AT&T IP Toll-Free and IP Transfer Connect service on MIS, MPLS PNT and AT&T VPN: Connecting Cisco Unified Customer Voice Portal 8.0(1) via the Cisco Unified Border Element (Enterprise Edition) 1.4 on ASR1000 Aggregation Services Routers using SIP

Dec 16, 2011—Version 3

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Introduction

Service Providers today, such as AT&T, are offering alternative methods to connect to the public switched telephone network (PSTN) via their IP network. Most of these services utilize Session Initiation Protocol (SIP) as the primary signaling method and a centralized IP to TDM gateway to provide on-net and off-net services. AT&T IP Toll Free 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, inter-working and session management services.

AT&T IPTF

AT&T IP Toll-Free Service (AT&T IPTF) is one of the BVoIP services that provides inbound (only) toll-free service from the PSTN to the customer premise equipment (CPE).

Cisco Unified Customer Voice Portal

The Cisco Unified Customer Voice Portal (CVP) 8.0(1) is a web-based platform that provides carrier-class interactive voice response (IVR) and internet protocol (IP) switching services over Voice Over IP (VoIP) Networks. When combined with the Cisco Intelligent Contact Management Enterprise (ICME) provides IVR services with efficient queuing at the edge of the Network.

- This application note describes how to configure a Cisco Unified Customer Voice Portal (CVP) 8.0(1) with a Cisco Unified Border Element (Cisco UBE) 1.4 on ASR 1000 Aggregation Services Router for connectivity to AT&T’s IP Toll-Free and IP Transfer Connect SIP trunk services. The deployment model covered in this application note is CVP comprehensive solution to PSTN (AT&T IP toll-free SIP). AT&T’s IP Toll-Free provides inbound call service only (PSTN to CPE).
- The AT&T IP Flexible Reach service was not tested within this cycle. Outbound calls (CPE to PSTN), including emergency 911 calls, are not documented or supported with this configuration guide. Flexible Reach is supported with Cisco Unified Communications Manager and is documented on the Cisco Interoperability Portal.
- Testing was performed in accordance to AT&T’s IP Toll-Free test plan version 9/23/2010. Key features verified are: Basic Call, DNIS translations, DNIS based Routing, CED based Routing, Call Center Queuing, Codec Negotiation, Legacy Transfer Connect (8YY transfer), Intra-site Transfers, Intra-site Conferencing, IP Transfer Connect- SIP Redirect, and IP Transfer Connect- SIP REFER Unattended Transfer.
- A limited regression testing was performed in accordance to AT&T IP Toll-Free test plan version 10/6/2011 using Cisco IOS version 15.1.2T4.1 on the Ingress VXML Gateway to verify “Ptime = 0” issue (CSCtl07660) and SIP INFO to RFC2833 feature (CSTj93573).
- For IP Toll-Free service with MIS/MPLS PNT (and optionally with AT&T VPN) access, the Cisco UBE IP address (facing the Customer Edge Router) can be private IP address. This will be NATed by the AT&T managed Customer Edge Router (or customer managed/MRS managed Customer Edge Router for AT&T VPN). Consult with AT&T provisioning engineer to resolve any IP addressing issues.
- The Cisco Unified Border Element 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 Unified Customer Voice Portal. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying Cisco UBE in a production environment, to ensure these commands are set per each dial-peer requiring interoperating to AT&T SIP network.
Network Topology

CVP 8.0(1) Comprehensive solution topology
Call flow sequence (Basic G.711 call flow)

Initial call to IVR service is G.711. Call in Q
Call is routed to G.711 agent that is Ready
System Components

Hardware Requirements

- Cisco ASR 1000 Aggregation Services Router running Cisco UBE
- Cisco ISR C3825 gateway running Cisco UBE
  - Cisco Unified Border Element is an integrated Cisco IOS Software application that runs on various IOS platforms, follow the link for more details: [http://www.cisco.com/go/cube](http://www.cisco.com/go/cube)
- Cisco Unified SIP Proxy (CUSP)
- Cisco MCS 7800 Series server (Cisco Unified Communications Manager)
- Cisco IP Phones (The topology diagram shows 7961 and 7975, but any Cisco IP phone model supporting RFC2833 can be used)
- Cisco MCS 7800 Series server (Cisco Unified Customer Voice Portal Server)¹
- Cisco MCS 7800 Series server (Cisco Unified Intelligent Contact Management Enterprise)
- Cisco MCS 7800 Series server (Microsoft Windows Server 2003 R2 SP2 AD/Domain Server)

Software Requirements

- ASR 1002 Cisco Unified Border Element (Enterprise Edition) Release 1.4 with Cisco IOS XE Software version 03.02.01.S.151-1.S1. This configuration was tested with `asr1000rp1-adventerprisek9.03.02.01.S.151-1.S1.bin`.
- Cisco Unified Border Element Release 8.5 with Cisco IOS version 15.1.2T4.1 release. This configuration was tested with `c3825-ipvoicek9-mz.151-2.T4.1.bin`. File can be obtained from: [https://upload.cisco.com/cgi-bin/swc/fileexg/main.cgi?CONTYPES=ATT-Managed-Services CCO access is required.]
- Cisco Unified CM 8.0 and later 8.x releases. This solution was tested with 8.0.3.10000-8
- Cisco Unified SIP Proxy version 1.1.4
- Cisco Unified Intelligent Contact Management Enterprise 8.0.1.0
- Microsoft Windows Server 2003 R2 SP2
- Cisco CTI OS 8.0.1

Features

Features Supported

- Basic Call using G.711ulaw
- Calling Party Number Presentation and Restriction
- AT&T Legacy Transfer Connect (8YY transfer). See limitations section for details.
- Intra-site Call Transfers
- Intra-site Conference

¹ The Cisco Unified Customer Voice Portal Server implements a SIP back-to-back user agent (B2BUA) and IVR Service that is the central component in a Cisco Unified Intelligent Contact Management Enterprise-integrated Cisco Unified CVP implementation.
• CPE initiated Hold and Resume
• Incoming DNIS Translation
• DNIS based Routing
• CED based Routing
• Call Center Queuing
• Codec Negotiation. See limitations section for details.
• IP Transfer Connect- SIP Redirect
• IP Transfer Connect- SIP REFER Unattended transfer, see limitations section for details.
• Dynamic DTMF Payload Type

Features Not Supported
• Cisco UBE (Ent) version 3.2.1S does not support mid call Codec negotiation.
• Cisco UBE (Ent) version 3.2.1S does not support passing UUI messages in the REFER header to the Network side.

Limitations
These are the known limitations, caveats, or integration issues:
• The CVP 8.0(1) solution guide for IP Toll-Free service is based on the use of G.711 codec.
• Although AT&T SIP INVITE proposes a list of codecs (e.g. G729, G711, G726) the CVP initial call to the IVR service will only work using G.711 codec . See Cisco UBE configuration section for details.
• VXML functionality for CVP solution is not planned on the ASR 1000 CUBE, therefore the solution requires a separate ISR based VXML Gateway. The VXML Gateway is also configured with CUBE ENT functionality to support SIP INFO to RFC 2833 Interworking.
• ASR 1000 CUBE does not support mid-call codec negotiation on release 3.2.1S.
• When using mixed environment of G.729 and G.711 desktop agents end points, it is required to configure a hardware based conference bridge (CFB) resource to be used by Cisco Unified Communications Manager IP phone to initiate a three-way conference between G.729 and G.711 media end-points. See configuration section for details. The hardware Conference Bridge resource can be co-resident with the Cisco ISR based VXML Gateway.
• Caller DTMF key presses using RFC2833:
  • for DTMF on inbound (example to enter account number or select menu options) use "dtmf-relay rtp-nte" on the incoming dial peer from AT&T Network. For DTMF on outbound (this is required for Legacy Transfer Connect 8YY feature) set the dtmf-relay to sip-notify and rtp-nte. In order to accomplish this, the incoming dial-peer needs to be split into two legs, one incoming to the Cisco UBE from AT&T and one outgoing to the SIP Proxy. Use "dtmf-relay sip-notify rtp-nte" for the SIP Proxy leg in order to get legacy Transfer Connect Human to work for manual entered DTMF digits when using CTIO Agent Desktop’s Dialing pad. See Cisco UBE configuration section for details.
  • Intra site Blind/Attended transfer: use CTI Post Route DN (in this case DN 6000) in the "number to Dial" field instead of the actual agent number to do intra site transfer in order to hear ring back when agent 2 is ringing. CTI Route Point DN should be used with a "send to VRU" node in the UCCE script, in order to route the post route through CVP. From the CTI Agent Desktop, use the transfer tab to initiate the transfer. In the CTI Dialing Pad window, use Post Route DN 6000 as a number to Dial. As an option, use the More tab to enter specific DN in "Var1” field to transfer to a specific target agent.
  • For Legacy Transfer Connect using VRU label, CVP sends the dtmf digits in SIP INFO message to the ingress VXML Gateway and the Gateway does process the dtmf SIP INFO to RFC 2833 upstream to the Network. This functionality was not supported on Cisco UBE
15.1.2T (CSCtj93573). Also CVP 8.0 does not expect any inband response after initiating the transfer and will not deal with any inband responses coming back from the Network side.

- CVP 8.0 with ASR CUBE supports Legacy TCS - BT subscription only. This is the network based - unattended transfer capability where the network sends a BYE message to CVP.

- **ADR-486 Busy Here:** The Alternate Destination Routing (ADR) feature provides the ability to redirect calls to other sites/agents. The ADR on Busy allows the call to be redirected to a secondary site/agent when ADR on Busy condition is detected at the primary site/agent. In order to get the “486 Busy Here” condition to the Network side, the following workarounds need to be configured:

- The CVP 8.0 by default is not passing the "486 Busy Here” message to the Network side. A workaround is to define a voice class SIP profile in the ingress Gateway and apply this voice class to the outgoing to CVP dial-peer. See Ingress Gateway configuration section for details.

- **ADR-408 Request Timeout:** There is new AT&T Network Element that is timing out at 3 seconds after initial SIP INVITE. In a controlled lab environment, it was observed that Cisco UBE with default value of 500 for the sip-ua timer "trying" will send "408 Request Timeout” after 3.5 seconds. It seems the Network timeout value of 3 seconds will trigger ADR when encountering Request Timeout condition. It is not clear if this will cause issues in real production environments. The timer value on Cisco UBE can be reduced to respond faster than 3 seconds to prevent Network timeout, however this might not be optimum setting.

- **IP Transfer Connect-SIP REDIRECT:**

- **REDIRECT 302 Moved Temporarily:** In order to get Redirect 302 header to be sent to the network side, the CUSP SIP proxy service ‘recursive’ needs to be turned off using CLI command “no route recursion” (defaults to On).

- **REDIRECT/REFER header with UUI message:** passing UUI info in REDIRECT/REFER header to the Network side is not supported by CISCO UBE.

- **IP Transfer Connect-SIP REFER Unattended:**

- CVP 8.0(1) supports Unattended SIP REFER transfer only. Attended SIP REFER transfer is not supported.

- REFER transfer via CISCO UBE requires that the Network transfer Enable variable be set to 1 in the initial call’s VRU script and also in the Refer label script. In addition "Network Transfer preferred” check box need to be checked for both CCMPIM in the CCMPG and VRUPG.

- For REFER call flows, AT&T sends mid call re-INVITE with SDP (with Media Attribute (a):sendonly) to place Redirecting Party on hold. Cisco UBE however does not forward the mid call re-INVITE downstream to CVP but does respond to the Network with desired SIP 200 OK with SDP (with Media Attribute (a):inactive). The Network is unaware of this limitation and considers the Redirecting Party is on Hold. Router query on the REFER operation is also not supported in CVP 8.0(1)

- CVP 8.0(1) does not support error scenarios or reconnect. If REFER is rejected by the Network (example 403 Forbidden, 404 Not Found), CVP 8.0(1) will not play error announcement. CVP will disconnect the call by sending BYE. This could be problematic from the user experience point of view.

---

**Configuration**

Cisco ASR1K CUBE version

ASR-CUBE-GW#sh version
Cisco IOS Software, IOS-XE Software (PPC_LINUX_IOSD-ADVENTERPRISEK9-M), Version 15.1(1)S1, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Thu 10-Feb-11 23:47 by mcpre
Cisco IOS-XE software, Copyright (c) 2005-2011 by cisco Systems, Inc. All rights reserved. Certain components of Cisco IOS-XE software are licensed under the GNU General Public License ("GPL") Version 2.0. The software code licensed under GPL Version 2.0 is free software that comes with ABSOLUTELY NO WARRANTY. You can redistribute and/or modify such 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

ASR-CUBE-GW uptime is 1 day, 22 hours, 1 minute
Uptime for this control processor is 1 day, 22 hours, 3 minutes
System returned to ROM by reload
System image file is "bootflash:asr1000rp1-adventerprisek9.03.02.01.S.151-1.S1.bin"
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: http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

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

cisco ASR1002 (2RU) processor with 1710406K/6147K bytes of memory.
4 Gigabit Ethernet interfaces
32768K bytes of non-volatile configuration memory.
4194304K bytes of physical memory.
7798783K bytes of eUSB flash at bootflash:

Configuration register is 0x2102

ASR-CUBE-GW#
**Configuring Cisco ASR1K CUBE**

Critical commands are marked bold with footnote and description at bottom of the page

ASR-CUBE-GW#
ASR-CUBE-GW#sh run
Building configuration...

Current configuration : 4457 bytes
!
! Last configuration change at 17:29:34 UTC Tue Nov 9 2010
!
version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname ASR-CUBE-GW
!
boot-start-marker
boot system bootflash:asr1000rp1-adventerprisek9.BLD_V151_1_S_XE32_THROTTLE_LATE_ST_20101012_060049_2.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging buffered 5000000
enable secret 5 $1$G1jk$XNh5Cnu1yoM2IGvoEEFe21
enable password cisco
!
no aaa new-model
!
!
ip source-route
!
!
no ip domain lookup
ip host CVP 172.20.224.252
ip host vxml-gw 172.20.224.5
!
!
!
!
multilink bundle-name authenticated
!
!
voice-card 0/1
ASR1K CUBE needs this config to get dtmf Dynamic Payload Type Interworking.

dsp services dspfarm
!
!
!
voice service voip
address-hiding
allow-connections sip to sip
redirect ip2ip
signaling forward unconditional
fax protocol pass-through g711ulaw
h323
sip
header-passing error-passthru
asymmetric payload full
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
voice class codec 1
  codec preference 1 g711ulaw
!
voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g729br8
  codec preference 3 g711ulaw
!
voice class codec 3
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g729br8
!
voice class codec 4
  codec preference 1 g729r8
!
voice class codec 5
  codec preference 1 g729br8
!
voice class codec 6
  codec preference 1 g729r8
  codec preference 2 g711ulaw
!
voice class codec 7
  codec preference 1 g729br8
  codec preference 2 g711ulaw
!
!
http client cache memory pool 15000
http client cache memory file 500
http client cache refresh 600
http client connection idle timeout 60
http client connection timeout 10
http client response timeout 30
mrcp client timeout connect 10
mrcp client timeout message 10
mrcp client rtpsetup enable
redundancy
  mode none
  !
  !
  
interface GigabitEthernet0/0/0
ip address 172.20.224.10 255.255.255.0
negotiation auto
!
interface GigabitEthernet0/0/1
description connection to ATT Network
ip address 70.X.X.X 255.255.255.248
negotiation auto
!
interface GigabitEthernet0/0/2
no ip address
shutdown
negotiation auto
!
interface GigabitEthernet0/0/3
no ip address
shutdown
negotiation auto
!
interface Service-Engine0/1/0
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
ip address 172.20.224.11 255.255.255.0
negotiation auto
!
!
no ip http server
no ip http secure-server
ip route 207.X.X.200 255.255.255.255 70.X.X.110
!
logging esm config
access-list 101 permit ip host 70.X.X.X host 207.X.X.200
access-list 101 permit ip host 207.X.X.200 host 70.X.X.X
access-list 101 deny ip any any
!
!
control-plane
!
!
!

dspfarm profile 1 transcode

Outside IP addresses intentionally blocked throughout the document for security purpose and to minimize threat of outside attack on our DMZ Network.
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729br8
codec ilbc
maximum sessions 100
associate application CUBE
!
dial-peer voice 888 voip
description CVP -needed for SIP Refer/Redirect transfer connect
destination-pattern 10..
session protocol sipv2
session target ipv4:207.X.X.X:5060
 voice-class codec 3
dtmf-relay rtp-nle
no vad
!
dial-peer voice 1057 voip
description to CVP-outgoing to CVP via Ingress GW
destination-pattern 00000T
session protocol sipv2
session target ipv4:172.20.224.5:5060
 voice-class codec 3
dtmf-relay rtp-nle
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 9191 voip
description CVP -needed for SIP Refer/Redirect using 11 digit SDC number
destination-pattern 91919191919
session protocol sipv2
session target ipv4:207.X.X.X:5060
 voice-class codec 3
dtmf-relay rtp-nle
no vad
!
dial-peer voice 1058 voip
description Incoming from AT&T
session protocol sipv2
incoming called-number 00000T
 voice-class codec 3
dtmf-relay rtp-nle
fax protocol pass-through g711ulaw
no vad
!
!
line con 0
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
exec-timeout 15 0
password cisco
login
!
exception data-corruption buffer truncate
end

ASR-CUBE-GW#

ASR-CUBE-GW# sh inventory
NAME: "Chassis", DESCR: "Cisco ASR1002 Chassis"
PID: ASR1002 , VID: V02, SN: FOX1405GNTR

NAME: "module F0", DESCR: "Cisco ASR1000 Embedded Services Processor, 10Gbps"
PID: ASR1000-ESP10 , VID: V04, SN: JAE1340K39G

NAME: "Power Supply Module 0", DESCR: "Cisco ASR1002 AC Power Supply"
PID: ASR1002-PWR-AC , VID: V02, SN: ART1415U0BH

NAME: "Power Supply Module 1", DESCR: "Cisco ASR1002 AC Power Supply"
PID: ASR1002-PWR-AC , VID: V02, SN: ART1415U0AQ

NAME: "module 0", DESCR: "Cisco ASR1002 SPA Interface Processor 10"
PID: ASR1002-SIP10 , VID: V04, SN: JAE141501FT

NAME: "SPA subslot 0/1", DESCR: "DSP Shared Port Adapter"
PID: SPA-DSP , VID: V01, SN: JS6133ABQR

NAME: "SPA subslot 0/0", DESCR: "4-port Gigabit Ethernet Shared Port Adapter"
PID: 4XGE-BUILT-IN , VID: V00, SN: N/A

NAME: "subslot 0/0 transceiver 0", DESCR: "GE T"
PID: N/A , VID: E , SN: MTC140906AM

NAME: "subslot 0/0 transceiver 1", DESCR: "GE T"
PID: N/A , VID: E , SN: MTC140905KS

NAME: "module R0", DESCR: "Cisco ASR1002 Route Processor 1"
PID: ASR1002-RP1 , VID: V04, SN: JAE141501J
Cisco Ingress and VXML Gateway IOS version

Cisco IOS Software, 3800 Software (C3825-IPVOICEK9-M), Version 15.1(2)T4.1, MAIN
TENANCE INTERIM SOFTWARE
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2011 by Cisco Systems, Inc.
Compiled Tue 30-Aug-11 07:52 by prod_rel_team

ROM: System Bootstrap, Version 12.4(13r)T, RELEASE SOFTWARE (fc1)

c3825-VXML-GW uptime is 1 week, 3 hours, 29 minutes
System returned to ROM by reload at 19:54:39 UTC Tue Sep 13 2011
System image file is "flash:c3825-ipvoicek9-mz.151-2.T4.1.bin"
Last reload type: Normal Reload

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

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

If you require further assistance please contact us by sending email to
Cisco 3825 (revision 1.1) with 227328K/34816K bytes of memory.

Processor board ID FTX1051A2DZ

2 Gigabit Ethernet interfaces
2 Voice FXS interfaces

DRAM configuration is 64 bits wide with parity enabled.

479K bytes of NVRAM.

62720K bytes of ATA System CompactFlash (Read/Write)

License Info:

License UDI:

-------------------------------------------------
Device#   PID                  SN
-------------------------------------------------
*0        CISCO3825            FTX1051A2DZ

Configuration register is 0x2102
Configuring Cisco Ingress and VXML Gateway

Critical commands are marked bold with footnote and description at bottom of the page

c3825-VXML-GW#
c3825-VXML-GW#sh run
Building configuration...

Current configuration : 6152 bytes
! Last configuration change at 18:19:47 UTC Fri Nov 5 2010
! version 15.1
service timestamps debug datetime msec localtime
service timestamps log datetime msec
no service password-encryption
!
hostname c3825-VXML-GW
!
boot-start-marker
boot-end-marker
!
logging buffered 2000000
no logging console
enable password cisco
!
no aaa new-model
!
dot11 syslog
ip source-route
!
ip cef
!
!
!
ip host cvp 172.20.224.252
ip host CUP-CVP 172.20.224.241
ip host vxml-gw 172.20.224.5
no ipv6 cef
multilink bundle-name authenticated
!
!
!
!
voice-card 0
dspfarm
dsp services dspfarm\(^2\)
!
!
!
voice service voip

\(^2\) This command enables DSP farming, allowing DSP resources to register to Cisco Unified CM as MTP, CFB or Transcoder devices.
no ip address trusted authenticate

```
address-hiding
  allow-connections sip to sip
redirect ip2ip
signaling forward unconditional
fax protocol pass-through g711ulaw
h323
sip
header-passing
error-passthru
  early-offer forced
midcall-signaling passthru
  g729 annexb-all
voice class codec 1
  codec preference 1 g711ulaw

voice class codec 2
  codec preference 1 g729r8
  codec preference 2 g729br8
  codec preference 3 g711ulaw

voice class codec 3
  codec preference 1 g711ulaw
  codec preference 2 g729r8
  codec preference 3 g729br8

voice class sip-profiles 2
request INVITE sip-header Remote-Party-ID modify "<sip:(.*)@(.*)>" "--CVP <sip:\1@\2>"
```

3. Enables IP addressing hiding between the private network (CVP side) and the public network (AT&T IPTF side)
4. This command enables CISCO UBE basic IP-to-IP voice communication feature.
5. This command must be enabled at a global level to maintain integrity of SIP signaling between AT&T network and Cisco Unified CVP across CISCO UBE.
6. This command enables multiple codec support and performs codec filtering required for correct interoperability between AT&T SIP network and Cisco Unified CVP.
voice translation-profile DNIS
  translate called 1

http client cache memory pool 15000
http client cache memory file 500
http client cache refresh 600
http client connection timeout 60
http client connection idle timeout 10
http client response timeout 30
ivr prompt memory 15000
ivr asr-server rtsp://asr-en-us/recognizer
ivr tts-server rtsp://tts-en-us/synthesizer

application
  service new-call flash:bootstrap.vxml
  service survivability flash:survivability.tcl
  service CVPSelfService flash:CVPSelfServiceBootstrap.vxml
  service ringtone flash:ringtone.tcl
  service cvperror flash:cvperror.tcl
  service handoff flash:handoff.tcl
  service bootstrap flash:bootstrap.tcl

mrcp client timeout connect 10
mrcp client timeout message 10
mrcp client rtpsetup enable
vxml tree memory 500

license udi pid CISCO3825 sn FTX1051A2DZ

archive
  log config
  hidekeys

interface GigabitEthernet0/0
  no ip address
  shutdown
duplex auto
speed auto
media-type rj45
!
interface GigabitEthernet0/1
  ip address 172.20.224.5 255.255.255.0
duplex auto
speed auto
media-type rj45
!
ip forward-protocol nd
!
nop http server
nop http secure-server
!
ip route 0.0.0.0 0.0.0.0 172.20.224.1
ip route 172.20.224.0 255.255.255.0 GigabitEthernet0/1
!
logging esm config
!
!
control-plane
!
!
voice-port 0/0/0
!
voice-port 0/0/1
!
mgcp fax t38 ecm
!
scpp local GigabitEthernet0/1
scpp ccm 172.20.224.254 identifier 1 version 7.0
scpp
!
scpp ccm group 1
  associate ccm 1 priority 1
  associate profile 2 register MTP001A6D006911
  associate profile 1 register CFB001A6D006911
  associate profile 3 register 001A6D006911
!
dspfarm profile 2 transcode
  codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
codec g729br8
  maximum sessions 12
  associate application SCCP
!
dspfarm profile 1 conference
  codec g711ulaw
codec g711alaw
codec g729ar8

These commands configure the shared DSP resources such as conference bridge (CFB), and transcoder (MTP) devices for Cisco Unified CVP.

When using mixed environment of G.729 and G.711 desktop agents end points, it is required to configure a hardware based conference bridge (CFB) resource to be used by Cisco Unified Communications Manager IP phone to initiate a three-way conference between G.729 and G.711 media end-points.
codec g729abr8
codec g729r8
codec g729br8
maximum sessions 10
associate application SCCP
!
dspfarm profile 3 mtp
codec g711ulaw
maximum sessions hardware 50
associate application SCCP
!
dial-peer voice 919191 voip
description CVP SIP ringtone dial-peer
service ringtone
incoming called-number 91919191T
voice-class codec 1
voice-class sip rel1xx disable
dtmf-relay rtp-n te h245-signal h245-alphanumeric
no vad
!
dial-peer voice 929292 voip
description CVP SIP error dial-peer
service cvperror
incoming called-number 92929292T
voice-class codec 1
voice-class sip rel1xx disable
dtmf-relay rtp-n te h245-signal h245-alphanumeric
no vad
!
dial-peer voice 111111 voip
description CVP IVR dial-peer
service bootstrap
incoming called-number 1111111111T
voice-class codec 1
voice-class sip rel1xx disable
dtmf-relay rtp-n te h245-signal h245-alphanumeric
no vad
!
dial-peer voice 4000 pots
destination-pattern 40..
port 0/0/0
forward-digits 0
!
dial-peer voice 2000 pots
destination-pattern 20..
port 0/0/1
forward-digits 0
!
dial-peer voice 888 voip
description CVP -needed for SIP Refer/Redirect transfer connect
destination-pattern 10..
session protocol sipv2
session target ipv4:172.20.224.10:5060
voice-class codec 3
dtmf-relay rtp-n te
no vad
!
When using VRU label to send DTMF digits, CVP sends the \textit{dtmf} digits in SIP INFO message to the ingress Gateway and the Gateway does process the \textit{dtmf} SIP INFO to RFC 2833 upstream to the Network. This functionality is an enhancement feature on Cisco UBE 15.1.2T4.1 release.

Use this command on this dial-peer in order to recognize caller DTMF inputs and also to get legacy Transfer Connect to work for manual entered \textit{dtmf} digits when using CTIO Agent Desktop’s Dialing pad or when using VRU label to send DTMF digits.

\footnote{This command sets the SIP server target for incoming SIP calls. In this case the CUSP Proxy server.}
scheduler allocate 20000 1000
end

c3825-VXML-GW#
c3825-VXML-GW#
c3825-VXML-GW#sh inventory
NAME: "3825 chassis", DESCR: "3825 chassis"
PID: CISCO3825 , VID: V03 , SN: FTX1051A2DZ

NAME: "2nd generation two port FXS voice interface daughtercard on Slot 0 SubSlot 0", DESCR: "2nd generation two port FXS voice interface daughtercard"
PID: VIC2-2FXS , VID: V01 , SN: FOC12527C7K

NAME: "PVDMII DSP SIMM with four DSPs on Slot 0 SubSlot 4", DESCR: "PVDMII DSP SIMM with four DSPs"
PID: PVDM2-64 , VID: V01 , SN: FOC12295SS5

NAME: "PVDMII DSP SIMM with four DSPs on Slot 0 SubSlot 5", DESCR: "PVDMII DSP SIMM with four DSPs"
PID: PVDM2-64 , VID: V01 , SN: FOC12295SS9

NAME: "PVDMII DSP SIMM with four DSPs on Slot 0 SubSlot 6", DESCR: "PVDMII DSP SIMM with four DSPs"
PID: PVDM2-64 , VID: V01 , SN: FOC12135BST

NAME: "PVDMII DSP SIMM with four DSPs on Slot 0 SubSlot 7", DESCR: "PVDMII DSP SIMM with four DSPs"
PID: PVDM2-64 , VID: V01 , SN: FOC12295SXG

c3825-VXML-GW#
Configuring Cisco Unified SIP Proxy (CUSP) module

cvp-cusp> sh version

cvp-cusp uptime is 0 weeks, 0 days, 0 hours, 18 minutes
CPU Model: Intel(R) Pentium(R) M processor 1.40GHz
CPU Speed (MHz): 1400.263
CPU Cache (KByte): 2048
BogoMIPS: 2768.89
SKU: NME-APPRE-522-K9
Chassis Type: C3825
Chassis Serial: FTX1319A1UN
Module Type: NME
Module Serial: FOC13193AFV
UDI Name: Not Available
UDI Description: Not Available
SATA Drive: 160.0GB
SDRAM (MByte): 2048

cvp-cusp>
cvp-cusp>
cvp-cusp>
cvp-cusp> sh software versions

Cisco Unified SIP Proxy version (1.1.3.19)

cvp-cusp>
cvp-cusp>
cvp-cusp>
cvp-cusp>
cvp-cusp> show configuration active
Building CUSP configuration...
!
server-group sip global-load-balance call-id
server-group sip retry-after 0
server-group sip element-retries udp 2
server-group sip element-retries tls 1
server-group sip element-retries tcp 1
sip dns-srv
enable
no naptr
end dns
!
no sip header-compaction
no sip logging
!
sip max-forwards 70
sip network netA nocmp
non-invite-provisional 290
allow-connections
retransmit-count invite-client-transaction 2
retransmit-count invite-server-transaction 9
retransmit-count non-invite-client-transaction 9
retransmit-timer T1 500
retransmit-timer T2 4000
retransmit-timer T4 5000
retransmit-timer TU1 5000
retransmit-timer TU2 32000
retransmit-timer clientTn 64000
retransmit-timer serverTn 64000
udp max-datagram-size 1500
end network
!
sip overload reject retry-after 0
!
no sip peg-counting
!
sip privacy service
sip queue message
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue radius
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue request
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue response
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue st-callback
drop-policy head
low-threshold 80
size 2000
thread-count 10
end queue
!
sip queue timer
drop-policy none
low-threshold 80
size 2500
thread-count 8
end queue
!
sip queue xcl
drop-policy head
low-threshold 80
size 2000
thread-count 2
end queue
!
no route recursion
!
sip tcp connection-timeout 30
sip tcp max-connections 256
!
no sip tls
!
trigger condition in-netA
 sequence 1
 in-network netA
 end sequence
 end trigger condition
!
trigger condition mid-dialog
 sequence 1
 mid-dialog
 end sequence
 end trigger condition
!
trigger condition out-netA
 sequence 1
 out-network netA
 end sequence
 end trigger condition
!
accounting
 no enable
 no client-side
 no server-side
 end accounting
!
server-group sip group cm-cvp.cvplab.com netA
 element ip-address 172.20.224.254 5060 udp q-value 1.0 weight 10
 failover-resp-codes 503
 lbtype global
 ping
 end server-group
!
server-group sip group cvp-server.cvplab.com netA
 element ip-address 172.20.224.252 5060 udp q-value 1.0 weight 10
 failover-resp-codes 503
 lbtype global
 ping
 end server-group
!
server-group sip group vxml-gw.cvplab.com netA
 element ip-address 172.20.224.5 5060 udp q-value 1.0 weight 10
 failover-resp-codes 503
 lbtype global
 ping
 end server-group
!
route table cvp-route-table
 key 1111 target-destination vxml-gw.cvplab.com netA

In order to get Redirect 302 header to be sent upstream to the network side and not get processed in CUSP, the CUSP SIP proxy service ‘recursive’ needs to be turned off using CLI command “no route recursion” (defaults to On).
key 2222 target-destination cvp-server.cvplab.com netA
key 50 target-destination cvp-server.cvplab.com netA
key 5000 target-destination cvp-server.cvplab.com netA
key 543 target-destination cm-cvp.cvplab.com netA
key 9 target-destination vxml-gw.cvplab.com netA
end route table
!
policy lookup cvp-policy
  sequence 1 cvp-route-table request-uri uri-component user
    rule prefix
  end sequence
  end policy
!
trigger routing sequence 1 by-pass condition mid-dialog
trigger routing sequence 10 policy cvp-policy condition in-netA
!
server-group sip ping-options netA 172.20.224.249 5038
  method OPTIONS
  ping-type adaptive 5000 10000
  timeout 500
  end ping
!
server-group sip global-ping
  sip record-route netA udp 172.20.224.249 5060
  sip listen netA udp 172.20.224.249 5060
!
end
cvp-cusp(cusp)>
Configuring the Cisco Unified Communications Manager Release 8.0.3
Software Version—1 of 1
## SIP Profile

**Standard SIP Profile**

### SIP Profile Configuration

#### Status
- Status: Ready

All SIP devices using this profile must be restarted before any changes will take effect.

#### SIP Profile Information

<table>
<thead>
<tr>
<th>Name*</th>
<th>Description</th>
<th>Default SIP Profile</th>
<th>Default MTP Telephony Event Payload Type*</th>
<th>Resource Priority Namespace List</th>
<th>Early Offer for G.Clear Calls*</th>
<th>Redirect by Application</th>
<th>Disable Early Media on 180</th>
<th>Outgoing T.38 INVITE include audio online</th>
<th>Enable ANAT</th>
<th>Require SDP Inactive Exchange for Mid-Call Media Change</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Standard SIP Profile</td>
<td>Default SIP Profile</td>
<td>101</td>
<td>&lt; None &gt;</td>
<td>Disabled</td>
<td></td>
<td></td>
<td></td>
<td></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 TL (msec)*</td>
<td>500</td>
</tr>
<tr>
<td>Timer T2 (msec)*</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE*</td>
<td>3</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>92766</td>
</tr>
<tr>
<td>Call Pickup URI*</td>
<td>x-cisco-service-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI*</td>
<td>x-cisco-service-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI*</td>
<td>x-cisco-service-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI*</td>
<td>x-cisco-service-meetme</td>
</tr>
</tbody>
</table>
Standard SIP Profile—2 of 2

User Info

- None

DTMF DB Level

- Nominal

Call Hold Ring Back

- Off

Anonymous Call Block

- Off

Caller ID Blocking

- Off

Do Not Disturb Control

- User

Telnet Level for 7940 and 7960

- Disabled

Timer Keep Alive Expires (seconds)

- 120

Timer Subscribe Expires (seconds)

- 120

Timer Subscribe Delta (seconds)

- 5

Maximum Redirections

- 70

Off Hook To First Digit Timer (milliseconds)

- 15000

Call Forward URI

- cisco-serviceuri-cfnall

Abbreviated Dial URI

- cisco-serviceuri-abbrdial

Conference Join Enabled

☑

RFC 2543 Hold

☐

Semi Attended Transfer

☑

Enable VAD

☐

Stutter Message Waiting

☐

- Trunk Specific Configuration

Remote Incoming Request to new Trunk based on

- Never

RSVP Over SIP

☑

Local RSVP

☐

Fall back to local RSVP

SIP Rel1XX Options

- Disabled

- Copy  Reset  Apply Config  Add New
### SIP Trunk Security Profile Information

**Name**: UDP Non Secure SIP Trunk Profile  
**Description**: Non Secure SIP Trunk Profile authenticated by null String  
**Device Security Mode**: Non Secure  
**Incoming Transport Type**: TCP+UDP  
**Outgoing Transport Type**: UDP  
**Enable Digest Authentication**:  
**Nonce/Validity Time (mins)**: 500  
**X.509 Subject Name**:  
**Incoming Port**: 5060  

- **Enable Application Level Authentication**  
- **Accept Presence Subscription**  
- **Accept Out-of-Dialing REFER**  
- **Accept Unsolicited Notification**  
- **Accept Replaces Header**  
- **Transmit Security Status**
### Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>CVP_CallServer_SIP_Trunck</td>
</tr>
<tr>
<td>Description</td>
<td>CVP trunk via SIP Proxy CUSP</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL-HW</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_Name</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>

- **Media Termination Point Required**
- **Retry Video Call as Audio**
- **Transmit UTF-8 for Calling Party Name**
- **Unattended Port**
- **SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.**
- **Route Class Signalling Enabled**
- **Use Trusted Relay Point**
- **PSTN Access**

### Intercompany Media Engine (IME)

<table>
<thead>
<tr>
<th>IME E.164 Transformation Profile</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
</table>
### Multilevel Precedence and Preemption (MLPP) Information

- **MLPP Domain**: <None>

### Call Routing Information

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

### Inbound Calls

- **Significant Digits**: All 
- **Connected Line ID Presentation**: Default 
- **Connected Name Presentation**: Default 
- **Calling Search Space**: <None> 
- **AAA Calling Search Space**: <None> 
- **Prefix DN**: 

- **Redirecting Diversion Header Delivery - Inbound**: 

### 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>

### 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**: 
- **Calling Line ID Presentation**: Default 
- **Calling Name Presentation**: Default 
- **Caller ID DN**: 
- **Caller Name**: 

- **Redirecting Diversion Header Delivery - Outbound**: 

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### SIP Information

<table>
<thead>
<tr>
<th><strong>Destination Address</strong></th>
<th>172.20.224.249</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Destination Address IPv6</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Destination Port</strong></td>
<td>5660</td>
</tr>
<tr>
<td><strong>NTP Preferred Originating Codec</strong></td>
<td>711ulew</td>
</tr>
<tr>
<td><strong>Presence Group</strong></td>
<td>Standard Presence group</td>
</tr>
<tr>
<td><strong>SIP Trunk Security Profile</strong></td>
<td>UDP Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td><strong>Redirect Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>OUT-OF-DIALOG REFER Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>SUBSCRIBE Calling Search Space</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>SIP Profile</strong></td>
<td>Standard SIP Profile</td>
</tr>
<tr>
<td><strong>Diameter Signaling Method</strong></td>
<td>RFC 2833</td>
</tr>
</tbody>
</table>

### Geolocation Configuration

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

- Indicates required item.

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

Route Pattern 2222—1 of 2

Route Pattern Configuration

- Status
  Status: Ready

- Pattern Definition
  Route Pattern: 2222
  Route Partition: < None >
  Description: CCM RC VRU DN
  Numbering Plan: < Not Selected >
  Route Filter: < None >
  MLPP Precedence: Default
  Resource Priority Namespace Network Domain: < None >
  Route Class: Default
  Gateway/Route List: CVP_CallServer_SIP_Trunk
  Route Option:
    - Route this pattern
    - Block this pattern
    - No Error
  Call Classification: Offnet
  Allow Device Override
  Provide Outside Dial Tone
  Allow Overlap Sending
  Urgent Priority
  Require Forced Authorization Code
  Authorization Level: 0
  Require Client Matter Code
### Route Pattern 2222—2 of 2

#### 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

#### Connected Party Transformations

- **Connected Line ID Presentation**
  - Default
- **Connected Name Presentation**
  - Default

#### Called Party Transformations

- **Discard Digits**
  - < None >
- **Called Party Transform Mask**
  - [ ]
- **Prefix Digits (Outgoing Calls)**
  - [ ]
- **Called Party Number Type**
  - Cisco CallManager
- **Called Party Numbering Plan**
  - Cisco CallManager

#### ISDN Network-Specific Facilities Information Element

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

* Indicates required item.

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Page 39 of 155
### Find and List CTI Route Points

1 records found

#### CTI Route Point (1 of 1)

<table>
<thead>
<tr>
<th>Device Name</th>
<th>Description</th>
<th>Device Pool</th>
<th>Calling Search Space</th>
<th>Partition</th>
<th>Extension</th>
<th>Status</th>
<th>IP Address</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>5000</td>
<td>Post Route DN</td>
<td>Default</td>
<td></td>
<td>5000</td>
<td></td>
<td>Registered with CM-CUP</td>
<td>172.20.224.263</td>
<td></td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected | Reset Selected | Apply Config to Selected |
### Device Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager CM-CVP</td>
</tr>
<tr>
<td>IP Address</td>
<td>172.20.224.253</td>
</tr>
<tr>
<td>Device Name*</td>
<td>6000</td>
</tr>
<tr>
<td>Description</td>
<td>Post Route DN</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_None</td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL-HW</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>1-SampleAudioSource</td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Association Information

- Line [11 - 6000 (no partition)]
## Directory Number Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory Number</td>
<td>6000</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Alerting Name</td>
<td></td>
</tr>
<tr>
<td>ASCII Alerting Name</td>
<td></td>
</tr>
<tr>
<td>Associated Devices</td>
<td>6000</td>
</tr>
</tbody>
</table>

### Directory Number Settings

- **Voice Mail Profile**: < None >
- **Calling Search Space**: < None >
- **Presence Group**: < None >
- **User Hold MOH Audio Source**: < None >
- **Network Hold MOH Audio Source**: < None >

(Choose <None> to use system default)
<table>
<thead>
<tr>
<th>Page 45 of 155</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Cisco Unified IP Phone 7961, x542000, SCCP—2 of 6</strong></td>
</tr>
<tr>
<td><strong>User Hold MOH Audio Source</strong></td>
</tr>
<tr>
<td><strong>Network Hold MOH Audio Source Location</strong></td>
</tr>
<tr>
<td><strong>AAR Group</strong></td>
</tr>
<tr>
<td><strong>User Locale</strong></td>
</tr>
<tr>
<td><strong>Network Locale</strong></td>
</tr>
<tr>
<td><strong>Built In Bridge</strong></td>
</tr>
<tr>
<td><strong>Privacy</strong></td>
</tr>
<tr>
<td><strong>Device Mobility Mode</strong></td>
</tr>
<tr>
<td><strong>OWNER User ID</strong></td>
</tr>
<tr>
<td><strong>Phone Personalization</strong></td>
</tr>
<tr>
<td><strong>Services Provisioning</strong></td>
</tr>
<tr>
<td><strong>Phone Load Name</strong></td>
</tr>
<tr>
<td><strong>Single Button Barge</strong></td>
</tr>
<tr>
<td><strong>Join Across Lines</strong></td>
</tr>
<tr>
<td><strong>Use Trusted Relay Point</strong></td>
</tr>
<tr>
<td><strong>BLF Audible Alert Setting (Phone Idle)</strong></td>
</tr>
<tr>
<td><strong>BLF Audible Alert Setting (Phone Busy)</strong></td>
</tr>
<tr>
<td><strong>Always Use Prime Line</strong></td>
</tr>
<tr>
<td><strong>Always Use Prime Line for Voice Message</strong></td>
</tr>
<tr>
<td><strong>Calling Party Transformation CSS</strong></td>
</tr>
<tr>
<td><strong>1-SampleAudioSource</strong></td>
</tr>
<tr>
<td><strong>1-SampleAudioSource</strong></td>
</tr>
<tr>
<td><strong>Hub_None</strong></td>
</tr>
<tr>
<td><strong>&lt; None &gt;</strong></td>
</tr>
<tr>
<td><strong>&lt; None &gt;</strong></td>
</tr>
<tr>
<td><strong>&lt; None &gt;</strong></td>
</tr>
<tr>
<td><strong>Default</strong></td>
</tr>
<tr>
<td><strong>Default</strong></td>
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<tr>
<td><strong>Default</strong></td>
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<td><strong>Default</strong></td>
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<td><strong>Default</strong></td>
</tr>
<tr>
<td><strong>Default</strong></td>
</tr>
<tr>
<td><strong>Default</strong></td>
</tr>
<tr>
<td><strong>&lt; None &gt;</strong></td>
</tr>
<tr>
<td><strong>View</strong></td>
</tr>
</tbody>
</table>

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Geolocation: <None>
- Use Device Pool Calling Party Transformation CSS
- Retry Video Call as Audio
- Ignore Presentation Indicators (internal calls only)
- Allow Control of Device from CTI
- Logged into Hunt Group
- Remote Device
- Protected Device
- Hot line Device

### Protocol Specific Information

- **Packet Capture Mode**: None
- **Packet Capture Duration**: 0
- **Presence Group**: Standard Presence Group
- **Device Security Profile**: Cisco 7961 - Standard SCCP Non-Secure Profile
- **SUBSCRIBE Calling Search Space**: <None>
- **Unattended Port**: 
- **Require DTMF Reception**: 
- **RFC2833 Disabled**: 

### Certification Authority Proxy Function (CAPF) Information

- **Certificate Operation**: No Pending Operation
- **Authentication Mode**: By Null String
- **Authentication String**: 
- **Generate String**: 
- **Key Size (Bits)**: 1024
- **Operation Completes By**: 2010/11/12
- **Certificate Operation Status**: None

Note: Security Profile Contains Additional CAPF Settings.

### Expansion Module Information

- **Module 1**: <None>
- **Module 1 Load Name**: 
- **Module 2**: <None>
- **Module 2 Load Name**: 
### External Data Locations Information

<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 Information

- MLPP Domain: < None >
- MLPP Indication: Default
- MLPP Preemption: Default

### Do Not Disturb

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

### Secure Shell Information

- Secure Shell User: cisco
- Secure Shell Password: ***********************
<table>
<thead>
<tr>
<th>Product Specific Configuration Layout</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>□ Disable Speakerphone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>□ Disable Speakerphone and Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forwarding Delay</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Settings Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Video Capabilities</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Auto Line Select</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Web Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Logging Display</td>
<td>PC Controlled</td>
<td></td>
</tr>
<tr>
<td>Load Server</td>
<td>PC Controlled</td>
<td></td>
</tr>
<tr>
<td>Recording Tone</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td>100</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td>50</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Cisco Unified IP Phone 7961, x5432000, SCCP—6 of 6

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Advanced G.722 Codec*</td>
<td>Use System Default</td>
</tr>
<tr>
<td>W-Band Headset UI Control*</td>
<td>Enabled</td>
</tr>
<tr>
<td>W-Band Headset UI Control*</td>
<td>Enabled</td>
</tr>
<tr>
<td>W-Band Headset*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peer Firmware Sharing*</td>
<td>Disabled</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 Discover (LLDP-MED): Switch Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td>Unknown</td>
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<tr>
<td>LLDP Power Priority*</td>
<td>Normal</td>
</tr>
<tr>
<td>Display Refresh Rate*</td>
<td>Normal</td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td>User Controlled</td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td>Normal</td>
</tr>
<tr>
<td>802.1x Authentication*</td>
<td>Normal</td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure*</td>
<td>Normal</td>
</tr>
<tr>
<td>Minimum Ring Volume*</td>
<td>0-Silent</td>
</tr>
<tr>
<td>Headset Sidetone Level*</td>
<td>Use Phone Default</td>
</tr>
<tr>
<td>HTTPS Server*</td>
<td>http and https Enabled</td>
</tr>
<tr>
<td>Enable Dailing*</td>
<td>Enabled</td>
</tr>
</tbody>
</table>

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

**Status**

- Status: Ready

---

**Directory Number Information**

- **Directory Number**: 5432000
- **Route Partition**: < None >
- **Description**: 
- **Alerting Name**: 
- **ASCII Alerting Name**: 

- **Allow Control of Device from CTI**: 

- **Associated Devices**: SEP0023331B4C05

- **Dissociate Devices**: 

---

**Directory Number Settings**

- **Voice Mail Profile**: < None > (Choose < None > to use system default)
- **Calling Search Space**: < None >
- **Presence Group**: Standard Presence group
- **User Hold MOH Audio Source**: < None >
- **Network Hold MOH Audio Source**: < None >
- **Auto Answer**: Auto Answer Off
### AAR Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
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</tbody>
</table>

- Retain this destination in the call forwarding history

### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td>Use System Default</td>
</tr>
<tr>
<td>Forward All</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>External</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>External</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>
### Line DN 5432000—3 of 5

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Forward No Coverage External</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward on CTI Failure</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**No Answer Ring Duration (seconds)**

- **Call Pickup Group:** < None >

---

### Park Monitoring

<table>
<thead>
<tr>
<th>Feature</th>
<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>&lt; None &gt;</td>
<td>A</td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>A</td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>A</td>
</tr>
<tr>
<td>Park Monitoring</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>A</td>
</tr>
</tbody>
</table>

*Blank value means to call the parker's line.*

**Timer service parameter**

---

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

**Multiple Call/Call Waiting Settings on Device SEP0023331B4C05**

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 SEP0023331B4C05**

- ☑ Caller Name
- ☐ Caller Number
- ☐ Redirected Number
- ☑ Dialed Number

**Users Associated with Line**

- Associate End Users

---

* - indicates required item.

** - Changes to Line or Directory Number settings require restart.
### Association Information

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

#### Phone Type
- **Product Type:** Cisco 7961
- **Device Protocol:** SIP

#### Device Information
- **Registration:** Registered with Cisco Unified Communications Manager CM-CVP
- **IP Address:** 172.20.624.109
- **Active Load ID:** SIP4.9-1-1LTH1:13S
- **Device in:** Active
- **Device is trusted:**
- **MAC Address:** 002331B496D
- **Description:** SIP 5432003
- **Device Pool:** Default
- **Common Device Configuration:**
- **Phone Button Template:** Standard 7961 SIP
- **Softkey Template:**
- **Common Phone Profile:** Standard Common Phone Profile
- **Calling Search Space:**
- **AAR Calling Search Space:**
- **Media Resource Group List:** MERGL-HW
- **User Hold NOH Audio Source:** 1-sampleAudioSource
- **Network Hold NOH Audio Source Location:**
- **AAR Group:** Hub_None
- **User Locale:**
- **Network Locale:**

### Modified Button Items

<table>
<thead>
<tr>
<th>Button Item</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line 11 - 5432003 (no partition)</td>
</tr>
<tr>
<td>Line 12 - Add a new DN</td>
</tr>
<tr>
<td>Add a new SD</td>
</tr>
<tr>
<td>Add a new SD</td>
</tr>
<tr>
<td>Add a new SD</td>
</tr>
<tr>
<td>Add a new SD</td>
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<tr>
<td>Add a new SD</td>
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<tr>
<td>Add a new SD</td>
</tr>
<tr>
<td>Add a new SD</td>
</tr>
<tr>
<td>Add a new SD</td>
</tr>
<tr>
<td>Intercom 11 - Add a new Intercom</td>
</tr>
<tr>
<td>Do Not Disturb</td>
</tr>
<tr>
<td>Call Park</td>
</tr>
<tr>
<td>Call Pickup</td>
</tr>
<tr>
<td>CallBack</td>
</tr>
<tr>
<td>Conference List</td>
</tr>
<tr>
<td>Conference</td>
</tr>
<tr>
<td>End Call</td>
</tr>
<tr>
<td>Forward All</td>
</tr>
<tr>
<td>Group Call Pickup</td>
</tr>
<tr>
<td>Hold</td>
</tr>
<tr>
<td>Hunt Group Logout</td>
</tr>
<tr>
<td>Malicious Call Identification</td>
</tr>
<tr>
<td>Meet Me Conference</td>
</tr>
<tr>
<td>Mobility</td>
</tr>
<tr>
<td>New Call</td>
</tr>
<tr>
<td>26</td>
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<td>----</td>
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<td>27</td>
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<td>30</td>
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<td>31</td>
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<td>32</td>
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<td>33</td>
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</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Built In Bridge</td>
<td>Default</td>
</tr>
<tr>
<td>Privacy*</td>
<td>Default</td>
</tr>
<tr>
<td>Device Mobility Mode*</td>
<td>Default</td>
</tr>
<tr>
<td>Owner User ID</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Phone Personalization*</td>
<td>Default</td>
</tr>
<tr>
<td>Services Provisioning*</td>
<td>Default</td>
</tr>
<tr>
<td>Phone Load Name</td>
<td></td>
</tr>
<tr>
<td>Single Button Barge</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines</td>
<td>Default</td>
</tr>
<tr>
<td>Use Trusted Relay Point*</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Idle)*</td>
<td>Default</td>
</tr>
<tr>
<td>BLF Audible Alert Setting (Phone Busy)*</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line*</td>
<td>Default</td>
</tr>
<tr>
<td>Always Use Prime Line for Voice Messages*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- Use Device Pool Calling Party Transformation CSS
- Ignore Presentation Indicators (internal calls only)
- Allow Control of Device from CTI
- Logged Into Hunt Group
- Remote Device
- Prohibited Device
- Hotline Device
### Protocol Specific Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Presence Group</td>
<td>Standard Presence group</td>
</tr>
<tr>
<td>SIP Dial Rules</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MTP Preferred Originating Codec</td>
<td>711Low</td>
</tr>
<tr>
<td>Device Security Profile</td>
<td>Cisco 7961 - Standard SIP Non-Secure Profile</td>
</tr>
<tr>
<td>Recurring Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SUBSCRIBE Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>Standard SIP Profile</td>
</tr>
<tr>
<td>Digest User</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Termination Point Required</td>
<td>[ ]</td>
</tr>
<tr>
<td>Unattended Port</td>
<td>[ ]</td>
</tr>
<tr>
<td>Require DTMF Reception</td>
<td>[ ]</td>
</tr>
</tbody>
</table>

### Certification Authority Proxy Function (CAPF) Information

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Certificate Operation</td>
<td>No Pending Operation</td>
</tr>
<tr>
<td>Authentication Mode</td>
<td>By Null String</td>
</tr>
<tr>
<td>Authorization String</td>
<td>Generate String</td>
</tr>
<tr>
<td>Key Size (bits)</td>
<td>1024</td>
</tr>
<tr>
<td>Operation Completes By</td>
<td>2010-11-19 12 (YYYY-MM-DD HH)</td>
</tr>
<tr>
<td>Certificate Operation Status</td>
<td>None</td>
</tr>
</tbody>
</table>

Note: Security Profile Contains Additional CAPF Settings.
### Expansion Module Information

<table>
<thead>
<tr>
<th>Module 1</th>
<th>&lt;None&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module 1 Load Name</td>
<td></td>
</tr>
<tr>
<td>Module 2</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Module 2 Load Name</td>
<td></td>
</tr>
</tbody>
</table>

### 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 Information

| MLPP Domain | <None> |

### Do Not Disturb

- Do Not Disturb
- DND Option [Ringer Off]
- DND Incoming Call Alert [<None>]
### Secure Shell Information

<table>
<thead>
<tr>
<th>Param</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Shell User</td>
<td></td>
</tr>
<tr>
<td>Secure Shell Password</td>
<td></td>
</tr>
</tbody>
</table>

### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Param</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Settings Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Gratuitous ARP</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>PC Voice VLAN Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Auto Line Select</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Web Access</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Logging Display</td>
<td>PC Controlled</td>
<td></td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Recording Tone</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td>100</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td>50</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>5</td>
<td></td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Feature</td>
<td>Setting</td>
<td></td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>--------------------</td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codes†</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset UI Control†</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Wideband Handset UI Control†</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset†</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Wideband Handset†</td>
<td>Use Phone Default</td>
<td></td>
</tr>
<tr>
<td>Peer Firmware Sharing†</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port†</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port†</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port†</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port†</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority†</td>
<td>Unknown</td>
<td></td>
</tr>
<tr>
<td>Display Refresh Rate†</td>
<td>Normal</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Feature</td>
<td>Setting</td>
<td></td>
</tr>
<tr>
<td>-------------------------</td>
<td>------------------</td>
<td></td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td>User Controlled</td>
<td></td>
</tr>
<tr>
<td>Detect Unified CM</td>
<td>Normal</td>
<td></td>
</tr>
<tr>
<td>CM Connection Failure</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
<td>O-Silent</td>
<td></td>
</tr>
<tr>
<td>Headset Tone Level</td>
<td>Use Phone Default</td>
<td></td>
</tr>
<tr>
<td>HTTPS Server</td>
<td>Http and https Enabled</td>
<td></td>
</tr>
<tr>
<td>Enblc Dialog</td>
<td>Enabled</td>
<td></td>
</tr>
</tbody>
</table>

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

Directory Number Information

- Directory Number: 5432003
- Route Portion: < None >
- Description: < None >
- Alerting Name: 
- ASCII Alerting Name: 

- Allow Control of Device from CTI
- Associated Devices: SEP0023332BC98ED

Directory Number Settings

- Voice Mail Profile: < None >
- Calling Search Space: < None >
- Presence Group*: Standard Presence Group
- User Hold MOC Audio Source: < None >
- Network Hold MOC Audio Source: < None >
- Auto Answer*: Auto Answer Off

Status

Status: Ready

Edit Device
Edit Line Appearance
### AAR Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
<th>AAR Group</th>
</tr>
</thead>
<tbody>
<tr>
<td>AAR</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- Retain this destination in the call forwarding history

### Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>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>&lt; None &gt;</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy Internal</td>
<td></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>
<tr>
<td>Forward on CTI Failure</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td></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>or</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Retrieve Destination External</td>
<td>or</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

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

**Parked Monitoring Reversion Timer**

Timer service parameter

### NMPP Alternate Party Settings

<table>
<thead>
<tr>
<th>Target (Destination)</th>
<th>NMPP Calling Search Space</th>
<th>NMPP No Answer Ring Duration (seconds)</th>
</tr>
</thead>
</table>

### Line Settings for All Devices

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

### Line 1 on Device SEP023331B498D

- **Display (Internal Caller ID)**
  - Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

- **ASCII Display (Internal Caller ID)**
- **Line Text Label**
- **ASCII Line Text Label**
**Line DN 54532003—4 of 5**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>External Phone Number Mask</td>
<td></td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy</td>
<td>Use System Policy</td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy</td>
<td>Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Idle) *</td>
<td>Ring</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting</td>
<td>Use System Default</td>
</tr>
<tr>
<td>(Phone Idle)</td>
<td></td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting</td>
<td>Use System Default</td>
</tr>
<tr>
<td>(Phone Active)</td>
<td></td>
</tr>
<tr>
<td>Recording Option *</td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Log Missed Calls</td>
<td></td>
</tr>
</tbody>
</table>

---

**Multiple Call/Call Waiting Settings on Device SEP0023331B498D**

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

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

(Less than or equal to Max. Calls)

---

**Forwarded Call Information Display on Device SEP0023331B498D**

- [x] Caller Name
- [ ] Caller Number
- [ ] Redirected Number
- [x] Dialed Number
### Users Associated with Line

<table>
<thead>
<tr>
<th>Associate End Users</th>
</tr>
</thead>
</table>

---

**i**  *- indicates required item.*

**ii**  Changes to Line or Directory Number settings require restart.
**Region Information**

<table>
<thead>
<tr>
<th>Name</th>
<th>Default</th>
</tr>
</thead>
</table>

**Region Relationships**

<table>
<thead>
<tr>
<th>Region</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>64 kbps (G.722, G.721)</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>G729 Region</td>
<td>8 kbps (G.729)</td>
<td>304</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

NOTE: Regions(s) not displayed | Use System Default | Use System Default | Use System Default |

**Notify Relationship to other Regions**

<table>
<thead>
<tr>
<th>Regions</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>G729 Region</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

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Page 67 of 155
### Region Information

<table>
<thead>
<tr>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.729 Region</td>
</tr>
</tbody>
</table>

### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>8 kbps (G.729)</td>
<td>304</td>
<td>Use System Default</td>
</tr>
<tr>
<td>G.729 Region</td>
<td>6 kbps (G.729)</td>
<td>304</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

### Notify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Max Audio Bit Rate</th>
<th>Max Video Call Bit Rate (Includes Audio)</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td>G.729 Region</td>
<td>Keep Current Setting</td>
<td>Use System Default</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>Keep Current Setting</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>None</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
<td>0 kbps</td>
<td></td>
</tr>
</tbody>
</table>
### Device Pool Configuration

**Status**
- Status: Ready

**Device Pool Information**
- Device Pool: G729_DP (4 members***)

**Device Pool Settings**
- Device Pool Name*: G729_DP
- Cisco Unified Communications Manager Group*: Default
- Calling Search Space for Auto-registration: < None >
- Adjunct CSS: < None >
- Reverted Call Focus Priority: Default
- Local Route Group: < None >
- Intercompany Media Services Enrolled Group: < None >

**Roaming Sensitive Settings**
- Data/Time Group*: CNLocal
- Region*: G729 Region
- Media Resource Group List: < None >
- Location: < None >
- Network Locale: < None >
- SRST Reference*: Disable
- Connection Monitor Duration**: ***
- Single Button Barge*: Default
- Join Across Lines*: Default
- Physical Location: < None >
- Device Mobility Group: < None >
G.729 Device Pool—2 of 2

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

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

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

Connected Party Settings
Connected Party Transformation CSS: < None >

---

Save  Delete  Copy  Reset  Apply Config  Add New

---

* Indicates required item.
** Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.
*** Leave the field blank or enter -1 to use the configuration from the enterprise parameter.
**** These five parameters will overwrite device level settings when device is roaming and in the same device mobility group.
MRG—1 of 1

Cisco Unified CM Administration
For Unified Communications Solutions

Media Resource Group Configuration

Status
Status: Ready

Media Resource Group Status
Media Resource Group: MRG-CM-CVP (used by 0 devices)

Media Resource Group Information
Name: MRG-CM-CVP
Description: MRG-CM-CVP

Devices for this Group
Available Media Resources:
- 001A5CD006911
- CFB001A5D006911
- MTP001A5D006911
- MTP1C0D0069E00

Selected Media Resources:
- AMB2 (AMB)
- CFB2 (CFB)
- MOH2 (MOH)
- MTP2 (MTP)

Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

Save | Delete | Copy | Add New
### MRG-HW Configuration

**Status:** Ready

#### Media Resource Group Status

Media Resource Group: MRG-HW (used by 11 devices)

#### Name

MRG-HW

#### Description


#### Devices for this Group

**Available Media Resources:**

- MN1_2
- CPD_2
- MTP1C0F0C9E800
- MTP_2

**Selected Media Resources:**

- 001A5D1006911 (MTP)
- CPB001A6D106911 (CPB)
- MOH_2 (MOH)
- MTP1A5D006911 (XCODE)

Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

---

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### Media Termination Point Information

<table>
<thead>
<tr>
<th>Registration</th>
<th>Registered with Cisco Unified Communications Manager CM-CVP</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address</td>
<td>172.20.224.5</td>
</tr>
<tr>
<td>IPv6 Address</td>
<td>0000:0000:0000:0000:0000:0000:0000:0000:0000:0000:0000:0000</td>
</tr>
<tr>
<td>Media Termination Point Type</td>
<td>Cisco IOS Enhanced Software Media Termination Point</td>
</tr>
<tr>
<td>Media Termination Point Name</td>
<td>001A60006911</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Device Pool</td>
<td>Default</td>
</tr>
</tbody>
</table>

- **Status**: Ready
Transcoder Configuration

Transcoder Information

- Transcoder: NTP001AGD0806911 (NTP001AGD0806911)
- Registration: Registered with Cisco Unified Communications Manager CM-CVP
- IP Address: 172.20.228.5
- IPv6 Address: 000::0000:0000:0000:0000:0000:0000

Media Termination Point Hardware Info

- Transcoder Type*: Cisco Media Termination Point Hardware
- Description: MTP001AGD0806911
- MAC Address*: 001AGD0806911
- Device Pool*: Default
- Common Device Configuration: < None >
- Special Load Information: Leave blank to use default

Save | Delete | Copy | Reset | Apply Config | Add New

- Indicates required item.
### Application User Configuration

**Status**
- Status: Ready

**Application User Information**
- **User ID**: pguser
- **Password**: ************
- **Confirm Password**: ************
- **Digest Credentials**: 
- **Confirm Digest Credentials**: 
- **Presence Group**: Standard Presence group
- **Accept Presence Subscription**: 
- **Accept OUT-of-dialog REFER**: 
- **Accept Unsolicited Notification**: 
- **Accept Replaces Header**: 

**Device Information**
- **Available Devices**
- **Controlled Devices**
  - 5000
  - SEP0023313B436D
  - SEP0023313B4A0E
  - SEP0023313B4ABB
  - SEP0023313B4C0E

**Available Profiles**

---

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Page 76 of 155
### CTI Controlled Device Profiles

```

```

### CAPF Information

**Associated CAPF Profiles**

```

```

### Permissions Information

#### Groups

- Standard CTI Allow Control of All Devices
- Standard CTI Enabled

```

#### Roles

- Standard CTI Allow Control of All Devices
- Standard CTI Enabled

```

- **Add to User Group**
- **Remove from User Group**

- **Save**  **Delete**  **Copy**  **Add New**

> *: indicates required item.
Configuring the Cisco Unified Customer Voice Portal Release 8.0(1)

Cisco Unified Customer Voice Portal

Version: 8.0(1)
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<table>
<thead>
<tr>
<th>Hostname</th>
<th>IP Address</th>
<th>Device Type</th>
<th>Last known version</th>
</tr>
</thead>
<tbody>
<tr>
<td>CVP</td>
<td>172.20.224.252</td>
<td>Unified CVP Call Server</td>
<td>CVP8.0(1) Build=1440</td>
</tr>
</tbody>
</table>
### CVP Device Pool—1 of 2

#### Cisco Unified Customer Voice Portal

**Device Type:** Device Pool

#### Control Center - Network Map

**Device Pool:** default

**General**

<table>
<thead>
<tr>
<th>Device</th>
<th>HostName</th>
<th>IP Address</th>
<th>Device Type</th>
<th>Actions</th>
<th>Status</th>
<th>Active Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>ASR-CUFF-GW</td>
<td>172.20.224.10</td>
<td>Gateway</td>
<td>N/A</td>
<td>N/A</td>
<td></td>
</tr>
<tr>
<td></td>
<td>C8825-CUBE</td>
<td>172.20.224.12</td>
<td>Gateway</td>
<td>N/A</td>
<td>N/A</td>
<td></td>
</tr>
<tr>
<td></td>
<td>CVP</td>
<td>172.20.224.352</td>
<td>Unified CVP Call Server</td>
<td>Up</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Refresh:** Every 30 Seconds

---

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### General Configuration

- **VRU Connection Port**: 5000
- **Maximum Length of DNIS**: 10

#### Translation Routed DNIS Pool

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Add DNIS</td>
<td></td>
</tr>
<tr>
<td>Delete DNIS</td>
<td></td>
</tr>
</tbody>
</table>

### Advanced Configuration

- **Required**:
  1. Change in value requires device restart.
  2. Conditionally required.
### List of Gateways

<table>
<thead>
<tr>
<th>Hostname</th>
<th>IP Address</th>
<th>Device State</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASR-CUBE-GW</td>
<td>172.20.224.10</td>
<td>Configured</td>
<td>ASR1002 CUBE(eh) VXML GW</td>
</tr>
<tr>
<td>CUBE-CUBE</td>
<td>172.20.224.5</td>
<td>Configured</td>
<td>CUBE and VXML GW</td>
</tr>
</tbody>
</table>
### General

- **IP Address**: 172.20.234.3
- **Hostname**: e1000-CUBE
- **Device Type**: 3000
- **Description**: Impress and VXML GW
- **Trunk Group ID**: 300
- **Location ID**: 
- **Enable secure communication with the IP office**

### Username and Passwords

- **Username**: cisco
- **User password**: ************
- **Enable password**: ************
- **Port**: 23

---

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Configuring the Cisco Unified Intelligent Contact Management Enterprise (ICM) Release 8.0.1.0

ICM Version—1 of 1

About Configuration Manager

Intelligent Contact Management - Configuration Manager

Release 8.0.1.0 , Build 26631

03/08/10 12:26:18

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OK
### ICM/CCE/CCH Service Control

**Computer Name:** CISCO-ICM

<table>
<thead>
<tr>
<th>Services</th>
<th>State</th>
<th>Startup</th>
</tr>
</thead>
<tbody>
<tr>
<td>Cisco CBU Support Tools NodeAgent</td>
<td>Running</td>
<td>Automatic</td>
</tr>
<tr>
<td>Cisco ICM cvpCG2A</td>
<td>Running</td>
<td>Automatic</td>
</tr>
<tr>
<td>Cisco ICM cvpCTIOS1</td>
<td>Running</td>
<td>Automatic</td>
</tr>
<tr>
<td>Cisco ICM cvpDistributor</td>
<td>Running</td>
<td>Automatic</td>
</tr>
<tr>
<td>Cisco ICM cvpLoggerA</td>
<td>Running</td>
<td>Automatic</td>
</tr>
<tr>
<td>Cisco ICM cvpPG1A</td>
<td>Running</td>
<td>Automatic</td>
</tr>
<tr>
<td>Cisco ICM cvpPG2A</td>
<td>Running</td>
<td>Automatic</td>
</tr>
<tr>
<td>Cisco ICM cvpRouterA</td>
<td>Running</td>
<td>Automatic</td>
</tr>
</tbody>
</table>
ICM Configuration Manager—2 of 3

[Image of the ICM Configuration Manager window with various tools listed, including Explorer Tools and List Tools, with an ICM Instance specified as cvp]
Network transfer preferred” check box need to be checked for both CCMPIM in the CCMPG and VRUPG in order to do IP Transfer Connect-SIP REFER Unattended.
Network transfer preferred” check box need to be checked for both CCMPIM in the CCMPG and VRUPG in order to do IP Transfer Connect-SIP REFER Unattended.
Network VRU Media invalid entry script — 1 of 1

![Network VRU Script List](image)

- **Name**: Play_Media_invalid_entry
- **Network VRU**: CXF_VRU_Type10
- **VRU script name**: PM_Hello_invalid_entry_encounterA
- **Timeout**: 180 seconds
- **Configuration param**: Y
- **Customer**: ovp
- **Introucitable**: Y
- **Divertable**: N
- **Description**: None

**ICM Instance**: ovp
ICM Call Type 5002—1 of 1

[Image of the Call Type List window with attributes and service level configuration]

- **Name**: DN_5002
- **Call Type ID**: 5002
- **Customer**: 011

**Service Level**
- **Service Level Threshold**: 20
- **Service Level Type**: Ignore Abandoned Calls

**Bucket Intervals**: Default Bucket Intervals

**Description**: Test number
## ICM DN 6000 Script Selector

### Dialoged Number / Script Selector List

<table>
<thead>
<tr>
<th>Attributes</th>
<th>Dialed Number Mapping</th>
<th>Dialed Number Label</th>
</tr>
</thead>
<tbody>
<tr>
<td>Routing client</td>
<td>Device ПГ, PIC</td>
<td></td>
</tr>
<tr>
<td>Media routing domain</td>
<td>Cisco_Voice</td>
<td></td>
</tr>
<tr>
<td>Dialed number string / Script selector</td>
<td>cnWood</td>
<td></td>
</tr>
<tr>
<td>Name</td>
<td>ICM_PГ_PIC_6000</td>
<td></td>
</tr>
<tr>
<td>Customer</td>
<td>cpv</td>
<td></td>
</tr>
<tr>
<td>Default label</td>
<td>(Name)</td>
<td></td>
</tr>
<tr>
<td>Description</td>
<td>Post Route DN</td>
<td></td>
</tr>
<tr>
<td>Permit application routing</td>
<td>✓</td>
<td></td>
</tr>
<tr>
<td>Reserved by IVR</td>
<td>✔</td>
<td></td>
</tr>
</tbody>
</table>

### Dialoged Number / Script Selector

<table>
<thead>
<tr>
<th>Name</th>
<th>MRU(ПГ, PIC) 6000</th>
</tr>
</thead>
<tbody>
<tr>
<td>MRU(ПГ, PIC) 5000</td>
<td></td>
</tr>
<tr>
<td>MRU(ПГ, PIC) 5001</td>
<td></td>
</tr>
<tr>
<td>MRU(ПГ, PIC) 5002</td>
<td></td>
</tr>
</tbody>
</table>
ICM Label list — 3 of 4

<table>
<thead>
<tr>
<th>Label</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1111111111</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543000.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543200.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543300.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543400.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543500.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543600.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543700.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543800.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
<tr>
<td>543900.CVP_VRU_Type10</td>
<td>CVP_VRU_Type10</td>
</tr>
</tbody>
</table>

**Attributes**
- **Routing client**: CCM_PG_RC
- **Label**: 2221222222
- **Label type**: Normal
- **Network Target**
  - **Target type (filter)**: Network VRU
  - **Network target**: CVP_VRU_Type10
- **Customer**: cvp
- **Description**:
### CTI Toolkit Agent Desktop (Win32) (8.0.11510) - 5432000 Phone @ UCCE

<table>
<thead>
<tr>
<th>CellID</th>
<th>CallStatus</th>
<th>CallType</th>
<th>DNIS</th>
<th>ANI</th>
<th>CFD</th>
<th>DialedNumber</th>
<th>Wrapup</th>
<th>Var1</th>
<th>Var2</th>
<th>Var3</th>
<th>Var4</th>
<th>Var5</th>
<th>Var6</th>
</tr>
</thead>
<tbody>
<tr>
<td>16768241</td>
<td>Talking</td>
<td>PPREROUTE_ADD_IN</td>
<td>5432000</td>
<td>4085771426</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

*Ext: 5432000 Inst: 5432000 Agent ID: 5432000*  
*Agent Status: Talking Connected To: cisco.com*  
*Online*
<table>
<thead>
<tr>
<th>CallID</th>
<th>CallStatus</th>
<th>CallType</th>
<th>DNIS</th>
<th>ANI</th>
<th>CED</th>
<th>DailedNumber</th>
<th>Wrapup</th>
<th>Var1</th>
<th>Var2</th>
<th>Var3</th>
<th>Var4</th>
<th>Var5</th>
<th>Var6</th>
<th>Var7</th>
</tr>
</thead>
<tbody>
<tr>
<td>16768240</td>
<td>Talking</td>
<td>PREREQTE_ADD_IN</td>
<td>5432001</td>
<td>Restricted</td>
<td>5001</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Ext: 5432001 Inst: 5432001 Agent ID: 5432001
Agent Status: Talking Connected To: cisco.com Online
CTIOS Agent Desktop Transfer Initiate procedure —1 of 2

Click here to initiate transfer

Use CTI Post Route DN (in this case DN 6000) in the "number to Dial" field instead of the actual agent number to do intra site transfer in order for the caller to hear ring back upon Blind transfer when agent 2 is ringing.

Click here for Attended transfer

Click here for Blind transfer

As an option, use the More tab to enter specific DN in "Var1" field to transfer to a specific target agent.
CTIOS Agent Desktop Transfer Initiate procedure — 2 of 2

Use “Var1” field for transfer to a specific target agent, then click on “Single Step” for Blind transfer or “Transf Init” for Attended transfer.
This script is used for queuing calls to Sales group if agents are not available.

Refer transfer via Cisco UBE requires that the Network transfer Enable variable be set to 1 in the initial call’s VRU script and also in the Refer label script.
This script is used for queuing calls to Marketing group if agents are not available.
This script is used for Caller Entered Digit Routing test cases.
This script is used for the Alternate Destination Routing (ADR) on RNA feature.
CTI Route Point DN 6000 should be used to map to this Post Route routing script for intra site transfers to marketing group.

Note: "send to VRU" node is required in order to route the post route through CVP.
CTI Route Point DN 6000 should be used to map to this Post Route routing script for intra site transfers to sales group.

Note: "send to VRU" node is required in order to route the post route through CVP.
Script-VRU DNIS Routing User Busy — 1 of 1

This script is used for the Alternate Destination Routing (ADR) on User Busy feature.

![Diagram of Script-VRU DNIS Routing User Busy](image-url)
This script is used for Legacy Transfer Connect (inband) VRU initiated to 8YY number.
This script is used for Legacy Transfer Connect (inband) VRU initiated to POTS number...
This script is used for IP Transfer Connect using SIP Redirect without sending 18X for the initial call. CVP will redirect the call using Speed Dial Code (SDC) number provided by AT&T Network, in this case the SDC number is 1084.

Note the absence of "Send to VRU" Script before the RF label in order for the IP Transfer Connect using SIP REDIRECT to happen.
This script is used for IP Transfer Connect using SIP Redirect with 18X for the initial call. CVP will not normally send 180 before 302 by itself. The only way is to have RNA call flow and a router query with a RF label. CVP will redirect the call after RNA timeout using Speed Dial Code (SDC) number provided by AT&T Network, in this case the SDC number is 1084.
This script is used for Multiple SIP Redirect test case. CVP will use “VRU Redirect” script to redirect the first transfer then it will use this script with SDC number 1085 to Redirect the second transfer.
This script is used for IP Transfer Connect using SIP Redirect to the AT&T Network test announcement. CVP will redirect the call using SDC number 91919191919. AT&T Network will recognize this SDC and terminate the call with an announcement.
This script is used for IP Transfer Connect using SIP REFER method. CVP will initiate Blind transfer using Speed Dial Code (SDC) number provided by AT&T Network, in this case the SDC number is 1084.

REFER transfer via CISCO UBE requires that the Network transfer Enable variable be set to 1 in the initial call’s VRU script and also in the Refer label script.

The RF Label should be preceded with “Send to VRU” Script in order for the IP Transfer Connect using SIP REFER to happen. The REDIRECT script is very similar except there is no “send to VRU” script before RF Label.
This script is used for IP Transfer Connect interaction with Contact Center test case. The Redirecting Party prompts the caller for digits in order to determine if the call should go into queue or be transferred via REFER to AT&T Network. If the caller selects transfer option, then CVP will initiate Blind transfer using Speed Dial Code (SDC) number provided by AT&T Network, in this case the SDC number is 1084.
This script is used for IP Transfer Connect using SIP REFER to the AT&T Network test announcement. CVP will Blind transfer the call using SDC number 91919191919. AT&T Network will recognize this SDC and terminate the call with an announcement.
Script-VRU Refer RP — 1 of 1

This script was used to run “INFO PAK” test cases which validates the provisioning combinations of CPN, BN, OLI, and UUI information when the caller is transferred using SIP REFER method. CVP will initiate Blind transfer using Speed Dial Code (SDC) number 1084.
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td>ADR</td>
<td>Alternate Destination Routing</td>
</tr>
<tr>
<td>ASR</td>
<td>Aggregation Services Routers</td>
</tr>
<tr>
<td>BT</td>
<td>Blind Transfer (Courtesy Transfer - Unattended)</td>
</tr>
<tr>
<td>CED</td>
<td>Customer Entered Digits</td>
</tr>
<tr>
<td>Cisco UBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>Cisco UCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>DNIS</td>
<td>Dialed Number Information Service</td>
</tr>
<tr>
<td>DTMF</td>
<td>Dual-Tone Multi Frequency, use of two simultaneous voice-band tones for dialing.</td>
</tr>
<tr>
<td>POTS</td>
<td>Plain Old Telephone Service</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>SDC</td>
<td>Speed Dial Code</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>ST</td>
<td>Soft Transfer CC - Conference and Consult</td>
</tr>
<tr>
<td>UBE</td>
<td>Unified Border Element</td>
</tr>
<tr>
<td>UUI</td>
<td>User-to-User Information</td>
</tr>
</tbody>
</table>
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