</td>
</tr>
<tr>
<td>Secure Directory URL</td>
</tr>
<tr>
<td>Secure Idle URL</td>
</tr>
<tr>
<td>Secure Information URL</td>
</tr>
<tr>
<td>Secure Messages URL</td>
</tr>
<tr>
<td>Secure Services URL</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Extension Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Enable Extension Mobility</td>
</tr>
<tr>
<td>Log Out Profile</td>
</tr>
<tr>
<td>-- Use Current Device Settings --</td>
</tr>
<tr>
<td>Log in Time</td>
</tr>
<tr>
<td>Log out Time</td>
</tr>
</tbody>
</table>

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

<table>
<thead>
<tr>
<th>Do Not Disturb</th>
</tr>
</thead>
<tbody>
<tr>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>DND Option</td>
</tr>
<tr>
<td>DND Incoming Call Alert</td>
</tr>
</tbody>
</table>
Cisco IP Phone 7975 SIP Configuration (Contd.)

<table>
<thead>
<tr>
<th>Secure Shell Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell User:         administrator</td>
</tr>
<tr>
<td>Secure Shell Password:    Password masked</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Product Specific Configuration Layout</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameter Value</td>
</tr>
<tr>
<td>-----------------</td>
</tr>
<tr>
<td>Disable Speakerphone</td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
</tr>
<tr>
<td>Forwarding Delay</td>
</tr>
<tr>
<td>PC Port</td>
</tr>
<tr>
<td>Settings Access</td>
</tr>
<tr>
<td>Gettuts Access</td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
</tr>
<tr>
<td>Auto Line Select</td>
</tr>
<tr>
<td>Web Access</td>
</tr>
<tr>
<td>Days Display Not Active</td>
</tr>
<tr>
<td>Display On Time</td>
</tr>
<tr>
<td>Display On Duration</td>
</tr>
<tr>
<td>Display Idle Timeout</td>
</tr>
<tr>
<td>Span to PC Port</td>
</tr>
<tr>
<td>Logging Display</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Load Server</th>
<th></th>
</tr>
</thead>
<tbody>
<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 Tone Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td>Disabled</td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>5</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7975 SIP Configuration (Contd.)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codes</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Wideband Headset UL 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></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media</td>
<td></td>
</tr>
<tr>
<td>Discover (LLDP-MED): Switch Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP):</td>
<td>Enabled</td>
</tr>
<tr>
<td>PC Port</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td>Unknown</td>
</tr>
<tr>
<td>Wireless Headset Handset 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>
<tr>
<td>Headset 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>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7975 SIP Configuration (Contd.)

Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7975 SIP Configuration (Contd.)

Set Directory Number* = 2754. This is used in this example.
Set Description = Cisco 7975-Phone. This is used in this example.
Set Alerting Name = Cisco 7975-Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 7975-Phone. This is used in this example.

<table>
<thead>
<tr>
<th>Directory Number Information</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Directory Number</strong></td>
<td>2754</td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>Cisco 7975-Phone</td>
</tr>
<tr>
<td><strong>Alerting Name</strong></td>
<td>Cisco 7975-Phone</td>
</tr>
<tr>
<td><strong>ASCII Alerting Name</strong></td>
<td>Cisco 7975-Phone</td>
</tr>
<tr>
<td><strong>External Call Control Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Allow Control of Device from CTI</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Associated Devices</strong></td>
<td>SEP00083031F49B</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Directory Number Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Voice Mail Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>BLF Presence Group</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>User Hold MOH Audio Source</strong></td>
<td>Standard Presence group</td>
</tr>
<tr>
<td><strong>Network Hold MOH Audio Source</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Auto Answer</strong></td>
<td>Auto Answer Off</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7975 SIP Configuration (Contd.)

### Directory URIs

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

- No Answer Ring Duration (seconds): < None >
- Call Pickup Group: < None >
Cisco IP Phone 7975 SIP Configuration (Contd.)

Set Display (Caller ID) = Cisco 7975-Phone. This is used in this example.
Set ASCII Display (Caller ID) = Cisco 7975-Phone. This is used in this example.
Set Line Text Label = Cisco 7975-Phone. This is used in this example.
Set External Phone Number Mask = 7322162754. This is used in this example.

<table>
<thead>
<tr>
<th>Park Monitoring</th>
<th>Voice Dial</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring</td>
<td>□ or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No</td>
<td>□ or</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Retrieve</td>
<td>□ or</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Destination</td>
<td>□ or</td>
<td></td>
<td></td>
</tr>
<tr>
<td>External</td>
<td>□ or</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>□ or</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forward No</td>
<td>□ or</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Retrieve</td>
<td>□ or</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Destination</td>
<td>□ or</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Internal</td>
<td>□ or</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Set Park Monitoring Reversion Timer service parameter A blank value will use value set in Park Monitoring.

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

<table>
<thead>
<tr>
<th>Line Settings for All Devices</th>
<th>Hold Reversion Ring Duration (seconds)</th>
<th>Setting the Hold Reversion Ring Duration to zero will disable the feature</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Hold Reversion Notification Interval (seconds)</td>
<td>Setting the Hold Reversion Notification Interval to zero will disable the feature</td>
</tr>
<tr>
<td></td>
<td>Party Entrance Tone</td>
<td>Set Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Line 1 on Device SEP00083031F49B</th>
<th>Display (Caller ID)</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></td>
<td>ASCII Display (Caller ID)</td>
<td>Cisco 7975-Phone</td>
</tr>
<tr>
<td></td>
<td>Line Text Label</td>
<td>Cisco 7975-Phone</td>
</tr>
<tr>
<td></td>
<td>External Phone Number Mask</td>
<td>7322162754</td>
</tr>
<tr>
<td></td>
<td>Visual Message Waiting Indicator Policy</td>
<td>Use System Policy</td>
</tr>
<tr>
<td></td>
<td>Audible Message Waiting Indicator Policy</td>
<td>Default</td>
</tr>
<tr>
<td></td>
<td>Ring Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>
**Cisco IP Phone 7975 SIP Configuration (Continued...)**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ring Setting</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>Call Pickup Group Audio Alert Setting (Phone Active)</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>
</tbody>
</table>

- **Multiple Call/Call Waiting Settings on Device SEP00083031F498**

  Note: The range to select the Max Number of calls is: 1-50

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Number of Cells*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Busy Trigger*</td>
<td></td>
<td>(Less than or equal to Max. Cells)</td>
</tr>
<tr>
<td></td>
<td>4</td>
<td></td>
</tr>
</tbody>
</table>

- **Forwarded Call Information Display on Device SEP00083031F498**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
<tr>
<td>Caller Number</td>
<td></td>
</tr>
<tr>
<td>Redirected Number</td>
<td></td>
</tr>
<tr>
<td>Dialed Number</td>
<td>✔</td>
</tr>
</tbody>
</table>

- **Users Associated with Line**

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

- **Save** | **Delete** | **Reset** | **Apply Config** | **Add New**

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

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Note: Testing was conducted in tekVizion labs
Cisco IP Phone 9971 SIP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = SIP_Chen_9971. 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 Soft key Template = Standard User. 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 (Contd.)

Check “Owner = Anonymous (Public/Shared Space)” This is used in this example.
Note: Testing was conducted in tekVizion labs
Cisco IP Phone 9971 SIP Configuration (Contd.)

**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**

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

---

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

<table>
<thead>
<tr>
<th>Product Specific Configuration Layout</th>
</tr>
</thead>
<tbody>
<tr>
<td>Parameter Value</td>
</tr>
<tr>
<td>-----------------</td>
</tr>
<tr>
<td>Disable Speakerphone</td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
</tr>
<tr>
<td>PC Port</td>
</tr>
<tr>
<td>Back USB Port</td>
</tr>
<tr>
<td>Side USB Port</td>
</tr>
<tr>
<td>Cisco Camera</td>
</tr>
<tr>
<td>Console Access</td>
</tr>
<tr>
<td>Video Capabilities</td>
</tr>
<tr>
<td>Enable/Disable USB Classes</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>SDIO</td>
</tr>
<tr>
<td>Bluetooth</td>
</tr>
<tr>
<td>WiFi</td>
</tr>
<tr>
<td>Bluetooth Profiles</td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td>Settings Access</td>
</tr>
<tr>
<td>Gratuitous ARP</td>
</tr>
</tbody>
</table>

| PC Voice VLAN Access | Enabled |
| Web Access | Disabled |
| Show All Calls on Primary Line | Disabled |
| Day/Display Not Active | Sunday |
| | Monday |
| | Tuesday |
| Display On Time | 07:30 |
| Display On Duration | 10:30 |
| Display Idle Timeout | 01:00 |
| HTTPS Server | http and https Enabled |
| Enable Power Save Plus | Sunday |
| | Monday |
| | Tuesday |
| Phone On Time | 06:00 |
| Phone Off Time | 24:00 |
| Phone Off Idle Timeout | 60 |
| □ Enable Audible Alert |
Cisco IP Phone 9971 SIP Configuration (Contd.)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<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>Scan 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 Tone 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>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Log Profile</td>
<td>Default</td>
</tr>
<tr>
<td>Advertise G.722 and iSAC Codes</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td></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></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Advert ID</td>
<td></td>
</tr>
</tbody>
</table>
## Cisco IP Phone 9971 SIP Configuration (Contd.)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Setting</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>LLDP Power Priority</td>
<td>Unknown</td>
<td></td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td>User Controlled</td>
<td></td>
</tr>
<tr>
<td>RIPS Mode</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td>Normal</td>
<td></td>
</tr>
<tr>
<td>Switch Port Remote Configuration</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Port Remote Configuration</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Power Negotiation</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Restrict Data Rates</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>SSH Access</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Incoming Call Toast Timer</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Provide Dial Tone From Release Button</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Hide Video By Default</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Background Image</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Simplified New Call Util</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Enable VX C VPN for MAC</td>
<td></td>
<td></td>
</tr>
<tr>
<td>VX C VPN Option</td>
<td>Dual Tunnel</td>
<td></td>
</tr>
<tr>
<td>VX C Challenge</td>
<td>Challenge</td>
<td></td>
</tr>
<tr>
<td>VX C M Servers</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Revert to All Calls</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>RTCP for Video</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Record Call Log from Shared Line</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Show Remote Private Calls</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Record Call Log For Remote Private Calls</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Show Call History for Selected Line Only</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Actionable Incoming Call Alert</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>DF bit</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Default Line Filter</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Separate Audio and Video Mute</td>
<td>Disabled</td>
<td></td>
</tr>
</tbody>
</table>

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

- Indicates required item.

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

***Note: Security Profile Contains Additional CAPF Settings.

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

*****Note: A custom Softkey template without supplementary service Softkeys must be used for a Hosted Desktop Device.
Cisco IP Phone 9971 SIP Configuration (Contd.)

Set Directory Number* = 2753. This is used in this example.
Set Description = Cisco 9971-Phone. 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.
Cisco IP Phone 9971 SIP Configuration (Contd.)

### Enterprise Alternate Number

<table>
<thead>
<tr>
<th>Add Enterprise Alternate Number</th>
</tr>
</thead>
</table>

### +E.164 Alternate Number

<table>
<thead>
<tr>
<th>Add +E.164 Alternate Number</th>
</tr>
</thead>
</table>

### Directory URIs

<table>
<thead>
<tr>
<th>Primary</th>
<th>URI</th>
<th>Partition</th>
<th>Advertise Globally via LDAP</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

<table>
<thead>
<tr>
<th>Advertised Failover Number</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
</table>

### AAR Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<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>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- Calling Search Space Activation Policy
  - Forward All
    - Internal
    - External
  - Secondary Calling Search Space for Forward All
    - Internal
    - External
  - Forward Busy
    - Internal
    - External
  - Forward No Answer
    - Internal
    - External
  - Forward Coverage
    - Internal
    - External
- Forward on CTI Failure
- Forward Unregistered
  - Internal
  - External
- No Answer Ring Duration (seconds)
- Call Pickup Group
  - < None >
Cisco IP Phone 9971 SIP Configuration (Contd.)

Set Display (Caller ID) = Cisco 9971-Phone. This is used in this example.
Set ASCII Display (Caller ID) = Cisco 9971-Phone. This is used in this example.
Set Line Text Label = Cisco 9971-Phone. This is used in this example.
Set External Phone Number Mask = 7322162753. This is used in this example.
Cisco IP Phone 9971 SIP Configuration (Contd.)

<table>
<thead>
<tr>
<th>Audible Message Waiting Indicator Policy</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ring Setting (Phone Idle)*</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)*</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</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 SEPC07BB8CA1BB46

- **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 SEPC07BB8CA1BB46

- **✓** Caller Name
- **☐** Caller Number
- **☐** Redirected Number
- **✓** Dialed Number

### Users Associated with Line

- **Associate End Users**

---

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

![SIP Trunk Security Profile Configuration](image-url)

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Name</strong></td>
<td>ATT Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>Non Secure SIP Trunk Profile authenticated by null String</td>
</tr>
<tr>
<td><strong>Device Security Mode</strong></td>
<td>Non Secure</td>
</tr>
<tr>
<td><strong>Incoming Transport Type</strong></td>
<td>TCP+UDP</td>
</tr>
<tr>
<td><strong>Outgoing Transport Type</strong></td>
<td>UDP</td>
</tr>
</tbody>
</table>

**Navigation: System → Security → SIP Trunk Security Profile**
SIP Profile Configuration used by SIP trunk to Cisco UBE

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

Set SIP profile Name *= Standard SIP Profile for ATT. 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](image-url)
### SDP Information

- SDP Session-level Bandwidth Modifier for Early Offer and Re-invites: TIA8 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>32768</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 Redirections</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>

---

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

- **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**

**Trunk Specific Configuration**

- **Reroute Incoming Request to new Trunk based on**
  - Never
- **RSVP Over SIP**
  - Local RSVP
- **Resource Priority Namespace List**
  - < None >
- **Fall back to local RSVP**
- **SIP Rel1XX 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 TLS Learned Destination Route String**

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

**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
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. This is used for this example
Set Media Resource Group List = MRGL_MTP.
SIP Trunk to Cisco UBE Configuration (Contd.)

Set Significant Digits\(^*\) = 4. This is used in this example.
SIP Trunk to Cisco UBE Configuration (Contd.)

Set Destination Address = Set IP address of ASR-Cisco UBE.
Set SIP Trunk Security Profile* = ATT Non Secure Sip Trunk Profile.
Set SIP Profile* = Standard SIP Profile for ATT. This is used in this example.

Note:
Testing was conducted in tekVizion labs
SIP Trunk to Cisco UBE Configuration (Contd.)

<table>
<thead>
<tr>
<th>Normalization Script</th>
</tr>
</thead>
<tbody>
<tr>
<td>Normalization Script: &lt; None &gt;</td>
</tr>
<tr>
<td>Enable Trace</td>
</tr>
<tr>
<td>Parameter Name</td>
</tr>
<tr>
<td>1</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Recording Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>None</td>
</tr>
<tr>
<td>This trunk connects to a recording-enabled gateway</td>
</tr>
<tr>
<td>This trunk connects to other clusters with recording-enabled gateways</td>
</tr>
</tbody>
</table>

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

[Table with columns for parameter name and parameter value, followed by options for recording information and geolocation configuration.]
SIP Trunk to Fax Gateway Configuration

**Navigation:** Device → Trunk

Set **Device Name** = Trunk_SIP_FAX_Gateway. This is used for this example
Set **Description** = SIP Trunk to FAX Gateway. This is used for this example
Set **Device Pool** = G729. This is used for this example
Set **Media Resource Group List** = MRGL_MTP.

![Image of Cisco Unified CM Administration interface]

**Device Information**

- **SIP Trunk**
- **Description** = SIP Trunk to FAX Gateway
- **Device Pool** = G729
- **Media Resource Group List** = MRGL_MTP
SIP Trunk to Fax Gateway Configuration (Contd.)

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

**Intercompany Media Engine (IME)**

- E.164 Transformation Profile: < None >

**MLPP and Confidential Access Level Information**

- MLPP Domain: < None >
- Confidential Access Mode: < None >
- Confidential Access Level: < None >

**Call Routing Information**

- Remote-Party-Id
- Asserted-Identity
  - Asserted-Type*: Default
  - SIP Privacy*: Default

**Inbound Calls**

- Significant Digits*: All
- Connected Line ID Presentation*: Default
- Connected Name Presentation*: Default
- Calling Search Space: < None >
- AAR Calling Search Space: < None >
- Prefix DN
  - Redirecting Diversion Header Delivery - Inbound
SIP Trunk to Fax Gateway Configuration (Contd.)

### 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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Note: Testing was conducted in tekVizion labs
SIP Trunk to Fax Gateway Configuration (Contd.)

Set Destination Address = Set IP address of Fax Gateway.
Set SIP Trunk Security Profile* = ATT Non Secure Sip Trunk Profile.
Set SIP Profile* = Standard SIP Profile for ATT. This is used in this example.

<table>
<thead>
<tr>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.80.22.7</td>
<td></td>
<td>5060</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>MTP Preferred Originating Codes</th>
<th>BLF Presence Group</th>
<th>SIP Trunk Security Profile</th>
<th>Remote Calling Search Space</th>
<th>Out-Of-Dialog Refer Calling Search Space</th>
<th>SUBSCRIBE Calling Search Space</th>
<th>SIP Profile</th>
<th>DTMF Signaling Method</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>ATT Non Secure Sip Trunk Profile</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td>Standard SIP for ATT</td>
</tr>
</tbody>
</table>

**Normalization Script**

Normalization Script: < None >

**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 >

Save  Delete  Reset  Add New
### Route Pattern Configuration

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

---

**Find and List Route Patterns**

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>PXX</td>
<td>Network Based call forwarding</td>
<td></td>
<td></td>
<td>ATT SIP TRUNK</td>
<td></td>
</tr>
<tr>
<td>01191XXXXXXX</td>
<td>International Calling</td>
<td></td>
<td></td>
<td>ATT SIP TRUNK</td>
<td></td>
</tr>
<tr>
<td>7323680461</td>
<td></td>
<td></td>
<td></td>
<td>Trunk SIP Fax Gateway</td>
<td></td>
</tr>
<tr>
<td>9-92</td>
<td></td>
<td></td>
<td></td>
<td>ATT SIP TRUNK</td>
<td></td>
</tr>
<tr>
<td>9-22XX</td>
<td></td>
<td></td>
<td></td>
<td>ATT SIP TRUNK</td>
<td></td>
</tr>
</tbody>
</table>

---

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

---

Note: Testing was conducted in tekVizion labs
Route Pattern Configuration (Contd.)

Set Route Pattern* = 9. @ This is used to route to AT&T via ASR Cisco UBE.
Set Description = To PSTN via ATT SIP Trunk. This text is used to identify this Route Pattern.
Set Numbering Plan* = NANP. This text is used to identify the Numbering Plan.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
Set Calling Party Transform mask = 732216XXX. This is used for this example.
All other values are default.
Route Pattern Configuration (Contd.)

Set Discard Digits = PreDot.

![Route Pattern Configuration Table]

- **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
  - Service Parameter Name: < Not Exist >
  - Service Parameter Value

* - indicates required item.
Route Pattern Configuration (Contd.)

Set Route Pattern* = *X! This is used to route to AT&T via ASR 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.
Set Calling Party Transform mask = 732216XXX. This is used for this example.
Route Pattern Configuration (Contd.)

Set Route Pattern* = 7323680461. 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 (Contd.)

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

#### Called Party Transformations
- **Discard Digits**: 
- **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

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

[Image of the form layout]

---

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Note: Testing was conducted in tekVizion labs
Jabber Client Configuration

**Navigation:** Device → Phone

Select Phone Type* = Cisco Unified Client Services Framework
Set Device Name* = CSFUser1. This is used in this example.
Set Description = CSFUser1. This is used in this example.
Select Device Pool = G729. This is used in this example.
Select Phone Button Template* = Standard Client Services Framework.
Jabber Client Configuration (Contd.)

Media Resource Group List = MRGL_MTP
Check Owner = User. This is used in this example.
Set Owner user ID* = jabber1. This is used for this example
Jabber Client Configuration (Contd.)

<table>
<thead>
<tr>
<th>Number Presentation Transformation</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Caller ID For Calls From This Phone</strong></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)</td>
</tr>
<tr>
<td><strong>Remote Number</strong></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Protocol Specific Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Node</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
</tr>
<tr>
<td>BLF Presence Group</td>
</tr>
<tr>
<td>SIP Dial Rules</td>
</tr>
<tr>
<td>NTP Preferred Originating Codec</td>
</tr>
<tr>
<td>Device Security Profile</td>
</tr>
<tr>
<td>Recouting Calling Search Space</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
</tr>
<tr>
<td>SIP Profile</td>
</tr>
<tr>
<td>Digest User</td>
</tr>
</tbody>
</table>

- 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 7 12 12 (YYYY:NM:DD:HH)</td>
</tr>
<tr>
<td>Certificate Operation Status</td>
<td>None</td>
</tr>
<tr>
<td>Note: Security Profile Contains Additional CAPF Settings.</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

<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.)

#### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Video Calling</td>
<td>Enabled</td>
<td></td>
</tr>
</tbody>
</table>

#### Interactive Connectivity Establishment (ICE)

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<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>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>File Types to Block in File Transfer</td>
<td></td>
<td></td>
</tr>
<tr>
<td>URLs to Block in File Transfer</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Jabber Client Configuration (Contd.)

<table>
<thead>
<tr>
<th>Desktop Client Settings</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Automatically Start in Phone Control*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatically Control Tethered Desk Phone*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Extend and Connect Capability*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Display Contact Photos*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Number Lookups on Directory*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Jabber For Windows Software Update Server URL</td>
<td><a href="mailto:user1@lab.televisor.com">user1@lab.televisor.com</a></td>
</tr>
<tr>
<td>Problem Report Server URL</td>
<td></td>
</tr>
<tr>
<td>Analytics Collection*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Analytics Server URL</td>
<td></td>
</tr>
<tr>
<td>Cisco Support Field</td>
<td></td>
</tr>
</tbody>
</table>

Note:
- Testing was conducted in tekVizion labs.

* indicates required item.
** Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
*** Note: Security Profile Contains Addition CAPF Settings.
**** Note: A Protected device means it is capable of playing Secure and Non-Secure Tones. When the checkbox is checked, the user will hear a Secure or Non-Secure Tone when the call is connected.
***** Note: A custom Softkey template without supplementary service Softkeys must be used for a Hot line Device.
Voicemail Port Configuration

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

---

**Cisco Unified CM Administration**

**Find and List Voice Mail Ports**

![Image of Cisco Unified CM Administration interface]

**Status:**
2 records found

**Voice Mail Port (1 - 2 of 2)**

<table>
<thead>
<tr>
<th>Device Name</th>
<th>Description</th>
<th>Device Pool</th>
<th>Device Security Mode</th>
<th>Calling Search Space</th>
<th>Extension</th>
<th>Partition</th>
<th>Status</th>
<th>IP-4 Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>CiscoUM1-V11</td>
<td>VM Port</td>
<td>G729</td>
<td>Non</td>
<td>Secure Voice Mail Port</td>
<td>2501</td>
<td>Registered with dus24pubsub.lab.tekvizion.com</td>
<td>10.80.14.4</td>
<td></td>
</tr>
<tr>
<td>CiscoUM1-V12</td>
<td>VM Port</td>
<td>G729</td>
<td>Non</td>
<td>Secure Voice Mail Port</td>
<td>2502</td>
<td>Registered with dus24pubsub.lab.tekvizion.com</td>
<td>10.80.14.4</td>
<td></td>
</tr>
</tbody>
</table>
Voicemail Port Configuration (Contd.)

Set Port Name = CiscoUM1-VI1. This is used for this example.
Set Description = VM Port. This is used for this example.
Set Device Pool = G729
Set Directory Number* = 2501. This is used in this example.

### Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Port Name</td>
<td>CiscoUM1-VI1</td>
</tr>
<tr>
<td>Description</td>
<td>VM Port</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>G729</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
</tr>
<tr>
<td>Device Security Node*</td>
<td>Non Secure Voice Mail Port</td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Directory Number Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<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>
<tr>
<td>External Number Mask</td>
<td></td>
</tr>
</tbody>
</table>

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EDCS# xxx Rev #
Page 118 of 152
Note: Testing was conducted in tekVizion labs
Message Waiting Numbers Configurations

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

Set Message Waiting Number* = 2511
Set Message Waiting Indicator* = On
Set Message Waiting Number* = 2512
Set Message Waiting Indicator* = Off
Voicemail Pilot Configuration

Navigation: Advanced Features → Voice Mail → Voice Mail Pilot

Set Voice Mail Pilot Number = 2302. This is used for this example
Set Description = Voicemail. This is used for this example.
FAX Gateway Configuration

FAX-GATEWAY2# show version

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(3)M1, RELEASE SOFTWARE (fc1)

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

Copyright (c) 1986-2014 by Cisco Systems, Inc.

Compiled Sat 25-Oct-14 03:34 by prod_rel_team

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

FAX-GATEWAY2 uptime is 1 week, 6 days, 19 hours, 19 minutes
System returned to ROM by power-on
System image file is "flash0:c2900-universalk9-mz.SPA.154-3.M1.bin"
Last reload type: Normal Reload
Last reload reason: power-on

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If you require further assistance please contact us by sending email to
export@cisco.com.

Cisco CISCO2901/K9 (revision 1.0) with 483328K/40960K bytes of memory.
Processing board ID FTX174081SJ
2 Gigabit Ethernet interfaces
1 terminal line
2 Voice FXS interfaces
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  CISCO2901/K9  FTX174081SJ
Technology Package License Information for Module: 'c2900'

<table>
<thead>
<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Technology-package Current</th>
<th>Type</th>
<th>Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipbase</td>
<td>ipbasek9</td>
<td>Permanent</td>
<td>ipbasek9</td>
<td></td>
</tr>
<tr>
<td>security</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>uc</td>
<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
<td></td>
</tr>
<tr>
<td>data</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>NtwkEss</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>CollabPro</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td></td>
</tr>
</tbody>
</table>

Configuration register is 0x2102

FAX-GATEWAY2# sh running-config

Building configuration...

Current configuration : 8536 bytes

! Last configuration change at 11:58:36 UTC Wed Jul 1 2015 by cisco

! version 15.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname FAX-GATEWAY2
!
boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
no logging console
no logging monitor
enable secret 4 iR3uUX3Bo6oYbT6ajhFwJe39FR4g.1QCmm7yYduKGZI
!
no aaa new-model
!
!
!
!
!
!
!
!
ip domain name lab.tekvizion.com
ip name-server 10.64.1.3
ip cef
no ipv6 cef
multilink bundle-name authenticated

stcapp feature access-code

stcapp feature speed-dial

cts logging verbose

Note: Testing was conducted in tekVizion labs
crypto pki trustpoint TP-self-signed-2189441908

enrollment selfsigned

subject-name cn=IOS-Self-Signed-Certificate-2189441908

revocation-check none

dskeypair TP-self-signed-2189441908

!

crypto pki certificate chain TP-self-signed-2189441908

certificate self-signed 01

3082022B 30820194 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
31312F30 2D060355 04031326 494F532D 53656C66 2D535E967 6E65642D 43657724
696666AE 6174652D 32331389 34343433 3038301E 170D3133 31303031 32303234
303225A1 0D323030 31303030 305A3031 322F302D 06035504 03132649
4F532D53 656C662D 53656C66 2D535E967 6E65642D 43657724 66696361 7465623D 31383934
34331390 3830819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
810092C7 19826C36 792DA64E 8FB4D8BC 1DDD4D7A 0882107F B14FCB24 699A35A9
D521C888 5B434F4FC D394E945 81A1380A 2E753478 93190ADE 75AA8971 883E9214
C607CCDF 6FCCDE9C E95CE01A AEE4FCBE 3E9143C D11C638F FC3E4ED2 57569523
70A87D7C EFAD6688 C6244C79 5B955591 BF75EE61 DC4D0ADE 8D897AE2 CE76A938
983F0203 010001A3 5330513C 0F06O355 1D130101 F040530 010501FF 301F0603
551D2304 1B301679 14279B59 09E3EB37 0AE0DCE0 F8075BB6 DF93858A 45301D06
03551D0E 04160414 279B5909 E3EB370A E0DCE0F8 075BB6DF 93858A45 300D0609
2A864886 F70D0101 05050003 8181006E CF10B11F 9D8B59A9 AEACDEB8 26649CBB
0F6C9690 12EAB70 4BF5703D 98D2665A CD1B27D2 9B29351D 3ADF0B97 3C415F9A
0DDD8F88 66CE4689 2D089FE8 EF3FFE54 5C85608C EE45908F D1160BDA A9185D58
D3DA8795 428A7CE7 B95222FC 60796800 485EDA2F B6C86F7A DF66B562 74942705
quit

voice-card 0
dsp services dspfarm
!
!
!

voice service voip

no ip address trusted authenticate
address-hiding
mode border-element
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

midcall-signaling passthru
g729 annexb-all
!

voice class codec 1

codec preference 1 g729r8
codec preference 2 g711ulaw
!
!

voice class sip-profiles 1

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

!
!
!
!
!
!
!
!
!
!
!
!
!
!
license udi pid CISCO2901/K9 sn FTX174081SJ

hw-module pvdm 0/0

!
!
!
!
!
!
!
!
!
!
!
!
!
!
username cisco privilege 15 secret 4 tnhtc92DxBeIjYk8LWJrPV3652i4ntXrpb4RFmfqY

!

redundancy

!
!
!
!
!
!
!
!
!
!
!
!
!
!
interface Embedded-Service-Engine0/0

no ip address

shutdown

!
interface GigabitEthernet0/0
ip address 10.80.22.7 255.255.255.0
duplex auto
speed auto
!
interface GigabitEthernet0/1
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
ip http server
ip http authentication local
ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
!
control-plane

!

!

voice-port 0/0/0
no vad
shutdown
!

voice-port 0/0/1
no echo-cancel enable
no vad
cptone IN
station-id name fax test
station-id number 7323680461
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
! 
! 
ccm-manager music-on-hold 
! 
no ccm-manager fax protocol cisco 
! 
dial-peer voice 110 pots 
  service session 
  destination-pattern 7323680461 
  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 7323680461 
  voice-class codec 1 
  voice-class sip profiles 1 
  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 201 voip
description Gateway to CUCM
service session
destination-pattern 9T
session protocol sipv2
session target ipv4:10.80.14.2
session transport udp
voice-class codec 1
voice-class sip profiles 1
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
!
!
!
!

gatekeeper
shutdown
!
!
banner exec ^C
% Password expiration warning.

--=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-

Cisco Configuration Professional (Cisco CP) is installed on this device and it provides the default username "cisco" for one-time use. If you have already used the username "cisco" to login to the router and your IOS image supports the "one-time" user option, then this username has already expired. You will not be able to login to the router with this username after you exit this session.

It is strongly suggested that you create a new username with a privilege level of 15 using the following command.

username <myuser> privilege 15 secret 0 <mypassword>

Replace <myuser> and <mypassword> with the username and password you want to use.

--=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-

^C
banner login ^C

--=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-=-

Cisco Configuration Professional (Cisco CP) is installed on this device. This feature requires the one-time use of the username "cisco" with the password "cisco". These default credentials have a privilege level of 15.
YOU MUST USE CISCO CP or the CISCO IOS CLI TO CHANGE THESE PUBLICLY-KNOWN CREDENTIALS

Here are the Cisco IOS commands.

username <myuser> privilege 15 secret 0 <mypassword>
no username cisco

Replace <myuser> and <mypassword> with the username and password you want to use.

IF YOU DO NOT CHANGE THE PUBLICLY-KNOWN CREDENTIALS, YOU WILL NOT BE ABLE TO LOG INTO THE DEVICE AGAIN AFTER YOU HAVE LOGGED OFF.

For more information about Cisco CP please follow the instructions in the QUICK START GUIDE for your router or go to http://www.cisco.com/go/ciscocp

^C
!
line con 0
login local
line aux 0
line 2
no activation-character
no exec
transport preferred none
transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
stopbits 1
line vty 0 4
exec-timeout 0 0
login local
transport input telnet ssh
line vty 5 15
login local
transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)

CUC Version

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For information about Cisco Unified Communications Manager please visit our Unified Communications System Documentation web site.

For Cisco Technical Support please visit our Technical Support web site.
CUC Telephony Integration with Cisco UCM

Navigation: Telephony Integrations → Phone system

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

**Navigation:** Telephony Integration → Port Group

![Cisco Unity Connection Administration](image)

### Search Port Groups

<table>
<thead>
<tr>
<th>Port Group</th>
<th>Phone System Display</th>
<th>Port Count</th>
<th>Integration Method</th>
<th>Needs Reset</th>
</tr>
</thead>
<tbody>
<tr>
<td>CUCM-1</td>
<td>Default</td>
<td>2</td>
<td>SCCP (Skinny)</td>
<td>No</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted in tekVizion labs
CUC Port Group (Contd.)

Set Display Name* = CUCM-1. This is used in this example.
Check Enable Message waiting indicators.
Set MWI On Extension = 2511. This is used in this example.
Set MWI Off Extension= 2512. This is used in this example.
CUC Port Settings

**Navigation:** Telephony Integration → Port

![Cisco Unity Connection Administration](image)

*Note:* Testing was conducted in tekVizion labs
CUC Sample User Basic Settings

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

Set Alias = 0461. This is one of the extension used for this testing.
Set Extension = 0461. This is used for this example.
CUC Sample User Basic Settings (Contd.)

Set Partition = clus24-unity Partition. This is used for this example.
Select Search Space = clus24-unity Search Space.
Select Phone System = Default.
Auto Attendant

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

Set Display Name = Demo Auto Attendant. This is used for this example.
Set Phone System = Default
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.
Set Search Space = clus24-unity Search Space.
Auto Attendant (Contd.)

- **Cisco Unity Connection**
  - Users
  - Class of Service
  - Templates
  - Contacts
  - Distribution Lists
  - Call Management
    - System Call Handlers
    - Directory Handlers
    - Interview Handlers
    - Custom Recordings
    - Call Routing
  - Message Storage
  - Networking
  - Unified Messaging
  - Video
  - Dial Plan
  - System Settings
  - Telephony Integrations
  - Tools

- **Call Handler**
  - **Display Name**: Demo Auto Attendant
  - **Creation Time**: 2015-04-24 04:47:15.1 249
  - **Phone System**: Default
  - **Active Schedule**: All Hours
  - **Use System Default Time Zone**: Checked
  - **Time Zone**: (GMT-06:00) America/Chicago
  - **Language**: Use System Default Language
  - **Extension**: 2999
  - **Partition**: clus24-unity Partition

- **Search Scope**
  - **Search Space**: clus24-unity Search Space
  - **Inherit Search Space From Call**: Unchecked

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

CUP/IMP Version

System version: 10.5.2.10000-9
VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 80Gbytes, 2048Mbytes RAM

User administrator last logged in to this cluster on Wednesday, July 1, 2015 2:21:40 AM CDT, to node 10.80.14.3, from 172.16.29.152 using HTTPS

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

Navigation: System → Presence Topology
Node Configuration

Navigation: System → Presence Topology → Fully Qualified Domain Name

![Node Configuration Screenshot]
**Users**

**Navigation:** System → Presence Topology → clus24imp-pub.lab.tekvizion.com → Users

![User Assignment](image)

### Node User Assignment (clus24imp-pub.lab.tekvizion.com)

<table>
<thead>
<tr>
<th>User ID</th>
<th>First Name</th>
<th>Last Name</th>
<th>IM Address</th>
<th>Directory URL</th>
<th>Failed Over</th>
<th>Node</th>
<th>Presence Redundancy Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>jabber1</td>
<td>cisco</td>
<td><a href="mailto:jabber1@lab.tekvizion.com">jabber1@lab.tekvizion.com</a></td>
<td><a href="mailto:jabber1@lab.tekvizion.com">jabber1@lab.tekvizion.com</a></td>
<td>clus24imp-pub.lab.tekvizion.com</td>
<td></td>
<td></td>
<td>DefaultCUPS_subcluster</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted in tekVizion labs
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>ASR</td>
<td>Aggregation 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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