Sprint SIP Toll Free: Connecting Cisco Unified Customer Voice Portal 8.5 via the Cisco Unified Border Element 8.8 using SIP

Nov 6, 2012

Table of Contents

Introduction .......................................................................................................................... 3
Network Topology .............................................................................................................. 3
System Components ......................................................................................................... 4
Hardwar e Components ................................................................................................. 4
Software Requirements ................................................................................................. 4
Features ............................................................................................................................. 4
Tested Features ............................................................................................................... 4
Features Not Supported ................................................................................................. 5
Caveats ............................................................................................................................... 5
Configuration .................................................................................................................... 6
Configuring Cisco Unified Border Element ................................................................. 6
Configuring Cisco VXML Browser Gateway ............................................................... 13
Configuring the Cisco Unified Communications Manager ........................................... 18
  Software Version ......................................................................................................... 18
  SIP profile ..................................................................................................................... 19
  SIP Trunk Security Profile ............................................................................................. 20
  SIP Trunk ....................................................................................................................... 21
  Route Pattern 8222! (Used for calls to CVP) ................................................................. 23
  CTI Route Point ........................................................................................................... 24
  DN 2100 ....................................................................................................................... 26
  DN 2400 ....................................................................................................................... 27
  Region Configuration – One for with support for G.711, and another that is restricted to G.729 ........................................................................................................... 29
  Device Pools ................................................................................................................ 30
  Media Termination Point ............................................................................................. 32
  Transcoder .................................................................................................................... 32
  Media Resource Groups ............................................................................................... 33
  Media Resource List ..................................................................................................... 34
Configuring the Cisco Unified Customer Voice Portal .................................................... 35
  Software ......................................................................................................................... 35
  CVP Server ................................................................................................................... 36
  CVP Device Pool .......................................................................................................... 36
  CVP Server Configuration ........................................................................................... 37
Configuring the Cisco Unified Contact Center Enterprise ............................................ 39
  UCCE Version ............................................................................................................. 39
  PG Explorer – “CVP” Routing Client .......................................................................... 39
  PG Explorer – “CVP” Peripheral ................................................................................. 40
  PG Explorer – “UCCE” Routing Client ....................................................................... 41
  PG Explorer – “UCCE” Peripheral .............................................................................. 42
  ICM Network VRU Label 8222222222 – Used by CUCM when sending calls to CVP .............................................................. 43
  ICM Network VRU Label PSTN Destination ................................................................ 44

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
<table>
<thead>
<tr>
<th>Topic</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>Network VRU Media Hello World Script</td>
<td>45</td>
</tr>
<tr>
<td>Network VRU Hold Script</td>
<td>45</td>
</tr>
<tr>
<td>ICM Call Type “cvp”</td>
<td>46</td>
</tr>
<tr>
<td>ICM Call Type “cvp_support”</td>
<td>46</td>
</tr>
<tr>
<td>ICM DN 2100 Mapping</td>
<td>47</td>
</tr>
<tr>
<td>ICM DN 2400 Script selector</td>
<td>48</td>
</tr>
<tr>
<td>ICM DN 2400 Mapping</td>
<td>48</td>
</tr>
<tr>
<td>ICM DN script selector</td>
<td>49</td>
</tr>
<tr>
<td>ICM Person List</td>
<td>49</td>
</tr>
<tr>
<td>ICM Agent Explorer</td>
<td>50</td>
</tr>
<tr>
<td>ICM Agent Skill Group Support</td>
<td>50</td>
</tr>
<tr>
<td>ICM Agent Skill Group Support Members</td>
<td>51</td>
</tr>
<tr>
<td>ICM Label List</td>
<td>51</td>
</tr>
<tr>
<td>CTIOS Agent Desktop Display CPN</td>
<td>53</td>
</tr>
<tr>
<td>CTIOS Agent Desktop Restricted CPN</td>
<td>53</td>
</tr>
<tr>
<td>Script VRU DNIS Routing cvp</td>
<td>54</td>
</tr>
<tr>
<td>Script VRU Transfer to Queue</td>
<td>54</td>
</tr>
<tr>
<td>Script VRU Transfer Sending REFER (CVP Network Transfer)</td>
<td>55</td>
</tr>
<tr>
<td>Acronyms</td>
<td>56</td>
</tr>
<tr>
<td>Important Information</td>
<td>57</td>
</tr>
</tbody>
</table>
**Introduction**

Session Initiation Protocol (SIP) has emerged as a viable alternative to traditional circuit-switched interfaces for delivery of toll free traffic to contact centers. This Application Note details the configuration used for connectivity to the Sprint SIP Toll Free service. This document serves as guidance for the integration, but does not guarantee interoperability for every use case or release combinations.

The tested solution used Cisco Unified Contact Center Enterprise (Unified CCE) 8.5 with Cisco Unified Customer Voice Portal (Unified CVP) 8.5 using the Comprehensive Deployment Model.

**Network Topology**

*Figure 1. Basic Call Setup*
System Components

Hardware Components
- ISR G2 3925 (CUBE and VXML Browser)
- MCS 7800 Series servers (CUCM, CVP, and UCCE)
- Cisco 7961G SIP Phone
- Cisco 7961G SCCP Phone

Software Requirements
- IOS 15.2.1T (CUBE(Ent) version 8.8)
- IOS 15.2.1T (VXML Browser)
- Unified Contact Center Enterprise 8.5. (UCCE)
- Unified Customer Voice Portal 8.5 (CVP)
- Unified Communications Manager 8.5 (CUCM)
- Phone firmware 41.9

Features

Tested Features
- G.711ulaw and G.729 (no Annex B) codecs
- DNIS and ANI presentation
- SIP/TCP on CUBE internal interface (connection to CUCM and CVP), and SIP/UDP on external interface to Sprint’s network
- CVP-based Queuing
- CVP applications with DTMF via RFC2833
- CVP-based intra-site transfers using reINVITE
- CUCM-based intra-site transfers and conferences
- CUBE midcall codec negotiation and midcall transcoder insertion
- CUCM midcall codec negotiation (with transcoder insertion where needed)
- Survivability.tcl script (see caveats)
- REFER transfers with CUBE in REFER pass-through mode
- REFER transfers with CUBE in REFER consume mode
- CVP-based Redirect on No Answer
- Call hold
- SIP 302 “Moved Temporarily” with CUBE in pass-through mode
- SIP 302 “Moved Temporarily” with CUBE in consume mode
- Outbound calls

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Features Not Supported

- SIP over TLS and SRTP are not supported in the configuration tested
- Inband DTMF based transfers are not supported by Sprint
- Fax is not supported in the configuration tested
- REFER with Replaces is not supported
- CVP using H.323 is not supported

Caveats

- This Application Note is a result of functional testing only. The solution was not tested under load.
- Testing used CUBE on the Integrated Services Router. CUBE(Ent) for the Aggregation Services Router (ASR) has different capabilities, and does not support all the features listed previously.
- The current version of the survivability script does not allow REFER messages to be passed through CUBE and reach the service provider. If REFER pass-through is desired, the survivability script should not be used.
- CUBE High Availability with media failover was not tested.
- CVP Standalone Model was not tested.
- Outbound Call Progress Analysis (CPA) was not tested.
**Configuration**

**Configuring Cisco Unified Border Element**

```plaintext
version 15.2
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hosntame CUBE
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.152-1.T1.bin
warm-reboot
boot-end-marker
!
logging buffered 9999999
no logging console
logging monitor errors
enable secret 5 XXXXX
!
no aaa new-model
!
clock timezone CST -6 0
!
no ipv6 cef
ip source-route
ip auth-proxy max-login-attempts 5
ip admission max-login-attempts 5
!
!
ip cef
!
!
no ip domain lookup
!
multilink bundle-name authenticated
!
!
!
```

---

**Note:** Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
crypto pki token default removal timeout 0
!
!
voice-card 0
  dsp services dspfarm
!
!
voice service voip
  ip address trusted list
    ipv4 10.64.1.72
    ipv4 10.64.2.195
    ipv4 X.X.X.X (Sprint’s IP)
    ipv4 10.64.2.196
    ipv4 10.64.1.37
  address-hiding1
  allow-connections sip to sip
  no supplementary-service sip moved-temporarily2
  no supplementary-service sip refer3
  supplementary-service media-ренеготия4
  signaling forward none5
  h323
  sip
    rel1xx disable
    header-passing
    refer-to-passing6
    early-offer forced
    midcall-signaling passthru
!
voice class codec 1

1 With “address-hiding”, CUBE does not relay IP addresses from the service provider to CVP and vice versa – it replaces those with its own IP addresses instead.
2 Prevents “302 Redirect” responses from being relayed to Sprint, and instructs CUBE to redirect calls internally instead (Sprint does not support 302 Redirect).
3 Used when REFER consume is desired. If REFER pass-through is desired, use “supplementary-service sip refer”. Note that REFER pass-through does not work when the survivability.tcl script is used.
4 Controls midcall codec negotiation.
5 Controls whether GTD MIME bodies are passed through CUBE. CVP generates GTD when UUI is configured. If GTD needs to be passed through (unlikely), use “signaling forward unconditional”.
6 Controls whether the Refer-to header domain value provided by CVP will be passed on to Sprint unchanged or whether it will be replaced by CUBE’s own IP address (using the “no” form).
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.

Used to change the domain portion when 302 Redirect messages are sent to Sprint. Only required if 302 Redirect is used to transfer calls (not typical).
interface GigabitEthernet0/0
    description To SIP server providers
    ip address X.X.X.X 255.255.255.128
duplex full
    speed 100
interface GigabitEthernet0/1
    ip address 10.64.2.194 255.255.0.0
duplex full
    speed 100
interface GigabitEthernet0/2
    no ip address
    shutdown
duplex auto
    speed auto
interface GigabitEthernet0/3
    no ip address
    shutdown
duplex auto
    speed auto
interface FastEthernet0/0/0
interface FastEthernet0/0/1
interface FastEthernet0/0/2
interface FastEthernet0/0/3
interface VLAN1
  no ip address

ip default-gateway X.X.X.X
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 X.X.X.X
ip route 10.0.0.0 255.0.0.0 GigabitEthernet0/1
logging esm config
nls resp-timeout 1
cpd cr-id 1
control-plane

sccp local GigabitEthernet0/1
sccp ccm 10.64.2.194 identifier 1 version 4.0
sccp
sccp ccm group 1
  associate ccm 1 priority 1
  associate profile 1 register XCODE123456
  keepalive retries 1
  keepalive timeout 10
  switchover method immediate
  switchback method immediate

dspfarm profile 1 transcode
  codec g729r8
code g729br8

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
codec g711ulaw
maximum sessions 5
associate application SCCP
shutdown
!
!
dial-peer voice 9292 voip
description SIP error
dial-peer
service cverror
incoming called-number 92929292
voice-class codec 2
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 400 voip
description To CUCM
destination-pattern 1...
session protocol sipv2
session target ipv4:10.70.18.2
session transport tcp
voice-class codec 1 offer-all

dtmf-relay rtp-n-te
no vad
!
dial-peer voice 410 voip
description From CUCM
session protocol sipv2
session target ipv4:10.70.18.2
session transport tcp
incoming called-number .%
voice-class codec 1 offer-all

dtmf-relay rtp-n-te
no vad
!
!
dial-peer voice 2010 voip
description From Sprint
service survivability
session protocol sipv2
incoming called-number 91382795..

*“offer-all” enables midcall codec renegotiation end-to-end. If “offer-all” is not enabled in one of the dial peers or the voice class, CUBE will attempt midcall DSP insertion if the other dial peer in use has “offer-all” configured.
voice-class codec 1 offer-all
voice-class sip-profile 10
dtmf-relay rtp-nce
no vad
!
!
dial-peer voice 3010 voip
description To CVP
preference 1
destination-pattern 93182715..
session protocol sipv2
session target ipv4:10.64.2.195:5060
session transport tcp
voice-class codec 1 offer-all
dtmf-relay rtp-nce
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
    remote-party-id
    !
    !
telephony-service
    sds pfarm units 1
    sds pfarm transcode sessions 10
    sds pfarm tag 1 XCODE123456
    max-ephones 1
    max-dn 1
    ip source-address 10.64.2.194 port 2000
    max-conferences 8 gain -6
    transfer-system full-consult
    !
    !
    line con 0
    login local
    line aux 0

* Forces CUBE to insert the “Remote-Party-ID” header as required by CVP.
line vty 0 4
   exec-timeout 0 0
   login local
   transport input telnet ssh
!
exception data-corruption buffer truncate
exception crashinfo dump command sh controllers g0/0
exception crashinfo dump command sh controller g0/0
scheduler allocate 20000 1000
end

Configuring Cisco VXML Browser Gateway

version 15.2
service timestamps debug datatime msec localtime show-timezone year
service timestamps log datatime msec localtime show-timezone year
no service password-encryption
service internal
!
hostname VXML
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SPA.152-1.T1.bin
boot-end-marker
!
!
logging buffered 9999999
no logging console
logging monitor errors
enable secret 5 $XXXXXXXXXXXXXXXXXXXXXXXXX
!
no aaa new-model
!
clock timezone CST -6 0
!
dot11 syslog
ip source-route
!
ip cef
!
!
! 
no ipv6 cef 
!
multilink bundle-name authenticated 
!
!
!
!
!
voice service voip 
ip address trusted list
  ipv4 X.X.X.X
signaling forward unconditional
sip
  rel1xx disable
  min-se 360
  early-offer forced
!
!
voice class codec 1 
codec preference 1 g711ulaw 
!
!
!
!
!
http client cache memory pool 15000
http client cache memory file 500
http client connection timeout 60
http client response timeout 30
http client connection idle timeout 10
ivr prompt memory 15000 
!
application
  service new-call flash:bootstrap.vxml
  
  service ringtone flash:ringtone.tcl
  
  service cvperror flash:cvperror.tcl
  
  
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
service bootstrap flash:bootstrap.tcl
!

service handoff flash:handoff.tcl
!
!

vxml tree memory 500
crypto pki token default removal timeout 0
!
!
!
!
license udi pid CISCO3845-MB sn FOC11372RP6
archive
  log config
    hidekeys
username cisco privilege 15 secret 5 $XXXXXXXXXXXXXXXXXXXX
!
redundancy
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
!
interface GigabitEthernet0/1
  ip address 10.64.2.196 255.255.0.0
duplex full
  speed 100
media-type rj45
!
!
ip default-gateway 10.64.1.1

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
ip forward-protocol nd
ip http server
ip http access-class 23
ip http authentication local
no ip http secure-server
!
!
ip dns server
ip route 0.0.0.0 0.0.0.0 10.64.1.1
!
logging esm config
no logging trap
!
!
!
!
!
!
!
!
!
!
control-plane
!
!
!
!
!
!
dial-peer voice 9191 voip
description SIP ringtone dial-peer
service ringtone
incoming called-number 91919191
dtmf-relay rtp-npe
voice-class codec 1
no vad
!
dial-peer voice 9292 voip
description SIP error dial-peer
service cvperror
incoming called-number 92929292
dtmf-relay rtp-npe
voice-class codec 1
no vad
!
dial-peer voice 81 voip
description For Incoming Leg(Type 10 label end Correlation ID)
service bootstrap
incoming called-number 1234567890T

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
dtmf-relay rtp-nce
voice-class codec 1
no vad
!
!
gateway
timer receive-rtp 6
!
!
line con 0
  login local
line aux 0
line vty 0 4
  exec-timeout 0 0
  privilege level 15
  login local
  transport input all
!
scheduler allocate 20000 1000
netconf ssh
end
Configuring the Cisco Unified Communications Manager

Software Version

Last Successful Logon: Feb 17, 2012 6:28:14 PM

Copyright © 1999 - 2011 Cisco Systems, Inc.
All rights reserved.

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

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

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

For Cisco Technical Support please visit our Technical Support website.

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
SIP profile

Timer Limit (seconds) 180
Timer Register Delta (seconds) 5
Timer Register Engine (seconds) 4000
Timer To (infinity) 0
Timer To (infinity) 4000
Kern. IMAPR 0
Kern. IMAPR 10
Next Retry Max 40204
Max Media Rate 30728
Call Pickup User 1 user-remarque@tektel
Call Pickup User 2 user-remarque@tektel
Call Pickup Group user-remarque@tektel
Multimedia Service User user-remarque@tektel
User End Home
DPAM DB Level Normal
Call Hold Ring Both Off
Anonymous Call Signal Off
Caller ID Blocking Off
Do Not Disturb Control User
Talent Level for 7190 and 7960 Disabled
Timer Keep Alive Engine (seconds) 220
Timer Subscribe Engine (seconds) 150
Timer Subscribe Delta (seconds) 5
Maximum Redirection 70
Off Hook To End Digit Timer (milliseconds) 13000
Conference Join Threshold user-remarque@tektel
Caller ID on hold user-remarque@tektel
Send Attached Transfer user-remarque@tektel
Enable Session Bordering user-remarque@tektel
Remote incoming request to new trunk based on New
Remote Call 0
SIP Register Options Disabled
Deliver Conference Bridge Identifier
Early Offer support for voice and video calls (insert MTP if needed)
Send send-sdp-seq in mid-call INVITE

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
### SIP Trunk Security Profile

<table>
<thead>
<tr>
<th>Name*</th>
<th>User Secure SIP Trunk Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Security Mode</td>
<td>Non Secure</td>
</tr>
<tr>
<td>Incoming Transport Type</td>
<td>TCP/UDP</td>
</tr>
<tr>
<td>Enable Digest Authentication</td>
<td></td>
</tr>
<tr>
<td>Secure Validity Time (minutes)</td>
<td></td>
</tr>
<tr>
<td>Custom Subject Name</td>
<td></td>
</tr>
<tr>
<td>Incoming Port</td>
<td>5060</td>
</tr>
<tr>
<td>Enable Application-Level Authorization</td>
<td></td>
</tr>
<tr>
<td>Accept Presence Subscription</td>
<td></td>
</tr>
<tr>
<td>Accept Out-of-Dialog INFO**</td>
<td></td>
</tr>
<tr>
<td>Accept Unsolicited Notification</td>
<td></td>
</tr>
<tr>
<td>Accept Replaces Header</td>
<td></td>
</tr>
<tr>
<td>Transmit Security Status</td>
<td></td>
</tr>
</tbody>
</table>

*Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.*
In most implementations, it is desirable that MTP and transcoding resources used in calls to/from CVP be located close to where CVP is. Assigning a specific Media Resource Group List can accomplish this objective.

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizio Labs, an independent testing and certification facility, according to Cisco specifications.
Route Pattern 8222! (Used for calls to CVP)

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th>Status: Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern: 8222</td>
<td></td>
</tr>
<tr>
<td>N103 Pattern: &lt; Name &gt;</td>
<td></td>
</tr>
<tr>
<td>Description: calls to CVP</td>
<td></td>
</tr>
<tr>
<td>Routing: Not Selected</td>
<td></td>
</tr>
<tr>
<td>Route Filter: &lt; Name &gt;</td>
<td></td>
</tr>
<tr>
<td>LLQ (Priority): Default</td>
<td></td>
</tr>
<tr>
<td>Resource Priority: Network Domain: &lt; Name &gt;</td>
<td></td>
</tr>
<tr>
<td>Route Class: Default</td>
<td></td>
</tr>
<tr>
<td>Gateway/Route List: Extend</td>
<td></td>
</tr>
<tr>
<td>Route Option: Route this pattern, Block this pattern, No Error</td>
<td></td>
</tr>
<tr>
<td>Call Classification: Offnet</td>
<td></td>
</tr>
<tr>
<td>Allow DeviceOwnerId: Provide Outside Dial Tone</td>
<td></td>
</tr>
<tr>
<td>Require Forced Authorization Code:</td>
<td></td>
</tr>
<tr>
<td>Authorization Level: 0</td>
<td></td>
</tr>
<tr>
<td>Require Client Matter Code:</td>
<td></td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

- Use Calling Party's External Phone Number 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**

<table>
<thead>
<tr>
<th>Called Digit</th>
<th>Called Party Transform Mask</th>
<th>Prefix Digits (Outgoing Calls)</th>
<th>Called Party Number Type</th>
<th>Called Party Numbering Plan</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; Name &gt;</td>
<td></td>
<td></td>
<td>Cisco CallManager</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

**ISDN Network Specific Facilities Information Element**

<table>
<thead>
<tr>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
</table>

© 2012 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com
Page 23 of 58
EDCS# 1207649 Rev # 2

**Note:** Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
### CTI Route Point

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

**Status:**
- Status: Ready

#### Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager clus3pubsub 10.0.1.10</td>
</tr>
<tr>
<td>IP Address</td>
<td></td>
</tr>
<tr>
<td>Device is trusted</td>
<td></td>
</tr>
<tr>
<td>Device Name*</td>
<td>BlindTFR</td>
</tr>
<tr>
<td>Description</td>
<td>BlindTFR</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>View Details</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>Inrg_vrf</td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>&lt; None &gt;</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>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td></td>
</tr>
</tbody>
</table>

#### Association Information

- **Line 1** - 2100 (no partition)
- **Line 2** - Add a new DN

---

* indicates required item.

---

**Note:** Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Region Configuration – One for with support for G.711, and another that is restricted to G.729

<table>
<thead>
<tr>
<th>Region Relationships</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Region</td>
<td>Max Audio Bit Rate</td>
<td>Max Video Call Bit Rate [Includes Video]</td>
<td>Link Loss Type</td>
</tr>
<tr>
<td>Default</td>
<td>64 kbps (0.722, 0.711)</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**Note:** Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
### Device Pools

**Note:** Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Media Termination Point

Transcoder

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Media Resource Groups

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Media Resource List

![Media Resource List](image)

<table>
<thead>
<tr>
<th>Media Resource Group List Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Related Links</td>
</tr>
</tbody>
</table>

### Media Resource Group List Status

Media Resource Group List: eng_talk (used by 4 devices)

### Media Resource Group List Information

#### Media Resource Groups for this List

- **Available Media Resource Groups**
  - eng_talk

- **Selected Media Resource Groups**
  - eng_talk_001
  - eng_talk_002

**Note:** Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Configuring the Cisco Unified Customer Voice Portal

Software

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizon Labs, an independent testing and certification facility, according to Cisco specifications.
CVP Server Configuration

CVP needs Local Static Routes configured when SIP REFER and/or 302 Redirect is used. The Refer-To (and/or Contact) domain will be the one that matches the table.

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Configuring the Cisco Unified Contact Center Enterprise

UCCE Version

—

PG Explorer – “CVP” Routing Client

The “Network Transfer Preferred” check box is used to invoke CVP-based transfers (a.k.a. Network Transfers) when the same label can be used for both CVP and CUCM, and “NetworkTransferEnabled” is activated in the UCCE Script.

© 2012 Cisco Systems, Inc. All rights reserved.
Important notices, privacy statements, and trademarks of Cisco Systems, Inc. can be found on cisco.com

Page 39 of 58
EDCS# 1207649 Rev # 2

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
PG Explorer – “CVP” Peripheral

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
ICM Network VRU Label 8222222222 – Used by CUCM when sending calls to CVP

<table>
<thead>
<tr>
<th>Network VRU Explorer</th>
<th>Network VRU Banks</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name: CVP</td>
<td></td>
</tr>
<tr>
<td>Type: Type 10</td>
<td></td>
</tr>
<tr>
<td>Description: CVP-VRU for comprehensive call model</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Label</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Network VRU: CVP</td>
<td></td>
</tr>
<tr>
<td>Ringing client:</td>
<td></td>
</tr>
<tr>
<td>Label: 8222222222</td>
<td></td>
</tr>
<tr>
<td>Label Type: Normal</td>
<td></td>
</tr>
<tr>
<td>Customer: NIQUE</td>
<td></td>
</tr>
<tr>
<td>Description: User transfer label</td>
<td></td>
</tr>
</tbody>
</table>

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
ICM Network VRU Label PSTN Destination

Dynamic labels may also be used.

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Network VRU Media Hello World Script

Network VRU Hold Script

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
ICM Call Type “cvp”

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
ICM DN 2100 script selector

ICM DN 2100 Mapping

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
ICM DN 2400 Script selector

ICM DN 2400 Mapping

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
ICM Agent Skill Group Support Members

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.

ICM Label List
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
For CVP to send REFERs, either the call.user.sip.refertransfer variable needs to be set or the label must begin with “rf” (it is not necessary to have both).
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>CVP</td>
<td>Cisco Unified Customer Voice Portal</td>
</tr>
<tr>
<td>UCCE</td>
<td>Cisco Unified Contact Center Enterprise</td>
</tr>
</tbody>
</table>
**Important Information**

THE SPECIFICATIONS AND INFORMATION REGARDING THE PRODUCTS IN THIS MANUAL ARE SUBJECT TO CHANGE WITHOUT NOTICE. ALL STATEMENTS, INFORMATION, AND RECOMMENDATIONS IN THIS MANUAL ARE BELIEVED TO BE ACCURATE BUT ARE PRESENTED WITHOUT WARRANTY OF ANY KIND, EXPRESS OR IMPLIED. USERS MUST TAKE FULL RESPONSIBILITY FOR THEIR APPLICATION OF ANY PRODUCTS. IN NO EVENT SHALL CISCO OR ITS SUPPLIERS BE LIABLE FOR ANY INDIRECT, SPECIAL, CONSEQUENTIAL, OR INCIDENTAL DAMAGES, INCLUDING, WITHOUT LIMITATION, LOST PROFITS OR LOSS OR DAMAGE TO DATA ARISING OUT OF THE USE OR INABILITY TO USE THIS MANUAL, EVEN IF CISCO OR ITS SUPPLIERS HAVE BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.