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 ISR G2 Router with SIP Interface
MAR 2015
# Table of Contents

Introduction ........................................................................................................................................ 5

Network Topology .......................................................................................................................... 6

Hardware Components .................................................................................................................. 7

Software Requirements .................................................................................................................. 7

Features .......................................................................................................................................... 8

- Features – Supported .................................................................................................................. 8
- Network Based Features - Supported......................................................................................... 8
- Features - Not Supported ............................................................................................................ 8

Caveats ............................................................................................................................................ 9

- Fax .............................................................................................................................................. 9
- Auto-Attendant ............................................................................................................................. 9
- Hold/Resume & Music on Hold (MOH) ....................................................................................... 9
- Ringback Tone on Early Unattended Transfer ........................................................................... 9
- PBX Based Call Forward Unconditional ..................................................................................... 9
- SIP Provisional Acknowledgement/Early media ........................................................................ 9
- AT&T IP Teleconferencing (IPTC) ............................................................................................. 10

Configuration Considerations ....................................................................................................... 11

Emergency 911/E911 Services Limitations and Restrictions .......................................................... 11

ISR Configuration .......................................................................................................................... 12

Cisco UCM Configuration .............................................................................................................. 33

- Cisco UCM Version .................................................................................................................... 34
- Cisco UCM Audio Codec Preference List .................................................................................... 34
- Cisco UCM Region Configuration ............................................................................................... 35
- Device Pool Configuration ......................................................................................................... 36
- Annunciator Configuration ......................................................................................................... 40
- Conference Bridge Configuration ............................................................................................... 41
- Media Termination Point Configuration ..................................................................................... 42
- Music on Hold Server Configuration .......................................................................................... 43

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 2 of 160
Note: Testing was conducted in tekVizion labs
Music on Hold Service (IP Voice Media Streaming App) Parameter Settings ........................................ 44
Music on Hold Service (Duplex Streaming) Parameter Settings .......................................................... 45
Media Resource Group Configuration .................................................................................................. 46
Media Resource Group List Configuration ............................................................................................. 47
UC Service Configuration ...................................................................................................................... 48
Service Profile Configuration .................................................................................................................. 51
End User Configuration .......................................................................................................................... 54
Cisco IP Phone 7965 SCCP Configuration ............................................................................................ 59
Cisco IP Phone 9971 SIP Configuration ................................................................................................ 75
SIP Trunk Security Profile Configuration used by SIP trunk to Cisco UBE .............................................. 93
SIP Profile Configuration used by SIP trunk to Cisco UBE .................................................................. 94
SIP Trunk to Cisco UBE Configuration .................................................................................................. 98
Route Pattern Configuration .................................................................................................................... 109
Jabber Client Configuration ..................................................................................................................... 117
Voicemail Port Configuration ................................................................................................................ 124
Message Waiting Numbers Configurations ............................................................................................ 126
Voicemail Pilot Configuration ................................................................................................................ 127
FAX Gateway Configuration .................................................................................................................... 128
Cisco UCM SCCP Integration with Cisco Unity Connection (CUC) ....................................................... 143
CUC Version ........................................................................................................................................ 143
CUC Telephony Integration with Cisco UCM .......................................................................................... 144
CUC Port Group ..................................................................................................................................... 145
CUC Port Settings .................................................................................................................................. 147
CUC Sample User Basic Settings ........................................................................................................... 148
Auto Attendant ....................................................................................................................................... 150
Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP) ...................................... 153
CUP/IMP Version ..................................................................................................................................... 153
Presence Topology .................................................................................................................................. 154
Node Configuration .................................................................................................................................. 155
Users ......................................................................................................................................................... 156
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 Integrated Services Routers (ISR) Version 15.4(3) M1 with connectivity to AT&T’s IP Flex-Reach SIP trunk service. It also covers support and configuration example of Cisco Unity Connection (CUC) messaging integrated with Cisco Unified Communications Manager (Cisco UCM). The deployment model covered in this application note is Cisco Integrated Services Routers (ISR) to PSTN (AT&T IP Flexible Reach SIP). AT&T IP Flexible Reach provides inbound and outbound call service.

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

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

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

Note: Testing was conducted in tekVizion labs.
Hardware Components

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

Software Requirements

- Cisco 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: V-9.1.3 Build 13181
- VentaFax client version: 7.3.233.582 I

Note: Testing was conducted in tekVizion labs
Features

Features – Supported

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

Network Based Features - Supported

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

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

Features - Not Supported

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

Fax

- The maximum fax rate achieved using G711u (G3 or SG3) is only 14400 kbps.
- G711 Passthrough 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.

Ringback Tone on Early Unattended Transfer

- Caller does not hear ringback tone when a call is transferred to PSTN user.

PBX Based Call Forward Unconditional

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

SIP Provisional Acknowledgement/Early media

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

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

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

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

Emergency 911/E911 Services Limitations and Restrictions

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

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 11 of 160
Note: Testing was conducted in tekVizion labs
ISR Configuration

topcube#sh 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)

topcube uptime is 3 days, 8 hours, 0 minutes

System returned to ROM by reload at 23:08:42 UTC Thu Mar 12 2015

System image file is "flash0:c2900-universalk9-mz.SPA.154-3.M1.bin"

Last reload type: Normal Reload

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.
A summary of U.S. laws governing Cisco cryptographic products may be found at:

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

Cisco CISCO2921/K9 (revision 1.0) with 483328K/40960K bytes of memory.
Processor board ID FTX1746AJCC
3 Gigabit Ethernet interfaces
1 terminal line
1 Virtual Private Network (VPN) Module
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
250880K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:

License UDI:

----------------------------------------
Device#  PID         SN
----------------------------------------
*1   CISCO2921/K9   FTX1746AJCC
Technology Package License Information for Module: 'c2900'

---------------------------------------------------------------
<table>
<thead>
<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Technology-package</th>
<th>Technology-package</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Current</td>
<td>Type</td>
<td>Next reboot</td>
</tr>
</tbody>
</table>
---------------------------------------------------------------
| ipbase     | ipbasek9           | Permanent          | ipbasek9           |
| security   | securityk9         | Permanent          | securityk9         |
| uc         | uck9               | Permanent          | uck9               |
| data       | None               | None               | None               |
| NtwkEss    | None               | None               | None               |
| CollabPro  | None               | None               | None               |

Configuration register is 0x2102

topcube# sh running-configuration

Building configuration...

Current configuration: 10923 bytes

! Last configuration change at 07:56:59 UTC Fri Mar 13 2015 by cisco

! version 15.4
service tcp-keepalives-in
service timestamps debug datatime msec
service timestamps log datatime msec
no service password-encryption
!
hostname topcube
!
boot-start-marker
boot-end-marker
!
aqm-register-fnf
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
enable secret 4 Pe0NhiWw5iXZpE.k5VhTSCoGpcuVeRyrer9kEPz20Z6
!
no aaa new-model
!
!
!
!
!
!
!
!
!
!
no ip domain lookup

ip cef

no ipv6 cef

multilink bundle-name authenticated

cts logging verbose

voice-card 0
dspfarm
dsp services dspfarm
voice service voip
no ip address trusted authenticate
address-hiding 1
mode border-element 2
allow-connections sip to sip 3
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
header-passing
error-passthru 4
asserted-id pai 5
no update-callerid
everly-offer forced 6
midcall-signaling passthru 7
privacy-policy passthru 8
g729 annexb-all

1 Hide signaling and media peer addresses from endpoints other than gateway.
2 If the mode border-element command is not entered, border-element-related commands are not available for Cisco Unified Border Element voice connections on the Cisco 2900 and Cisco 3900 series platforms with a universal feature set. The mode border-element command is not available on any other platforms.
3 This command enables Cisco UBE basic IP-to-IP voice communication feature.
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.
voice class codec 1
codec preference 1 g729r8 bytes 30
codec preference 2 g711ulaw

voice class codec 2
codec preference 1 g711ulaw
codec preference 2 g729r8 bytes 30

voice class codec 3
codec preference 1 g711ulaw

voice class sip-profiles 1
response ANY sip-header Allow-Header modify "UPDATE," ""
request INVITE sip-header Diversion modify "</sip:(.*)@(.*)>" "</sip:732320\1@\2>"  
request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"  
response ANY sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30"
request INVITE sdp-header Audio-Attribute add "a=ptime:30"  

---

9 This command configures the codec preference to be assigned to dial-peers. Alternatively, single code can be configured into individual dial-peers.

10 This SIP profile expands the Diversion header number from a 4-digit extension to a full 10-digit DID number in order to obtain interoperability with AT&T's served users during call-forward scenarios. The six digits in "sip: 732216" are variable and must be replaced with the first 6 digits of the DID's provisioned for the customer site.

11 Cisco 6900-series IP phones use ptme value of 20 ms. AT&T networks prefer ptme value of 30 ms. This SIP profile modifies SDP ptme 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 ptme value should be applied. This is because G711 RTP was not defaulting to 20ms.
voice translation-rule 1
rule 1 /^.*\(40..\)/ /7323201/

voice translation-rule 2
rule 2 /^\+\(1\)\(7........\)/ /2/

voice translation-profile NPA
translate calling 1

voice translation-profile test+1
translate called 2

license udi pid CISCO2921/K9 sn FTX1746AJCC
hw-module pvdm 0/0

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.

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.
username cisco privilege 15 password 0 cisco

redundancy

! interface Embedded-Service-Engine0/0
no ip address
shutdown

interface GigabitEthernet0/0

\[14\] WAN interface to AT&T
ip address 192.65.79.58 255.255.255.224
duplex auto
speed auto
!
interface GigabitEthernet0/1
ip address 10.80.22.10 255.255.255.0
duplex auto
speed auto
!
interface GigabitEthernet0/2
no ip address
shutdown
duplex auto
speed auto
!
ip forward-protocol nd
!
no ip http server
no ip http secure-server
!
ip route 0.0.0.0 0.0.0.0 192.65.79.33
ip route 10.0.0.0 255.0.0.0 10.80.22.1
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1

---

15 LAN interface to Cisco UCM
16 Cisco UBE LAN interface IPv4 Address
control-plane

dial-peer voice 100 voip

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

destination-pattern 73236.....

no modem passthrough

17 Dial peer for AT&T facing network
session protocol sipv2 18
session target ipv4:207.242.225.210
voice-class codec 1 19
voice-class sip asymmetric payload full 20
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru 21
voice-class sip profiles 1 22
voice-class sip bind control source-interface GigabitEthernet0/0 23
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte 24
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none 25
no vad
!
dial-peer voice 200 voip
description "Outgoing To AT&T .IP PBX facing side"
no modem passthrough

18 Session protocol SIPv2 is used for this testing
19 Assigns voice class codec 1 settings to dial-peer (codec support and filtering).
20 Configures the dynamic SIP asymmetric payload support.
21 This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. In this example, the command is set at the dial-peer level, you can also set the command at a global level to affect all dial-peers without necessarily setting the command on each dial-peer.
22 This command enables the dial peer to use SIP profile 1
23 Configure the Cisco UBE SIP messaging to use the HSRP virtual address in SIP messaging. Once HSRP is configured under the physical interface and the bind command is issued, calls to the physical IP address will fail. This is because the SIP listening socket is now bound to the virtual IP address but the signaling packets use the physical IP address, and therefore cannot be handled.
24 This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call.
25 This command enables T38 fax protocol for calls terminating on this dial-peer
session protocol sipv2
incoming called-number [27]T
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nre
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 300 voip
description " Int'l calls to AT&T - AT&T facing side "
destination-pattern 011T
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
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 400 voip
description " Int'l calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number 011T
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
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 500 voip
description " N11 Calls to AT&T - AT&T facing side "

Note: Testing was conducted in tekVizion labs
destination-pattern .11
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 600 voip
description " N11 Calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number .11
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
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 800 voip
description " Incoming AT&T to IP-PBX. AT&T facing side "
translation-profile incoming test+1
huntstop
no modem passthrough
session protocol sipv2
incoming called-number +1[37][13][24]32040..
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
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 214 voip
description "Outgoing To AT&T"-AT&T facing side
destination-pattern [2-9]T
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dttmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 122 voip
description "OPERATOR TESTING"
destination-pattern 0
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
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 141 voip
description "Network Feature"
translation-profile outgoing NPA
destination-pattern *..
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 2151 voip
description "Outgoing To AT&T"-AT&T facing side
destination-pattern 7323204292
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0
voice-class sip bind media source-interface GigabitEthernet0/0
dtmf-relay rtp-nte
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
dial-peer voice 700 voip
description "Incoming AT&T to IP-PBX - IP-PBX facing side"
huntstop
destination-pattern [37][13][24]........
no modem passthrough
session protocol sipv2
session target ipv4:10.80.22.2:5060
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/1
voice-class sip bind media source-interface GigabitEthernet0/1
dtmf-relay rtp-nte
fax-relay sg3-to-g3
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
no remote-party-id
retry invite 2
timers expires 1800000
connection-reuse
protocol mode ipv4
!
!
!
gatekeeper
shutdown
!
!
!
line con 0
logging synchronous
line aux 0
line 2
session-timeout 90
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

session-timeout 90

exec-timeout 960 0

logging synchronous

login local

transport input all

!

scheduler allocate 20000 1000

!

end

---

**Cisco UCM Configuration**

The configuration screen shots shows general over view of lab configuration for this interoperability testing.
Cisco UCM Audio Codec Preference List

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

Cisco UCM 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

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 35 of 160
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 (continued...)
### Device Mobility Related Information

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

### Geolocation Configuration

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

### Call Routing Information

#### Incoming Calling Party Settings

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

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

<table>
<thead>
<tr>
<th>Phone Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID For Calls From This Phone</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Connected Party Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Party Transformation CSS</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Redirecting Party Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Redirecting Party Transformation CSS</td>
</tr>
</tbody>
</table>
Annunciator Configuration

**Navigation:** Media Resource → Annunciator

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

**Navigation:** Media Resources → Conference Bridge

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

![Conference Bridge Configuration](image_url)
Media Termination Point Configuration

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

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

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

Set Music on Hold Server Name* = MOH_2.
Set Description = MOH_clus32pubsub. This is used for this example.
Set Device Pool* = G729_pool.
Music on Hold Service (IP Voice Media Streaming App) Parameter Settings

**Navigation:** System → Service Parameter

Note: Make sure codecs G.729 Annex A and G.711 mulaw are configured in parameter Supported MOH Codecs.
Select Server* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.
Select Service* = Cisco IP Voice Media Streaming App (Active).

Music on Hold Service (Duplex Streaming) Parameter Settings

**Navigation:** System → Service Parameter

Select Server* = clus32pubsub--CUCM Voice/Video (Active). This is used in this example.
Select Service* = Cisco CallManager (Active).
Select Duplex Streaming Enabled * = True

Media Resource Group Configuration

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

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

Media Resource Group List Configuration

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

Set Name = MRGL_MTP.
Set selected Media Resource Groups = MRG_MTP.

**UC Service Configuration**

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


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

Select UC Service Type: = CTI
Set Name* = CTI_SRV. This is used in this example.
Set Description = CTI for Jabber Clients. This is used in this example.
Set Host Name/IP Address* = 10.80.22.2 (Cisco UCM Address)

UC Service Configuration (Contd...)

Select UC Service Type: = IM and Presence
Set Name* = IMP_SRV. This is used in this example.
Set Description = IM Presence. This is used in this example.
Set Host Name/IP Address* = 10.80.22.3  (Cisco UCM IM & Presence IP Address)

Service Profile Configuration

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

Set Name* = Jabber_SVC_Profile. This is used in this example.
Set Description = Jabber Service Profile. This is used in this example.
Check - Make this the default service profile for the system.

Service Profile Configuration (Contd...)

Note: Testing was conducted in tekVizion labs
End User Configuration

**Navigation:** User Management ➔ End User

Set User ID* = jabber1 – This is used in this example.
Set Password = Password for profile.
Set Directory URI = jabber1@lab.tekvizion.com.
End User Configuration (continued...)
End User Configuration (continued...)

- **Service Settings**
  - 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: SEPC07BBCA1B811
  - Available Profiles
  - CTI Controlled Device Profiles

- **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

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

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

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

**CAPF Information**
- Associated CAPF Profiles

**Permissions Information**
- **Groups**
  - Standard Audit Users
  - Standard CAR Admin Users
  - Standard CCM Admin Users
  - Standard CCM End Users
  - Standard CCM Gateway Administration
- **Roles**
  - Standard AXL API Access
  - Standard Admin Rep Tool Admin
  - Standard Audit Log Administration
  - Standard CCM Admin Users
  - Standard CCM End Users

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 58 of 160
Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7965 SCCP Configuration

Set MAC Address* = the below mac is used in this example.
Set Description = Cisco7965_Phone. this text is used to identify this Phone.
Set Device Pool* = G729 pool. This is used in this example.
Set Phone Button Template* = Standard 7965 SCCP. This is used in this example.
Set Softkey Template = Standard User. This is used in this example.
Note: Testing was conducted in tekVizion labs

Cisco IP Phone 7965 SCCP Configuration (Continued...)

Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource.
Check Owner = Anonymous (Public/Shared Space). This is used in this example.
### Cisco IP Phone 7965 SCCP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Service Provisioning</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Phone Load Name</td>
<td></td>
</tr>
<tr>
<td>Single Button Range</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay</td>
<td>Default</td>
</tr>
<tr>
<td>Point*</td>
<td></td>
</tr>
<tr>
<td>BLF Audible Alert</td>
<td>Default</td>
</tr>
<tr>
<td>Setting (Phone Idle)</td>
<td></td>
</tr>
<tr>
<td>BLF Audible Alert</td>
<td>Default</td>
</tr>
<tr>
<td>Setting (Phone Busy)</td>
<td></td>
</tr>
<tr>
<td>Always Use Prime Line</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice</td>
<td>Default</td>
</tr>
<tr>
<td>Message*</td>
<td></td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Device Mobility Mode</td>
<td></td>
</tr>
<tr>
<td>Owner User ID</td>
<td>Default</td>
</tr>
<tr>
<td>Phone Personalization</td>
<td></td>
</tr>
<tr>
<td>Device Mobility Settings</td>
<td></td>
</tr>
<tr>
<td>Location</td>
<td></td>
</tr>
<tr>
<td>AAA Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Built In Bridge*</td>
<td></td>
</tr>
<tr>
<td>Privacy</td>
<td>Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRQI_MTP</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
### External Data Locations Information (Leave blank to use default)

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

### Extension Information

- **Enable Extension Mobility**
- **Log Out Profile** -- Use Current Device Settings --
- **Log in Time**  &lt; None &gt;
- **Log out Time**  &lt; None &gt;

### MLPP and Confidential Access Level Information

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

### Do Not Disturb

- **Do Not Disturb**
- **DND Option** &lt; None &gt;
- **DND Incoming Call Alert** &lt; None &gt;

### Secure Shell Information

- **Secure Shell User**
- **Secure Shell Password**

---

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

[Diagram of Product Specific Configuration Layout]

[Diagram of Additional Configuration Options]

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

Table:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Recording Tone</td>
<td>Disabled</td>
</tr>
<tr>
<td>Recording Tone Local</td>
<td>100</td>
</tr>
<tr>
<td>Recording Tone Remote</td>
<td>50</td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When</td>
<td>Disabled</td>
</tr>
<tr>
<td>Incoming Call RTCP</td>
<td>Disabled</td>
</tr>
<tr>
<td>Feature</td>
<td>Setting</td>
</tr>
<tr>
<td>-------------------------------------</td>
<td>-----------</td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>5</td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codec</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP):</td>
<td>Enabled</td>
</tr>
<tr>
<td>Switch Port</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP):</td>
<td>Enabled</td>
</tr>
<tr>
<td>PC Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol</td>
<td>Enabled</td>
</tr>
<tr>
<td>Protocol - Media Endpoint</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Enabled</td>
<td></td>
<td>Unknown</td>
<td>Disabled</td>
<td></td>
<td></td>
<td>User Controlled</td>
<td>Normal</td>
<td>0-Silent</td>
</tr>
</tbody>
</table>
### Cisco IP Phone 7965 SCCP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Headset</th>
<th>Default</th>
<th>Sidetone Level</th>
<th>Default</th>
<th>Headset Send Gain</th>
<th>Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>HTTPS Server</td>
<td>http and https Enabled</td>
<td>Headset/Headset Monitor</td>
<td>Enabled</td>
<td>Headset Recording</td>
<td>Disabled</td>
</tr>
<tr>
<td>Enbloc Dialing</td>
<td>Enabled</td>
<td>Switch Port Remote Configuration</td>
<td>Disabled</td>
<td>PC Port Remote Configuration</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td>Disabled</td>
<td>SSR Access</td>
<td>Disabled</td>
<td>LOGIN Access</td>
<td>Enabled</td>
</tr>
<tr>
<td>FIPS Mode</td>
<td>Disabled</td>
<td>80-bit SRTCP</td>
<td>Disabled</td>
<td>Customer Support Use</td>
<td></td>
</tr>
</tbody>
</table>
Cisco IP Phone 7965 SCCP Configuration (Continued...)

Set Directory Number* = 4085. This is used in this example.
Set Description = 7323204085. This is used in this example.
Set Alerting Name = Cisco 7965 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 7965 Phone. This is used in this example.
Cisco IP Phone 7965 SCCP Configuration (Continued...)
**Note:** Testing was conducted in tekVizion labs

---

### Cisco IP Phone 7965 SCCP Configuration (Continued...)

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

### Park Monitoring

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Park No Retrieve Destination</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Park External</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Park Reversion Timer</td>
<td>Reversion Timer service parameter</td>
</tr>
</tbody>
</table>

A blank value means to call the parker’s line.

### MLPP Alternate Party And Confidential Access Level Settings

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

### Line Settings for All Devices

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

---

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 71 of 160
Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7965 SCCP Configuration (Continued...)
### Line 1 on Device SEP44ADD9D56F39

<table>
<thead>
<tr>
<th>Display (Caller ID)</th>
<th>Cisco 7965 Phone</th>
<th>Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASCII Display (Caller ID)</td>
<td>Cisco 7965 Phone</td>
<td></td>
</tr>
<tr>
<td>Line Text Label</td>
<td>Cisco 7965 Phone</td>
<td></td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td>7323204085</td>
<td></td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy*</td>
<td>Use System Policy</td>
<td></td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)*</td>
<td>Use System Default</td>
<td>Applies to this line when any line on the phone has a call in progress.</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Call Pickup Group Audio Alert Setting (Phone Active)</th>
<th>Use System Default</th>
</tr>
</thead>
<tbody>
<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>
</tbody>
</table>

**Log Missed Calls**

---

Cisco IP Phone 7965 SCCP Configuration (Continued...)
### Multiple Call/Call Waiting Settings on Device SEP44ADD9D56F39

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

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

*Less than or equal to Max. Calls*

### Forwarded Call Information Display on Device SEP44ADD9D56F39

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

### Users Associated with Line

- [ ] Associate End Users

---

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 = Cisco 9971 Phone. this text is used to identify this Phone.
Set Device Pool*= G729 Pool. This is used in this example.
Set Phone Button Template*= Standard 9971 SIP. This is used in this example.
Set Media Resource Group List = MRGL_MTP. This is used in this example.
Set User Hold MOH Audio Source = 1-SampleAudioSource.
Set Network Hold MOH Audio Source = 1-SampleAudioSource
Cisco IP Phone 9971 SIP Configuration (Continued...)

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 76 of 160
Note: Testing was conducted in tekVizion labs
<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Built In Bridge</td>
<td>Default</td>
</tr>
<tr>
<td>Privacy</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Mode</td>
<td>Default</td>
</tr>
<tr>
<td>Owner</td>
<td>User</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>jaber1</td>
</tr>
<tr>
<td>Phone Personalization</td>
<td>Default</td>
</tr>
<tr>
<td>Services Provisioning</td>
<td>Default</td>
</tr>
<tr>
<td>Phone Load Name</td>
<td></td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Idle)</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Busy)</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

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

**Number Presentation Transformation**

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

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

---

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

<table>
<thead>
<tr>
<th><strong>Packet Capture Mode</strong></th>
<th>None</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Packet Capture Duration</strong></td>
<td>0</td>
</tr>
<tr>
<td><strong>BLF Presence Group</strong></td>
<td>Standard Presence group</td>
</tr>
<tr>
<td><strong>SIP Dial Rules</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>MTP Preferred Originating Codec</strong></td>
<td>711ulaw</td>
</tr>
<tr>
<td><strong>Device Security Profile</strong></td>
<td>Cisco 9971 - Standard SIP Non-Secure Profile</td>
</tr>
<tr>
<td><strong>Rerouting Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>SUBSCRIBE Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>SIP Profile</strong></td>
<td>Standard SIP Profile w/Early Media Disabled</td>
</tr>
<tr>
<td><strong>Digest User</strong></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- [ ] Media Termination Point Required
- [ ] Unattended Port
- [ ] Require DTMF Reception

### Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th><strong>Certificate Operation</strong></th>
<th>No Pending Operation</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Authentication Mode</strong></td>
<td>By Null String</td>
</tr>
<tr>
<td><strong>Authentication String</strong></td>
<td>Generate String</td>
</tr>
<tr>
<td><strong>Key Size (Bits)</strong></td>
<td>2048</td>
</tr>
<tr>
<td><strong>Operation Completes By</strong></td>
<td>2015 3 27 12 (YYYY:MM:DD:HH)</td>
</tr>
</tbody>
</table>

**Certificate Operation Status:** None

Note: Security Profile Contains Addition CAPF Settings.

### Expansion Module Information

| **Module 1** | < None > |
| **Module 1 Load Name** | |
| **Module 2** | < None > |
| **Module 2 Load Name** | |
| **Module 3** | < None > |
| **Module 3 Load Name** | |
Cisco IP Phone 9971 SIP Configuration(Continued...)
### External Data Locations Information (Leave blank to use default)

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

### 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>MLPP Indication*</td>
<td>Default</td>
</tr>
<tr>
<td>MLPP Preemption*</td>
<td>Default</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Do Not Disturb

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

---

Note: Testing was conducted in tekVizion labs
### Secure Shell Information

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

### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>PC Port *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Back USB Port *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Side USB Port *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Camera *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Console Access *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Video Capabilities *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Enable/Disable USB Classes</td>
<td>Mass Storage</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Human Interface Device</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Audio Class</td>
<td></td>
</tr>
<tr>
<td>SDIO *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Bluetooth *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Wifi *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Bluetooth Profiles *</td>
<td>Handsfree</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Human Interface Device</td>
<td></td>
</tr>
<tr>
<td>Settings Access *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access *</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Web Access *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Show All Calls on Primary Line *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Sunday</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Monday</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Tuesday</td>
<td></td>
</tr>
</tbody>
</table>
## Cisco IP Phone 9971 SIP Configuration (Continued...)

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

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Tone*</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>Enabled</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>
<tr>
<td>Log Profile</td>
<td></td>
</tr>
</tbody>
</table>

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

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

© 2015 Cisco Systems, Inc. All rights reserved.

Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

EDCS# xxx Rev #

Page 84 of 160

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

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<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 UI</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Enable VXC VPN for MAC</td>
<td></td>
<td></td>
</tr>
<tr>
<td>VXC VPN Option *</td>
<td>Dual Tunnel</td>
<td></td>
</tr>
<tr>
<td>VXC Challenge *</td>
<td>Challenge</td>
<td></td>
</tr>
<tr>
<td>VXC-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</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Incoming Call Alert *</td>
<td></td>
<td></td>
</tr>
<tr>
<td>DF bit *</td>
<td>0</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>
<tr>
<td>Softkey Control *</td>
<td>Feature Control Policy</td>
<td></td>
</tr>
<tr>
<td>Start Video Port</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Stop Video Port</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Lowest Alerting Line State Priority *</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>TLS Resumption Timer *</td>
<td>3600</td>
<td></td>
</tr>
<tr>
<td>Audio EQ *</td>
<td>Default : Default</td>
<td></td>
</tr>
</tbody>
</table>

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 85 of 160
Note: Testing was conducted in tekVizion labs
Cisco IP Phone 9971 SIP Configuration (Continued...)

Set Directory Number* = 4084. This is used in this example.
Set Description = 7323204084. This is used in this example.
Set Alerting Name = Cisco 9971 Phone. This is used in this example.
Set ASCII Alerting Name = Cisco 9971 Phone. This is used in this example.
Cisco IP Phone 9971 SIP Configuration (Continued...)
### Directory URIs

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

[Add Row]

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

Advertised Failover Number: < None >

### AAR Settings

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

### Cell Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space Activation Policy</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Forward All</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy External</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
## Cisco IP Phone 9971 SIP Configuration (Continued...)

### Park Monitoring

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

### MLPP Alternate Party And Confidential Access Level Settings

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

### Line Settings for All Devices

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

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

Set Display (caller ID) = Cisco9971-Phone. This is used in this example.
Set ASCII Display (caller ID) = Cisco9971-Phone. This is used in this example.
Set Line Text Label = Cisco9971-Phone. This is used in this example.
Set External Phone Number Mask = 7323204084. This is used in this example.
Cisco IP Phone 9971 SIP Configuration (Continued...)

<table>
<thead>
<tr>
<th>Display (Caller ID)</th>
<th>Cisco 9971-Phone</th>
<th>Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASCII Display (Caller ID)</td>
<td>Cisco 9971-Phone</td>
<td></td>
</tr>
<tr>
<td>Line Text Label</td>
<td>Cisco 9971-Phone</td>
<td></td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td>733204084</td>
<td></td>
</tr>
<tr>
<td>Visual Message Warning Indicator Policy*</td>
<td>Use System Policy</td>
<td></td>
</tr>
<tr>
<td>Audible Message Warning Indicator Policy*</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)*</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>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
<td></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>
<tr>
<td>Log Missed Calls</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Multiple Call/Call Waiting Settings on Device SEPC07BBCA18811

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 SEPC07BBCA18811

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

Users Associated with Line

<table>
<thead>
<tr>
<th>Full Name</th>
<th>User ID</th>
<th>Permission</th>
</tr>
</thead>
<tbody>
<tr>
<td>cisco</td>
<td>jabber1</td>
<td></td>
</tr>
</tbody>
</table>

Associate End Users  Select All  Clear All  Delete Selected

Save  Delete  Reset  Apply Config  Add New
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)

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name*</td>
<td>ATT Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Description</td>
<td>Non Secure SIP Trunk Profile authenticated by null String</td>
</tr>
<tr>
<td>Device Security Mode</td>
<td>Non Secure</td>
</tr>
<tr>
<td>Incoming Transport Type*</td>
<td>TCP+UDP</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
</tr>
<tr>
<td>Enable Digest Authentication</td>
<td></td>
</tr>
<tr>
<td>Nonce Validity Time (mins)*</td>
<td>600</td>
</tr>
<tr>
<td>X.509 Subject Name</td>
<td></td>
</tr>
<tr>
<td>Incoming Port*</td>
<td>5060</td>
</tr>
<tr>
<td>Enable Application level authorization</td>
<td></td>
</tr>
<tr>
<td>Accept presence subscription</td>
<td></td>
</tr>
<tr>
<td>Accept out-of-dialog referrers**</td>
<td></td>
</tr>
<tr>
<td>Accept unsolicited notification</td>
<td></td>
</tr>
<tr>
<td>Accept replaces header</td>
<td></td>
</tr>
<tr>
<td>Transmit security status</td>
<td></td>
</tr>
<tr>
<td>Allow charging header</td>
<td></td>
</tr>
<tr>
<td>SIP V.150 Outbound SDP Offer Filtering*</td>
<td>Use Default Filter</td>
</tr>
</tbody>
</table>
SIP Profile Configuration used by SIP trunk to Cisco UBE

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

Set SIP profile Name * = Standard SIP Profile w/Early Media Disabled. This is used for this example
Check Disable Early Media on 180
Set SIP Rel1xx Options* = Send PRACK if 1xx contains SDP

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

---

![Image of Cisco Unified CM Administration interface](https://example.com/cisco-unified-cm-administration-sip-profile)

- **Name** = Standard SIP Profile w/Early Media Disabled
- **Default MTP Telephony Event Payload Type** = 101
- **User-Agent and Server header information** = Send Unified CM Version Information as User-Agen
- **Version in User Agent and Server Header** = Major And Minor
- **Dial String Interpretation** = Phone number consists of characters 0-9, *, #, and +
- **Confidential Access Level Headers** = Disabled

**Redirect by Application**: Enable

- **Disable Early Media on 100**: Checked

---

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

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

<table>
<thead>
<tr>
<th>Parameters used in Phone</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)*</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)*</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)*</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)*</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE*</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>32765</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-cisco-serviceuri-ospickup</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>
</tbody>
</table>
```

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

- Conference Join Enabled
- RFC 2543 Hold
- Semi Attended Transfer
- Enable VAD
- 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

<table>
<thead>
<tr>
<th>Caller ID DN</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>
### Trunk Specific Configuration

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

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

Note: Testing was conducted in tekVizion labs
SIP Trunk to Cisco UBE Configuration

Navigation: Device → Trunk

Note: Testing was conducted in tekVizion labs
Set Device Name* = ATT_SIP_TRUNK. This is used for this example
Set Description = ATT SIP Trunk to PSTN. This is used for this example
Set Device Pool* = G729_pool. This is used for this example
Set Media Resource Group List = MRGL_MTP.

SIP Trunk to Cisco UBE Configuration (Continued...)

Note: Testing was conducted in tekVizion labs
Set Significant Digits\(^*\) = 4. This is used in this example.
SIP Trunk to Cisco UBE Configuration (Continued...)

### Incoming Calling Party Settings

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

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

### Incoming Called Party Settings

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

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

### Connected Party Settings

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

### Outbound Calls

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

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

<table>
<thead>
<tr>
<th>Calling Party Selection</th>
<th>Calling Line ID Presentation</th>
<th>Calling Name Presentation</th>
<th>Calling and Connected Party Info Format</th>
<th>Redirecting Diversion Header Delivery</th>
<th>Redirecting Party Transformation CSS</th>
<th>Use Device Pool Redirecting Party Transformation CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>△</td>
<td>Default</td>
<td>Default</td>
<td>△</td>
<td>Outbound</td>
<td>&lt; None &gt;</td>
<td>✔</td>
</tr>
</tbody>
</table>

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

Set Destination Address = Set IP address of ISR-Cisco UBE.
Set SIP Trunk Security Profile* = ATT_Non Secure Sip Trunk Profile.
Set SIP Profile* = ATT_SIP_Profile. This is used in this example.
### SIP Information

#### Destination
- **Destination Address**: 10.80.22.10
- **Destination Address IPv6**: 
- **Destination Port**: 5060

<table>
<thead>
<tr>
<th>MTP Preferred Originating Code*</th>
<th>711ulaw</th>
</tr>
</thead>
<tbody>
<tr>
<td>BLF Presence Group*</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>SIP Trunk Security Profile*</td>
<td>ATT Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>Remoting Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Out-Of-Dialog Refer Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Profile*</td>
<td>Standard SIP Profile w/Early Media Disabled</td>
</tr>
<tr>
<td>DTMF Signaling Method*</td>
<td>No Preference</td>
</tr>
</tbody>
</table>

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

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

#### Recording Information
- **None**
- **This trunk connects to a recording-enabled gateway**
- **This trunk connects to other gateways with recording-enabled gateways**

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

---

**SIP Trunk to Fax Gateway Configuration.**

**Navigation**: Device → Trunk

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

Set **Description** = **Trunk_SIP_FAX_Gateway**. This is used for this example

Set **Device Pool*** = **G729 pool**. This is used for this example
Set Media Resource Group List = MRGL_MTP.

<table>
<thead>
<tr>
<th>Product</th>
<th>SIP Trunk</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None(Default)</td>
</tr>
<tr>
<td>Device Name*</td>
<td>Trunk_SIP_FAX_Gateway</td>
</tr>
<tr>
<td>Description</td>
<td>Trunk to SIP FAX Gateway</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>G729 Pool</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunnelled Protocol*</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant*</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding*</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode*</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>

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

Note: Testing was conducted in tekVizion labs
<table>
<thead>
<tr>
<th>Feature</th>
<th>Description</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 sRTP 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

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

### Call Routing Information

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>□ Remote-Party-Id</td>
<td></td>
</tr>
<tr>
<td>□ Asserted-Identity</td>
<td></td>
</tr>
<tr>
<td>Asserted-Type*</td>
<td>Default</td>
</tr>
<tr>
<td>SIP Privacy*</td>
<td>Default</td>
</tr>
</tbody>
</table>

### Inbound Calls

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits*</td>
<td>All</td>
</tr>
<tr>
<td>Connected Line ID Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td></td>
</tr>
<tr>
<td>□ Redirecting Diversion Header Delivery - Inbound</td>
<td></td>
</tr>
</tbody>
</table>
SIP Trunk to Fax Gateway Configuration (Continued...)

--- Incoming Calling Party Settings ---

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

<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

---
SIP Trunk to Fax Gateway Configuration (Continued...)

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

<table>
<thead>
<tr>
<th><strong>Caller Information</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID DN</td>
</tr>
<tr>
<td>Caller Name</td>
</tr>
<tr>
<td>Maintain Original Caller ID DN and Caller Name in Identity Headers</td>
</tr>
</tbody>
</table>
### SIP Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Destination Address</td>
<td>1.0.0.22.7</td>
</tr>
<tr>
<td>Destination Port</td>
<td>5060</td>
</tr>
</tbody>
</table>

- **MTP Preferred Originating Codec**: 711ulaw
- **BLP Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: ATT Non Secure SIP Trunk Profile
- **Rerouting Calling Search Space**: < None >
- **Out-Of-Dialog Refer Calling Search Space**: < None >
- **SUBSCRIBE Calling Search Space**: < None >

#### SIP Profile
- **Standard SIP Profile w/Early Media Disabled**
- **DTHF Signalling Method**: No Preference

### Normalization Script

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

### Recording Information

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

### Geolocation Configuration

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

---

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

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

Set Route Pattern* = 9. @ This is used to route to AT&T via ISR Cisco UBE.
Set Description = To PSTN via ATT SIP Trunk. This text is used to identify this Route Pattern.
Set Gateway/Route List* = ATT_SIP_TRUNK. This is used for this example.
All other values are default.
Route Pattern Configuration (Continued...)

Pattern Definition

- **Route Pattern**: 9.4
- **Route Partition**: < None >
- **Description**: To PSTN via ATT SIP Trunk

Gateway/Route List: ATT_SIP_TRUNK

Calling Party Transformations

- **Use Calling Party's External Phone Number Mask**
- **Calling Party Transform Mask**
- **Prefix Digits (Outgoing Calls)**
- **Calling Line ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Calling Party Number Type**: Cisco CallManager
- **Calling Party Numbering Plan**: Cisco CallManager
Route Pattern Configuration (Continued...)

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

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

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

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

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

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

Note: Testing was conducted in tekVizion labs
Set Route Pattern* = 4084 this is used to route to Fax Client via Fax Gateway.
Set Description = To FAX. This text is used to identify this Route Pattern.
Set Gateway/Route List* = Trunk_SIP_FAX_Gateway. This is used for this example.
All other values are default.
Route Pattern Configuration (Continued...)

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

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

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Protocol</td>
</tr>
<tr>
<td>Carrier Identification Code</td>
</tr>
</tbody>
</table>

- Save  - Delete  - Copy  - Add New
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 Pool. This is used in this example.
Select Phone Button Template* = Standard Client Services Framework.
Jabber Client Configuration (Contd...)

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

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Built In Bridge*</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Mode*</td>
<td>Default</td>
</tr>
<tr>
<td>Owner</td>
<td>[Select]</td>
</tr>
<tr>
<td>Owner User ID*</td>
<td>jabber1</td>
</tr>
<tr>
<td>Mobility User ID</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Primary Phone</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line*</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Message*</td>
<td>Default</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Ignore Presentation Indicators (internal calls only)</td>
<td>[Select]</td>
</tr>
<tr>
<td>Allow Control of Device from CTI</td>
<td>[Select]</td>
</tr>
<tr>
<td>Logged Into Hunt Group</td>
<td>[Select]</td>
</tr>
<tr>
<td>Remote Device</td>
<td>[Select]</td>
</tr>
<tr>
<td>Require off-premise location</td>
<td>[Select]</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
### Number Presentation Transformation

**Caller ID For Calls From This Phone**

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

**Remote Number**

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

### Protocol Specific Information

**Packet Capture Mode**
- None

**Packet Capture Duration**
- 0

**BLF Presence Group**
- Standard Presence group

**SIP Dial Rules**
- < None >

**MTP Preferred Originating Codec**
- 711ulaw

**Device Security Profile**
- Cisco Unified Client Services Framework - Standard

**Rerouting Calling Search Space**
- < None >

**SUBSCRIBE Calling Search Space**
- < None >

**SIP Profile**
- Standard SIP Profile w/Early Media Disabled
  - View Details

**Digest User**
- < None >

- Media Termination Point Required
- Unattended Port
- Require DTMF Reception

---

**Jabber Client Configuration (Contd...)**
Certification Authority Proxy Function (CAPF) Information

- Certificate Operation: No Pending Operation
- Authentication Mode: By NULL String
- Authentication String: Generate String
- Key Size (Bits): 2048
- Certificate Operation None
- Status: Note: Security Profile Contains Additional CAPF Settings.

Extension Information

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

MLPP and Confidential Access Level Information

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

Do Not Disturb

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

Jabber Client Configuration (Contd...)

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 121 of 160
Note: Testing was conducted in tekVizion labs
Note: Testing was conducted in tekVizion labs
### Desktop Client Settings

<table>
<thead>
<tr>
<th>Setting</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.tekvizion.com">user1@lab.tekvizion.com</a></td>
</tr>
<tr>
<td>Problem Report Server URL</td>
<td></td>
</tr>
<tr>
<td>Analytics Collection*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>

**Analytics Server URL**
- [Blank]

**Cisco Support Field**
- [Blank]

**Buttons:**
- Save
- Delete
- Copy
- Reset
- Apply Config
- Add New
Voicemail Port Configuration

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

![Voicemail Port Configuration Screen](image-url)

<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>Status</th>
<th>IP Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>CiscoUM1-V11</td>
<td>VoiceMail</td>
<td>G729 Pool</td>
<td>Non Secure Voice Mail Port</td>
<td>2301</td>
<td>Registered with clus32pubsub</td>
<td>10.80.22.4</td>
<td></td>
</tr>
<tr>
<td>CiscoUM1-V12</td>
<td>VoiceMail</td>
<td>G729 Pool</td>
<td>Non Secure Voice Mail Port</td>
<td>2302</td>
<td>Registered with clus32pubsub</td>
<td>10.80.22.4</td>
<td></td>
</tr>
</tbody>
</table>

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

Set Port Name = CiscoUM1-VI1. This is used for this example.
Set Description = VoiceMail. This is used for this example.
Set Device Pool = G729 Pool
Set Directory Number* = 2301. This is used in this example.

<table>
<thead>
<tr>
<th>Device Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Registration:</strong> Registered with Cisco Unified Communications Manager clus32pubsub</td>
</tr>
<tr>
<td><strong>IPv4 Address:</strong> 10.80.22.4</td>
</tr>
<tr>
<td><strong>Device is trusted</strong></td>
</tr>
<tr>
<td><strong>Port Name</strong> = CiscoUM1-VI1</td>
</tr>
<tr>
<td><strong>Description</strong> = VoiceMail</td>
</tr>
<tr>
<td><strong>Device Pool</strong> = G729 Pool</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Directory Number Information</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Directory Number</strong> = 2301</td>
</tr>
<tr>
<td><strong>Partition</strong> = &lt; None &gt;</td>
</tr>
<tr>
<td><strong>Calling Search Space</strong> = &lt; None &gt;</td>
</tr>
<tr>
<td><strong>AAR Group</strong> = &lt; None &gt;</td>
</tr>
<tr>
<td><strong>Internal Caller ID Display</strong> = VoiceMail</td>
</tr>
<tr>
<td><strong>Internal Caller ID Display (ASCII format)</strong> = VoiceMail</td>
</tr>
<tr>
<td><strong>External Number Mask</strong></td>
</tr>
</tbody>
</table>
Message Waiting Numbers Configurations

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

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

![Image of Cisco Unified CM Administration interface](image_url)
Voicemail Pilot Configuration

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

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

FAX-GATEWAY2#sh 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, 16 hours, 39 minutes

System returned to ROM by reload at 14:38:17 UTC Tue Mar 10 2015

System image file is "flash0:c2900-universalk9-mz.SPA.154-3.M1.bin"

Last reload type: Normal Reload

Last reload reason: Reload Command

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you
agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.

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

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

Cisco CISCO2901/K9 (revision 1.0) with 483328K/40960K bytes of memory.
Processor 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
--------------------------------------------------------------------------
Technology Package License Information for Module: 'c2900'

<table>
<thead>
<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Technology-package</th>
</tr>
</thead>
<tbody>
<tr>
<td>Current</td>
<td>Type</td>
<td>Next reboot</td>
</tr>
</tbody>
</table>

| ipbase     | ipbasek9           | Permanent          | ipbasek9                       |
| security   | None               | None               | None                           |
| uc         | uck9               | Permanent          | uck9                           |
| data       | None               | None               | None                           |
| NtwkEss    | None               | None               | None                           |
| CollabPro  | None               | None               | None                           |

Configuration register is 0x2102

FAX-GATEWAY2#sh run

Building configuration...
Current configuration : 7131 bytes

! Last configuration change at 14:41:28 UTC Wed Mar 25 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.1QCmm7yYduKGZ!

! no aaa new-model

!
Note: Testing was conducted in tekVizion labs

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

crypto pki trustpoint TP-self-signed-2189441908

  enrollment selfsigned

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

revocation-check none

rsakeypair 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 2D536967 6E65642D 43657274
  69666966 6174652D 32313839 34343139 3038301E 170D3133 31303031 32303234
  30325A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 31383934
  34313930 3830819F 300D0609 2A864886 F70D0101 01050003 818D0030 81890281
  810092C7 1982BC36 792DA64E 8FB4DBBC 1DDD4D7A 0882107F B14FCB24 699A35A9
  D521C88A 5B43F4FC D394E945 81A1380A 2E753478 93190ADE 75AA8971 883E9214
  C607CCD9 6FFCDE9C E95CE01A AEE4FCBE 3E91A43C D11C638F FC3E4ED2 57569523
  70A8D7C6 EFAD6688 C6244C79 5B955391 BF75EE61 DC4D0ADE 8D897AE2 CE76A938
  983F0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF 301F0603
Note:

Testing was conducted in tekVizion labs.
codec preference 2 g729r8

voice class sip-profiles 1

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

license udi pid CISCO2901/K9 sn FTX174081SJ

hw-module pvdm 0/0
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
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

ip route 0.0.0.0 0.0.0.0 10.80.22.1
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 10.80.0.0 255.255.0.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
!
!
!
!
control-plane
!
!
voice-port 0/0/0
no vad
shutdown
!
voice-port 0/0/1
no echo-cancel enable
no vad
station-id name fax test
station-id number 7323204084
caller-id enable
!
!
!
!
!
!
!
!
!
!
!
!
!
no mgcp timer receive-rtcp
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
ccm-manager music-on-hold
!
no ccm-manager fax protocol cisco
!
dial-peer voice 110 pots
  service session
  destination-pattern 4084
  port 0/0/1
 !
dial-peer voice 200 voip
  description CUCM to Gateway
  service session
  session protocol sipv2
  session transport udp
  incoming called-number 4084
  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 [2-9]T
session protocol sipv2
session target ipv4:10.80.22.2
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
!
!
!
gatekeeper
shutdown
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

Copyright © 1999 - 2015 Cisco Systems, Inc.
All rights reserved.

This product contains cryptographic features and is subject to United States and local country laws governing import, export, transfer and use. Delivery of Cisco cryptographic products does not imply third-party authority to import, export, distribute or use encryption. Importers, exporters, distributors and users are responsible for compliance with U.S. and local country laws. By using this product you agree to comply with applicable laws and regulations. If you are unable to comply with U.S. and local laws, return this product immediately.
CUC Telephony Integration with Cisco UCM

**Navigation:** Telephony Integrations → Phone system

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

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

Set Display Name* = CUCM-1. This is used in this example.
Check Enable Message waiting indicators.
Set MWI on Extension = 2298. This is used in this example.
Set MWI off Extension= 299. This is used in this example.
CUC Port Settings

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

**Navigation:** Cisco Unity connection → Users → Users

Set Alias = 4084. This is one of the extension used for this testing.
Set Extension = 4084. This is used for this example.
CUC Sample User Basic Settings (Continued...)

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

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

Auto Attendant (Continued...)
<table>
<thead>
<tr>
<th><strong>Call Handler</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Display Name</strong></td>
</tr>
<tr>
<td><strong>Creation Time</strong></td>
</tr>
<tr>
<td><strong>Phone System</strong></td>
</tr>
<tr>
<td><strong>Active Schedule</strong></td>
</tr>
<tr>
<td><strong>Time Zone</strong></td>
</tr>
<tr>
<td><strong>Language</strong></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td></td>
</tr>
<tr>
<td><strong>Extension</strong></td>
</tr>
<tr>
<td><strong>Partition</strong></td>
</tr>
</tbody>
</table>

**Search Scope**
- Search Space: clus32unity Search Space
- Inherit Search Space from Call

© 2015 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
EDCS# xxx Rev #
Page 152 of 160
Note: Testing was conducted in tekVizion labs
Note:

Testing was conducted in tekVizion labs
Presence Topology

**Navigation**: System → Presence Topology
Node Configuration

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

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

![Node User Assignment](image)

<table>
<thead>
<tr>
<th>User ID</th>
<th>First Name</th>
<th>Last Name</th>
<th>IM Address</th>
<th>Directory URI</th>
<th>Node</th>
<th>Presence Redundancy Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>jabber1</td>
<td>cisco</td>
<td></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>clus32imp.lab.tekvizion.com</td>
<td>DefaultCUPSSubcluster</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 32 9.1.2. This is used for this example.
Presence Gateway *= clus32pubsub.lab.tekvizion.com
## Acronyms

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

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.
### Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

### European Headquarters
CiscoSystems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-
europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

### Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

### AsiaPacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

Cisco Systems has more than 200 offices in the following countries and regions. Addresses, phone numbers, and fax numbers are listed on the Cisco Web site at [http://www.cisco.com/go/offices](http://www.cisco.com/go/offices).

Argentina • Australia • Austria • Belgium • Brazil • Bulgaria • Canada • Chile • China PRC • Colombia • Costa Rica • Croatia • Czech Republic • Denmark • Dubai, UAE • Finland • France • Germany • Greece • Hong Kong SAR • Hungary • India • Indonesia • Ireland • Israel • Italy • Japan • Korea • Luxembourg • Malaysia • Mexico • The Netherlands • New Zealand • Norway • Peru • Philippines • Poland • Portugal • Puerto Rico • Romania • Russia • Saudi Arabia • Scotland • Singapore • Slovakia • Slovenia • South Africa • Spain • Sweden • Switzerland • Taiwan • Thailand • Turkey Ukraine • United Kingdom • United States • Venezuela • Vietnam • Zimbabwe

© 2015 Cisco Systems, Inc. All rights reserved.
CCENT, Cisco Lumin, Cisco Nexus, the Cisco logo and the Cisco Square Bridge logo are trademarks of Cisco Systems, Inc.; Changing the Way We Work, Live, Play, and Learn is a service mark of Cisco Systems, Inc.; and Access Registrar, Aironet, BPX, Catalyst, CCDA, CCDP, CCVP, CCIE, CCIP, CCNA, CCNP, CCSP, Cisco, the Cisco Certified Internetwork Expert logo, Cisco IOS, Cisco Press, Cisco Systems, Cisco Systems Capital, the Cisco Systems logo, Cisco Unity, EtherFast, EtherSwitch, Fast Step, Follow Me Browsing, FormShare, GigaDrive, HomeLink, Internet Quotient, IOS, iPhone, iQ Expertise, the iQ logo, iQ Net Readiness Scorecard, iQuick Study, LightStream, Linksys, Meeting Place, MGX, Networking Academy, Network Registrar, Packet, PIX, ProConnect, ScriptShare, SMARTnet, StackWise, The Fastest Way to Increase Your Internet Quotient, and TransPath are registered trademarks of Cisco Systems, Inc. and/or its affiliates in the United States and certain other countries. All other trademarks mentioned in this document or Website are the property of their respective owners. The use of the word partner does not imply a partnership relationship between Cisco and any other company. (0705R)
Printed in the USA