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 4431 Router with SIP Interface APR 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
<table>
<thead>
<tr>
<th>Topic</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Music on Hold Service (IP Voice Media Streaming App) Parameter Settings</td>
<td>44</td>
</tr>
<tr>
<td>Music on Hold Service (Duplex Streaming) Parameter Settings</td>
<td>45</td>
</tr>
<tr>
<td>Media Resource Group Configuration</td>
<td>46</td>
</tr>
<tr>
<td>Media Resource Group List Configuration</td>
<td>47</td>
</tr>
<tr>
<td>UC Service Configuration</td>
<td>48</td>
</tr>
<tr>
<td>Service Profile Configuration</td>
<td>51</td>
</tr>
<tr>
<td>End User Configuration</td>
<td>54</td>
</tr>
<tr>
<td>Cisco IP Phone 7965 SCCP Configuration</td>
<td>59</td>
</tr>
<tr>
<td>Cisco IP Phone 9971 SIP Configuration</td>
<td>75</td>
</tr>
<tr>
<td>SIP Trunk Security Profile Configuration used by SIP trunk to Cisco UBE</td>
<td>93</td>
</tr>
<tr>
<td>SIP Profile Configuration used by SIP trunk to Cisco UBE</td>
<td>94</td>
</tr>
<tr>
<td>SIP Trunk to Cisco UBE Configuration</td>
<td>98</td>
</tr>
<tr>
<td>Route Pattern Configuration</td>
<td>109</td>
</tr>
<tr>
<td>Jabber Client Configuration</td>
<td>117</td>
</tr>
<tr>
<td>Voicemail Port Configuration</td>
<td>124</td>
</tr>
<tr>
<td>Message Waiting Numbers Configurations</td>
<td>126</td>
</tr>
<tr>
<td>Voicemail Pilot Configuration</td>
<td>127</td>
</tr>
<tr>
<td><strong>FAX Gateway Configuration</strong></td>
<td>128</td>
</tr>
<tr>
<td><strong>Cisco UCM SCCP Integration with Cisco Unity Connection (CUC)</strong></td>
<td>143</td>
</tr>
<tr>
<td>CUC Version</td>
<td>143</td>
</tr>
<tr>
<td>CUC Telephony Integration with Cisco UCM</td>
<td>144</td>
</tr>
<tr>
<td>CUC Port Group</td>
<td>145</td>
</tr>
<tr>
<td>CUC Port Settings</td>
<td>147</td>
</tr>
<tr>
<td>CUC Sample User Basic Settings</td>
<td>148</td>
</tr>
<tr>
<td>Auto Attendant</td>
<td>150</td>
</tr>
<tr>
<td><strong>Cisco UCM Integration with Cisco Unified CM IM and Presence (CUP/IMP)</strong></td>
<td>153</td>
</tr>
<tr>
<td>CUP/IMP Version</td>
<td>153</td>
</tr>
<tr>
<td>Presence Topology</td>
<td>154</td>
</tr>
<tr>
<td>Node Configuration</td>
<td>155</td>
</tr>
<tr>
<td>Users</td>
<td>156</td>
</tr>
</tbody>
</table>

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EDCS# xxx Rev #

Page 3 of 160

Note: Testing was conducted in tekVizion labs
Introduction

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

- This application note describes the necessary steps and configurations of Cisco Unified Communications Manager (Cisco UCM) 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) S2 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 ISR4431/K9 (1RU) processor with 1687596K/6147K bytes of memory.
- Processor board ID FTX1845AJ9S
  - 4 Gigabit Ethernet interfaces
  - 32768K bytes of non-volatile configuration memory.
  - 4194304K bytes of physical memory.
  - 7057407K bytes of flash memory at bootflash:

Software Requirements

- ISR: ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.4 (3)S2, RELEASE SOFTWARE (fc3).
- System image file is "bootflash:isr4400-universalk9.03.13.02.5.154-3.S2-ext.SPA.bin".
- Cisco Unity Connection version: System version: 10.5.2.10000-5
- Cisco Unified CM IM and Presence: System version: 10.5.2.10000-9
- Cisco Jabber client version: 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
- RTCP
- 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

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.
- G711Passsthrough test is achieved using “fax protocol pass-through g711ulaw”.
- Fax protocol T38 has been tested.

Auto-Attendant

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

Hold/Resume & Music on Hold (MOH)

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

Ring back Tone on Early Unattended Transfer

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

PBX Based Call Forward Unconditional

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

SIP Provisional Acknowledgement/Early media

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

Following scenarios were 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.

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EDCS# xxx Rev #
Page 11 of 160
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ISR Configuration

ATT_IPFR_4K#show version

Cisco IOS XE Software, Version 03.13.02.S - Extended Support Release
Cisco IOS Software, ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 15.4(3)S2, RELEASE SOFTWARE (fc3)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2015 by Cisco Systems, Inc.
Compiled Fri 30-Jan-15 15:19 by mcpre

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GPL code under the terms of GPL Version 2.0. For more details, see the
documentation or "License Notice" file accompanying the IOS-XE software,
or the applicable URL provided on the flyer accompanying the IOS-XE
software.

ROM: IOS-XE ROMMON

ATT_IPFR_4K uptime is 1 week, 2 days, 5 hours, 35 minutes
Uptime for this control processor is 1 week, 2 days, 5 hours, 37 minutes
System returned to ROM by reload
System image file is "bootflash:/isr4400-universalk9.03.13.02.S.154-3.S2-ext.SPA.bin"

Last reload reason: Reload Command

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

Technology Package License Information:

<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>

Note: Testing was conducted in tekVizion labs
cisco ISR4431/K9 (1RU) processor with 1687596K/6147K bytes of memory.

Processor board ID FTX1845AJ9S

4 Gigabit Ethernet interfaces

32768K bytes of non-volatile configuration memory.

4194304K bytes of physical memory.

7057407K bytes of flash memory at bootflash:

Configuration register is 0x2102

**ATT_IPFR_4K#** `sh running-configuration`

Building configuration...

Current configuration : 11300 bytes

! Last configuration change at 14:28:41 UTC Wed Apr 22 2015 by cisco

! version 15.4

service config

service timestamps debug datetime msec

service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname ATT_IPFR_4K
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
enable secret 5 $1$EWC0$6QpPfaUZ/L04rhx3qQlj7/
enable password tekV1z10n
!

aaa new-model
!
!
!
aaa session-id common

no ip domain lookup

ip domain name tekvizion.com
subscriber templating
multilink bundle-name authenticated

cts logging verbose

voice service voip

no ip address trusted authenticate
address-hiding
mode border-element
allow-connections sip to sip
redirect ip2ip
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip

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.
header-passing
error-passthr 4
asserted-id pai 5
no update-callerid
early-offer forced 6
midcall-signaling passthr 7
privacy-policy passthr 8
g729 annexb-all
!
voice class codec 1 9
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

---

4 This command allows SIP error messages to pass-through end-to-end without modification through Cisco UBE.
5 This command enables router to send P-Asserted ID within the SIP Message Header. Alternatively, this command can also be applied to individual dial-peers (voice-class sip asserted-id pai).
6 This command enables delay offer-to-early offer conversion of initial SIP INVITE message to calls matched to this dial-peer level.
7 This command must be enabled at a global level to maintain integrity of SIP signaling between AT&T network and Cisco Unified Communications Manager (Cisco UCM) across Cisco UBE.
8 This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. This command can either be set at a global level, such as in this example, or it can be set at the dial-peer level.
9 This command configures the codec preference to be assigned to dial-peers. Alternatively, single code can be configured into individual dial-peers.
voice class sip-profiles 1

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

request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "<sip:732320\1\2>" 10

request INVITE sdp-header Audio-Attribute modify "a=ptime:20" "a=ptime:30" 11

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

request INVITE sdp-header Audio-Attribute add "a=ptime:30" 12

voice translation-rule 1 13

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

voice translation-rule 2

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

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

11 Cisco 6900-series IP phones use ptime value of 20 ms. AT&T networks prefer ptime value of 30 ms. This SIP profile modifies SDP ptime value from 20 to 30 ms and it should be applied to dial-peers where G729 is the preferred codec. If the customer creates a dial-peer specifically for G711, a sip-profile without modifying the ptime value should be applied. This is because G711 RTP was not defaulting to 20ms.

12 This SIP profile is required in order to advertise the ptime=30 attribute in the outgoing SIP INVITE from Cisco UBE to AT&T. Currently RFC’s do not have a standard method to advertise ptime values for each offered codec within a SDP offering with multiple codecs. This SIP profile allows for Cisco UBE to include the ptime attribute with a value of 30ms.

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.
!voice translation-profile NPA
 translate calling 1
!
voice translation-profile test+1
 translate called 2
!
!
license udi pid ISR4431/K9 sn FOC18261KJL
license boot level appxk9
license boot level uck9
spanning-tree extend system-id
!
username cisco privilege 15 password 0 tekV1z10n
!
redundancy
 mode none
!
!
!
ip tftp source-interface GigabitEthernet0
ip ssh version 1
!
!
!
interface GigabitEthernet0/0/0
no ip address
shutdown
negotiation auto

interface GigabitEthernet0/0/1
no ip address
shutdown
negotiation auto

interface GigabitEthernet0/0/2 ¹⁴
description LAN interface
ip address 10.80.22.10 255.255.255.0 ¹⁵
negotiation auto

interface GigabitEthernet0/0/3 ¹⁶
description WAN interface
ip address 192.65.79.58 255.255.255.224
negotiation auto

¹⁴ WAN interface to AT&T
¹⁵ LAN interface to Cisco UCM
¹⁶ Cisco UBE LAN interface IPv4 Address
interface GigabitEthernet0
vrf forwarding Mgmt-intf
ip address 10.64.2.202 255.255.0.0
negotiation 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 172.16.0.0 255.255.0.0 10.80.22.1
ip route vrf Mgmt-intf 0.0.0.0 0.0.0.0 10.64.1.1
!
!
!
!
!
!
!
!
!
!
!
!
!
!
control-plane
!
!
!
!
!
!
!
!
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
!
!
!
dial-peer voice 200 voip
description "Outgoing To AT&T .IP PBX facing side"
no modem passthrough
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 GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
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
dial-peer voice 214 voip

description "Outgoing To AT&T"-AT&T facing side
destination-pattern [2-9]T
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n-te
no fax-relay sg3-to-g3
fax rate 14400

---

17 Dial peer for AT&T facing network
18 Session protocol SIPv2 is used for this testing
19 Assigns voice class codec 1 settings to dial-peer (codec support and filtering).
20 Configures the dynamic SIP asymmetric payload support.
21 This command allows for privacy settings to be transparently passed between AT&T network and Cisco UCM. In this example, the command is set at the dial-peer level, you can also set the command at a global level to affect all dial-peers without necessarily setting the command on each dial-peer.
22 This command enables the dial peer to use SIP profile 1
23 Configure the Cisco UBE SIP messaging to use the HSRP virtual address in SIP messaging. Once HSRP is configured under the physical interface and the bind command is issued, calls to the physical IP address will fail. This is because the SIP listening socket is now bound to the virtual IP address but the signaling packets use the physical IP address, and therefore cannot be handled.
24 This command used to pass RTP NTE (RFC2833) DTMF with respect to the dial peers used for the call.
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

do 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 GigabitEthernet0/0/3

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

dtmd-relay rtp-nue

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

25 This command enables T38 fax protocol for calls terminating on this dial-peer
destination-pattern [37][13][24]....... no modem passthrough session protocol sipv2 session target ipv4:10.80.22.2:5060 voice-class codec 1 voice-class sip asymmetric payload full voice-class sip asserted-id pai voice-class sip privacy-policy passthru voice-class sip early-offer forced voice-class sip profiles 1 voice-class sip bind control source-interface GigabitEthernet0/0/2 voice-class sip bind media source-interface GigabitEthernet0/0/2 dtmf-relay rtp-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 100 voip description "Outgoing To AT&T"-AT&T facing side destination-pattern 73236..... 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 profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n-te
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 300 voip
description " Int'l calls to AT&T - AT&T facing side "
destination-pattern 011T
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-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 GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
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 "
destination-pattern .11
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210

Note: Testing was conducted in tekVizion labs
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n-te
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 600 voip
description " N11 Calls to AT&T - IP-PBX facing side "
no modem passthrough
session protocol sipv2
incoming called-number .11
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-n-te
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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 141 voip
description "Network Feature"
translation-profile outgoing NPA
destination-pattern *..
no modem passthrough
session protocol sipv2
session target ipv4:207.242.225.210
voice-class codec 1
voice-class sip asymmetric payload full
voice-class sip asserted-id pai
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-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 GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-npe
fax rate 14400
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!
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
!
!
line con 0
logging synchronous
stopbits 1
line aux 0
stopbits 1
line vty 0
session-timeout 90
exec-timeout 960 0
no activation-character
logging synchronous
transport preferred ssh
transport input all
stopbits 1
line vty 1 4
exec-timeout 960 0
logging synchronous
transport input all
!
!
end

Cisco UCM Configuration

The configuration screen shots shows general over view of lab configuration for this interoperability testing.
Cisco UCM Version:

Cisco UCM Audio Codec Preference List

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

Cisco UCM 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

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 Mobility Related Information**

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

**Geolocation Configuration**

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

**Call Routing Information**

**Incoming Calling Party Settings**

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

<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>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Incoming Called Party Settings**

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

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

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

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

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

Buttons: Save, Delete, Copy, Reset, Apply Config, Add New
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.

Note: Testing was conducted in tekVizion labs
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)

```
<table>
<thead>
<tr>
<th>UC Service Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status</td>
<td></td>
</tr>
<tr>
<td>Status: Ready</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>Add a UC Service</td>
<td></td>
</tr>
<tr>
<td>UC Service Type:</td>
<td>CTI</td>
</tr>
<tr>
<td>Product Type:</td>
<td>CTI</td>
</tr>
<tr>
<td>Name*</td>
<td>CTI_SRV</td>
</tr>
<tr>
<td>Description</td>
<td>CTI for Jabber Clients</td>
</tr>
<tr>
<td>Host Name/IP Address*</td>
<td>10.80.22.2</td>
</tr>
<tr>
<td>Port</td>
<td>2748</td>
</tr>
<tr>
<td>Protocol</td>
<td>TCP</td>
</tr>
</tbody>
</table>
```

UC Service Configuration (Contd...)

Select UC Service Type: = IM and Presence
Set Name* = IMP_SRV. This is used in this example.

Note: Testing was conducted in tekVizion labs
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
Conferencing Profile

- Primary: <None>
- Secondary: <None>
- Tertiary: <None>
- Server Certificate Verification: Any

Credentials source for web conference service: Not set

Directory Profile

- Primary: <None>
- Secondary: <None>
- Tertiary: <None>

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

Username: administrator
Password: ********

- Search Base 1
- Search Base 2
- Search Base 3

- Recursive Search on All Search Bases
- Search Timeout (seconds): 5

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

IM and Presence Profile

- Primary: IMP_SRV
- Secondary: <None>
- Tertiary: <None>

CTI Profile

- Primary: CTI_SRV
- Secondary: <None>
- Tertiary: <None>
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...)
User Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>User Status</td>
<td>Enabled Local User</td>
</tr>
<tr>
<td>User ID*</td>
<td>jabber1</td>
</tr>
<tr>
<td>Password</td>
<td>-----------------------</td>
</tr>
<tr>
<td>Confirm Password</td>
<td>-----------------------</td>
</tr>
<tr>
<td>Self-Service User ID</td>
<td></td>
</tr>
<tr>
<td>PIN</td>
<td>-----------------------</td>
</tr>
<tr>
<td>Confirm PIN</td>
<td>-----------------------</td>
</tr>
<tr>
<td>Last name*</td>
<td>cisco</td>
</tr>
<tr>
<td>Directory URI</td>
<td><a href="mailto:jabber1@leb.tekvizion.com">jabber1@leb.tekvizion.com</a></td>
</tr>
</tbody>
</table>

Manager User ID

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Department</td>
<td></td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Associated PC</td>
<td></td>
</tr>
<tr>
<td>Digest Credentials</td>
<td></td>
</tr>
<tr>
<td>Confirm Digest Credentials</td>
<td></td>
</tr>
<tr>
<td>User Profile</td>
<td>Use System Default (“Standard (Factory Default) U) - View Details</td>
</tr>
</tbody>
</table>

Note: Testing was conducted in tekVizion labs
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**
  - **BLF Presence Group**
  - **SUBSCRIBE Calling Search Space**
  - **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

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

- **CAPF Information**
  - Associated CAPF Profiles

- **Permissions Information**
  - **Groups**
    - 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

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

![Diagram of SCCP Configuration settings]

- **External Data Locations Information (Leave blank to use default)**
  - Information
  - Directory
  - Messages
  - Services
  - Authentication Server
  - Proxy Server
  - Idle
  - Idle Timer (seconds)
  - Secure Authentication URL
  - Secure Directory URL
  - Secure Idle URL
  - Secure Information URL
  - Secure Messages URL
  - Secure Services URL

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

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

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

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

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Page 63 of 160
Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7965 SCCP Configuration (Continued...)

![Configuration Layout Diagram]

- **Product Specific Configuration Layout**
  - Enable Speakerphone
  - Enable Speakerphone and Headset
  - Forwarding Delay
  - PC Port
  - Settings Access
  - Gratuitous ARP
  - PC Voice VLAN
  - Access
  - Video Capabilities
  - Auto Line Select
  - Web Access
  - Days Display Not Active
  - Display On Time

![Configuration Details Diagram]

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

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

<table>
<thead>
<tr>
<th>Recording Tone</th>
<th>Disabled</th>
</tr>
</thead>
<tbody>
<tr>
<td>Local Volume</td>
<td>100</td>
</tr>
<tr>
<td>Remote Volume</td>
<td>50</td>
</tr>
<tr>
<td>Recording Tone</td>
<td></td>
</tr>
<tr>
<td>Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When</td>
<td></td>
</tr>
<tr>
<td>Incoming Call</td>
<td></td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
</tr>
<tr>
<td>Feature</td>
<td>Value</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): Switch Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

**Discovery**

| Protocol - Media Endpoint                  |                |
| Discover (LLDP-MED): Switch Port          |                |
| Link Layer Discovery                       | Enabled        |
| Discovery Protocol (LLDP): PC Port        |                |
| LLDP Advert ID                             |                |
| LLDP Power Priority                        | Unknown        |
| Wireless Headset Hookswitch Control       | Disabled       |
| IPv6 Load Server                           |                |
| IPv6 Log Server                            |                |
| 802.1x Authentication                      | User Controlled|
| Detect Unified                             | Normal         |
| CM Connection Failure                      |                |
| Minimum Ring Volume                        | 0-Silent       |
Cisco IP Phone 7965 SCCP Configuration (Continued...)

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

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

Set Directory Number* = 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...)
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EDCS# xxx Rev #
Page 70 of 160
Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7965 SCCP Configuration (Continued...)

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

**Park Monitoring**

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

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

| Target (Destination) | 
| MLPP Calling Search Space | < None > |
| MLPP No Answer Ring Duration (seconds) | < None > |
| Confidential Access Mode | 
| Confidential Access Level | < None > |
| Call Control Agent Profile | < None > |

**Line Settings for All Devices**

| Hold Reversion Ring Duration (seconds) | Setting the Hold Reversion Ring Duration to zero will disable the feature |
| Hold Reversion Notification Interval (seconds) | Setting the Hold Reversion Notification Interval to zero will disable the feature |
| Party Entrance Tone | Default |
Note: Testing was conducted in tekVizion labs
Cisco IP Phone 7965 SCCP Configuration (Continued...)
**Multiple Call/Call Waiting Settings on Device SEP44ADD9D56F39**

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

- Maximum Number of Calls*: 4
- Busy Trigger*: 2 (Less than or equal to Max. Calls)

**Forwarded Call Information Display on Device SEP44ADD9D56F39**

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

**Users Associated with Line**

[Associate End Users]

[Save] [Delete] [Reset] [Apply Config] [Add New]
Cisco IP Phone 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...)

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

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

## Extension Information

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

## MLPP and Confidential Access Level Information

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

## Do Not Disturb

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

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

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

### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Parameter Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
</tr>
<tr>
<td>PC Port * Enabled</td>
<td></td>
</tr>
<tr>
<td>Back USB Port * Enabled</td>
<td></td>
</tr>
<tr>
<td>Side USB Port * Enabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Camera * Disabled</td>
<td></td>
</tr>
<tr>
<td>Console Access * Disabled</td>
<td></td>
</tr>
<tr>
<td>Video Capabilities *</td>
<td></td>
</tr>
<tr>
<td>Enable/Disable USB Classes</td>
<td></td>
</tr>
<tr>
<td>SDIO *</td>
<td></td>
</tr>
<tr>
<td>Bluetooth * Disabled</td>
<td></td>
</tr>
<tr>
<td>Wifi *</td>
<td></td>
</tr>
<tr>
<td>Bluetooth Profiles *</td>
<td></td>
</tr>
<tr>
<td>Settings Access * Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP * Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access *</td>
<td></td>
</tr>
<tr>
<td>Web Access * Disabled</td>
<td></td>
</tr>
<tr>
<td>Show All Calls on Primary Line *</td>
<td></td>
</tr>
<tr>
<td>Days Display Not Active</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>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 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>Parameter</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>Default, Preset, Telephony</td>
</tr>
</tbody>
</table>
Cisco IP Phone 9971 SIP Configuration (Continued…)

Table of Configuration Settings:

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Advertise G.722 and iSAC Codecs</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP-MED): Switch Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td>Unknown</td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td>User Controlled</td>
</tr>
<tr>
<td>FIPS Mode</td>
<td>Disabled</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>Normal</td>
</tr>
<tr>
<td>Switch Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Power Negotiation*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Restrict Data Rates*</td>
<td>Disabled</td>
</tr>
<tr>
<td>SSH Access*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Incoming Call Toast Timer*</td>
<td>5</td>
</tr>
<tr>
<td>Provide Dial Tone from Release Button*</td>
<td>Disabled</td>
</tr>
</tbody>
</table>
Cisco IP Phone 9971 SIP Configuration (Continued...)

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

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

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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 H323</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>
</tbody>
</table>

Select Forwarding History

### Call Forward and Call Pickup Settings

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

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

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

No Answer Ring Duration (seconds)

Call Pickup Group < None >

---

### Park Monitoring

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

Park Monitoring Reversion Timer

Reversion Timer service parameter

A blank value will use value set in Park Monitoring

---

### MLPP Alternate Party And Confidential Access Level Settings

<table>
<thead>
<tr>
<th>Target (Destination)</th>
</tr>
</thead>
</table>

MLPP Calling Search Space < None >

MLPP No Answer Ring Duration (seconds)

Confidential Access Mode < None >

Confidential Access Level < None >

Call Control Agent Profile < None >

---

### Line Settings for All Devices

<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>
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...)
### Multiple Call/Call Waiting Settings on Device SEPC07BBCA1B811

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

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Maximum Number of Calls</td>
<td>4</td>
<td></td>
</tr>
<tr>
<td>Busy Trigger</td>
<td>2</td>
<td>(Less than or equal to Max. Calls)</td>
</tr>
</tbody>
</table>

### Forwarded Call Information Display on Device SEPC07BBCA1B811

- [ ] Caller Name
- [ ] Caller Number
- [ ] Redirected Number
- [x] 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 Profile Configuration used by SIP trunk to Cisco UBE

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

Set SIP profile Name * = Standard SIP Profile w/Early Media Disabled. This is used for this example
Check Disable Early Media on 180
Set SIP Rel1xx Options* = Send PRACK if 1xx contains SDP
Note*= Some PSTN network call prompters utilize early-media cut-through to offer menu options to the caller (DTMF select menu) before the call is connected. In order for Cisco UCM/Cisco UBE solution to achieve successful early-media cut-through, the Cisco UCM to Cisco UBE call leg must be enabled with SIP PRACK. To enable SIP PRACK on the Cisco UCM, the SIP Profile “SIP Rel1XX Options” setting must be set to “Send PRACK”.

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

### SDP Information

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</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>

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)*</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)*</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)*</td>
<td>3600</td>
</tr>
<tr>
<td>Timer T1 (msec)*</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)*</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE*</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port*</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port*</td>
<td>32768</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-serviceuri-gpickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info*</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level*</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back*</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block*</td>
<td>Off</td>
</tr>
</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

- Normalization Script: < None >

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

Incoming Requests FROM URI Settings

- Caller ID DN
- Caller Name
### Trunk Specific Configuration

<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>SIP Rel1XX Options*</td>
<td>Send PRACK if 1xx Contains SDP</td>
</tr>
<tr>
<td>Video Call Traffic Class*</td>
<td>Mixed</td>
</tr>
<tr>
<td>Calling Line Identification Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Session Refresh Method*</td>
<td>Invite</td>
</tr>
<tr>
<td>Early Offer support for voice and video calls</td>
<td>Disabled (Default value)</td>
</tr>
<tr>
<td>Enable ANAT</td>
<td></td>
</tr>
<tr>
<td>Deliver Conference Bridge Identifier</td>
<td></td>
</tr>
<tr>
<td>Allow Passthrough of Configured Line Device Caller Information</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Incoming Calls</td>
<td></td>
</tr>
<tr>
<td>Reject Anonymous Outgoing Calls</td>
<td></td>
</tr>
<tr>
<td>Send TLS Learned Destination Route String</td>
<td></td>
</tr>
</tbody>
</table>

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

Navigation: Device → Trunk
Set Device Name* = ATT_SIP_TRUNK. This is used for this example
Set Description = ATT SIP Trunk to PSTN. This is used for this example
Set Device Pool* = G729_pool. This is used for this example
Set Media Resource Group List = MRGL_MTP.

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

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

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

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

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

### Connected Party Settings
- Connected Party Transformation CSS: < None >
- Use Device Pool Connected Party Transformation CSS

### Outbound Calls
- Called Party Transformation CSS: < None >
- Use Device Pool Called Party Transformation CSS
- Calling Party Transformation CSS: < None >
- Use Device Pool Calling Party Transformation CSS
- Calling Party Selection: Originator
- Calling Line ID Presentation: Default
- Calling Name Presentation: Default
- Calling and Connected Party Info Format: Deliver DN only in connected party
- Redirecting Diversion Header Delivery: Outbound
- Redirecting Party Transformation CSS: < None >
- Use Device Pool Redirecting Party Transformation CSS

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 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

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Page 103 of 160
Note: Testing was conducted in tekVizion labs
Set Media Resource Group List = MRGL_MTP.

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

![Image of Cisco Unified CM Administration interface showing SIP Trunk status and device information.]

### SIP Trunk Status
- **Service Status**: Full Service
- **Duration**: Time in Full Service: 0 day 2 hours 36 minutes

### Device Information
- **Product**: SIP Trunk
- **Device Protocol**: SIP
- **Trunk Service Type**: None (Default)
- **Device Name**: Trunk_SIP_FAX_Gateway
- **Description**: Trunk to SIP FAX Gateway
- **Device Pool**: G729 Pool
- **Media Resource Group List**: MRGL_MTP
- **Location**: Hub_None
- **AAR Group**: < None >
- **Tunneled Protocol**: None
- **QSIG Variant**: No Changes
- **ASN.1 ROSE OID Encoding**: No Changes
- **Packet Capture Mode**: None
- **Packet Capture Duration**: 0

---

Note: Testing was conducted in tekVizion labs
<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Media Termination Point</td>
<td>Required</td>
</tr>
<tr>
<td>Retry Video Call as Audio</td>
<td>Enabled</td>
</tr>
<tr>
<td>Path Replacement Support</td>
<td>Enabled</td>
</tr>
<tr>
<td>Transmit UTF-8 for Calling Party Name</td>
<td>Required</td>
</tr>
<tr>
<td>Transmit UTF-8 Names in QSIG APDU</td>
<td>Required</td>
</tr>
<tr>
<td>Unattended Port</td>
<td>Enabled</td>
</tr>
<tr>
<td>SRTP Allowed</td>
<td>When using both sRTP and TLS</td>
</tr>
<tr>
<td>Consider Traffic on This Trunk</td>
<td>Secure</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>Enabled</td>
</tr>
<tr>
<td>Run On All Active Unified CM Nodes</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

**Intercompany Media Engine (IME)**

<table>
<thead>
<tr>
<th>Transformation Profile</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>E.164</td>
<td>Transformation Profile</td>
</tr>
</tbody>
</table>

**MLPP and Confidential Access Level Information**

<table>
<thead>
<tr>
<th>Information Type</th>
<th>Setting</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>Type</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Remote-Party-Id</td>
<td>Enabled</td>
</tr>
<tr>
<td>Asserted-Identity</td>
<td>Enabled</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>Type</th>
<th>Setting</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>Yes</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>No</td>
</tr>
</tbody>
</table>

--- Connected Party Settings ---

Connected Party Transformation CSS: < None >

Use Device Pool Connected Party Transformation CSS

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

<table>
<thead>
<tr>
<th>Outbound Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Called Party Transformation CSS</strong></td>
</tr>
<tr>
<td><strong>Use Device Pool Called Party Transformation CSS</strong></td>
</tr>
<tr>
<td><strong>Calling Party Transformation CSS</strong></td>
</tr>
<tr>
<td><strong>Use Device Pool Calling Party Transformation CSS</strong></td>
</tr>
<tr>
<td><strong>Calling Party Selection</strong></td>
</tr>
<tr>
<td><strong>Calling Line ID Presentation</strong></td>
</tr>
<tr>
<td><strong>Calling Name Presentation</strong></td>
</tr>
<tr>
<td><strong>Calling and Connected Party Info Format</strong></td>
</tr>
<tr>
<td><strong>Redirecting Diversion Header Delivery - Outbound</strong></td>
</tr>
<tr>
<td><strong>Redirecting Party Transformation CSS</strong></td>
</tr>
<tr>
<td><strong>Use Device Pool Redirecting Party Transformation CSS</strong></td>
</tr>
</tbody>
</table>

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

**Destination**
- **Destination Address is on SRV**
  - Destination Address: 10.00.22.7
  - Destination Address IPv6: 
  - Destination Port: 5060

- **MTP Preferred Originating Code**
  - 711ulaw

- **BIP Presence Group**
  - Standard Presence group

- **SIP Trunk Security Profile**
  - ATT Non Secure SIP Trunk Profile

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

- **Out-Of-Dialing Refer Calling Search Space**
  - < None >

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

- **SIP Profile**
  - Standard SIP Profile with Early Media Disabled

- **DTHF Signaling Method**
  - No Preference

---

### 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 clusters with recording-enabled gateways**

### Geolocation Configuration

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

---

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Page 108 of 160
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

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>4004</td>
<td>To FAX</td>
<td></td>
<td></td>
<td>Trunk_SIP_FAX_Gateway</td>
<td></td>
</tr>
<tr>
<td>9.*XI</td>
<td>Network-Based Call Forwarding</td>
<td></td>
<td></td>
<td>ATT_SIP_TRUNK</td>
<td></td>
</tr>
<tr>
<td>9.@</td>
<td>To PSTN via ATT SIP Trunk</td>
<td></td>
<td></td>
<td>ATT_SIP_TRUNK</td>
<td></td>
</tr>
</tbody>
</table>

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

Pattern Definition

- **Route Pattern**: 9 02
- **Route Partition**: <None>
- **Description**: To PSTN via ATT SIP Trunk

Gateway/Route List

- **Gateway/Route List**: ATT_SIP_TRUNK

Route Option

- **Call Classification**: Offnet
- **External Call Control Profile**: <None>
- **Allow Device Override**: Provide Outside Dial Tone
- **Allow Overlap Sending**: [ ]
- **Urgent Priority**: [ ]
- **Require Forced Authorization Code**: [ ]
- **Authorization Level**: 0
- **Require Client Mapper Code**: [ ]

Calling Party Transformations

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

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EDCS# xxx Rev #
Page 110 of 160
Note: Testing was conducted in tekVizion labs
### Connected Party Transformations

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

### Called Party Transformations

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

### ISDN Network-Specific Facilities Information Element

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

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

**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 >

[Save] [Delete] [Copy] [Add New]
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>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation*</td>
<td>Default</td>
</tr>
</tbody>
</table>

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

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

Note: Testing was conducted in tekVizion labs
Jabber Client Configuration

**Navigation:** Device → Phone

Select Phone Type* = Cisco Unified Client services framework
Set Device Name* = CSFUser1. This is used in this example.
Set Description = CSFUser1. This is used in this example.
Select Device Pool = G729 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>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_MTP</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Built In Bridge*</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Mode*</td>
<td>Default</td>
</tr>
<tr>
<td>Owner</td>
<td>User</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>unchecked</td>
</tr>
<tr>
<td>Allow Control of Device from CTI</td>
<td>checked</td>
</tr>
<tr>
<td>Logged Into Hunt Group</td>
<td>unchecked</td>
</tr>
<tr>
<td>Remote Device</td>
<td>unchecked</td>
</tr>
<tr>
<td>Require off-premise location</td>
<td>unchecked</td>
</tr>
</tbody>
</table>
## Jabber Client Configuration (Contd...)

### Number Presentation Transformation

**Caller ID For Calls From This Phone**

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

### Remote Number

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

### Protocol Specific Information

- **Packet Capture Mode**: None
- **Packet Capture Duration**: 0
- **BLF Presence Group**: Standard Presence group
- **SIP Dial Rules**: `<None>`
- **MTP Preferred Originating Codec**: `711ulaw`
- **Device Security Profile**: Cisco Unified Client Services Framework - Standard
- **Rerouting Calling Search Space**: `<None>`
- **SUBSCRIBE Calling Search Space**: `<None>`
- **SIP Profile**: Standard SIP Profile w/Early Media Disabled
- **Digest User**: `<None>`

- **Media Termination Point Required**: [ ]
- **Unattended Port**: [ ]
- **Require DTMF Reception**: [ ]
Note: Testing was conducted in tekVizion labs
Jabber Client Configuration (Contd...)
**Desktop Client Settings**

- **Automatically Start in Phone Control**: Disabled
- **Automatically Control Tethered Desk Phone**: Disabled
- **Extend and Connect Capability**: Enabled
- **Display Contact Photos**: Enabled
- **Number Lookups on Directory**: Enabled
- **Jabber For Windows Software Update Server URL**: Enable
- **Problem Report Server URL**: Enable
- **Analytics Collection**: Disabled

**Analytics Server URL**

- **Cisco Support Field**

---

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

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

![Voicemail Port Configuration](image)

- **Device Name:** CiscoUM1-V11
  - **Description:** VoiceMail
  - **Device Pool:** G729 Pool
  - **Device Security Mode:** Non Secure Voice Mail Port
  - **Calling Search Space:** 2301
  - **Extension:** 2301
  - **Partition:** Registered with clus32pubsub
  - **IP Address:** 10.80.22.4

- **Device Name:** CiscoUM1-V12
  - **Description:** VoiceMail
  - **Device Pool:** G729 Pool
  - **Device Security Mode:** Non Secure Voice Mail Port
  - **Calling Search Space:** 2302
  - **Extension:** 2302
  - **Partition:** Registered with clus32pubsub
  - **IP Address:** 10.80.22.4

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.
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
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:

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

<table>
<thead>
<tr>
<th>Device#</th>
<th>PID</th>
<th>SN</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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EDCS# xxx Rev #
Page 129 of 160
Note: Testing was conducted in tekVizion labs
Technology Package License Information for Module: 'c2900'

<table>
<thead>
<tr>
<th>Technology Package</th>
<th>Current</th>
<th>Type</th>
<th>Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipbase</td>
<td>ipbase9</td>
<td>Permanent</td>
<td>ipbasek9</td>
</tr>
<tr>
<td>security</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>uc</td>
<td>ucken</td>
<td>Permanent</td>
<td>ucken9</td>
</tr>
<tr>
<td>data</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>NtwkEss</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>CollabPro</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
</tbody>
</table>

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.1QCmm7yYduKGZl

! no aaa new-model

!
ip domain name lab.tekvizion.com
ip name-server 10.64.1.3
ip cef
no ipv6 cef
multilink bundle-name authenticated

stcapp feature access-code

stcapp feature speed-dial
cts logging verbose

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
6966696D 6174652D 323313839 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 8FB4D8BC 1DDD4D7A 0882107F B14FCB24 699A35A9
D521C88A 5B43F4FC D394E945 81A1380A 2E753478 93190ADE 75AA8971 883E9214
C607CCDF 6FCCDE9C 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," ***

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!
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

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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

[Diagram of CUC Port Group interface]
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

![Image of Cisco Unity Connection Administration interface]

<table>
<thead>
<tr>
<th>Display Name</th>
<th>Phone System Display Name</th>
<th>Extension</th>
<th>Server</th>
<th>Enabled</th>
<th>Answer Calls</th>
<th>Message Notification</th>
</tr>
</thead>
<tbody>
<tr>
<td>CUCM-1-001</td>
<td>CUCM</td>
<td>cluster2unity</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td></td>
</tr>
<tr>
<td>CUCM-1-002</td>
<td>CUCM</td>
<td>cluster2unity</td>
<td>X</td>
<td>X</td>
<td>X</td>
<td></td>
</tr>
</tbody>
</table>

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

**Navigation:** System ➔ Presence Topology

Note: Testing was conducted in tekVizion labs
Node Configuration

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

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

![Image of Cisco Unified CM IM and Presence Administration interface](image-url)

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

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<table>
<thead>
<tr>
<th>Corporate Headquarters</th>
<th>European Headquarters</th>
<th>Americas Headquarters</th>
<th>AsiaPacific Headquarters</th>
</tr>
</thead>
<tbody>
<tr>
<td>170 West Tasman Drive</td>
<td>International BV</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlerbergpark</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td>Haarlerbergweg 13-19</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
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