Verizon IP Contact Center: Connecting Cisco Unified Customer Voice Portal 8.5 via the Cisco Unified Border Element 8.5 using SIP

May 04, 2012

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

Session Initiation Protocol (SIP) has emerged as a viable alternative to traditional circuit-switched interfaces for delivery of toll fee traffic to contact centers. This Application Note details the configuration used for connectivity to the Verizon IP Contact Center 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.

The Verizon IP Contact Service can deliver calls through VoIP Inbound or IP-IVR services. The interoperability characteristics of each are different, and both were tested for this Application Note. Also, the signaling for these services also changes when the Network Call Redirection (NCR) feature is turned on. Please regard to the “Known Caveats, Limitations and Other Comments” section for specific details.

Network Topology

Figure 1. Basic Call Setup
System Components

Hardware Components
- ISR G2 3925
- Cisco 7961G SIP Phone
- Cisco 7961G SCCP Phone

Software Requirements
- IOS 15.1.3T (CUBE)
- 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, and SIP/UDP on external interface
- CVP-based Queuing
- CVP applications with DTMF (see caveats)
- 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)
- DTMF Transfers using Verizon IP-IVR
- 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
- Verizon IP-IVR
- Verizon Network Call Redirection
- Verizon Release Link Trunking (RLT) (see caveats)
- Verizon Proprietary Headers

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
- Verizon IP-IVR does not support G729 codec while its Media Server is engaged (most common case)
- Verizon IP Contact Center Service does not support outbound calls, including during transfers (other Verizon SIP offers must be used instead)
- Fax is not supported on Verizon IP Contact Center
- SIP 302 Redirect is not supported on Verizon IP Contact Center (CUBE may consume 302 Redirect messages)
- 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.
- Verizon IP Contact Center does not support outbound calls. However, it does support DTMF transfers and REFER transfers, for both PSTN destinations as well as Verizon IP Contact Center destinations. Other Verizon SIP trunk services may be used for outbound calls, but they were not tested for this Application Note.
- 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.
- Verizon IP IVR only supports G.711ulaw, so calls routed through this service cannot be re-negotiated to G.729 end-to-end by default. Customers that want Verizon IP-IVR and G.729 in their contact centers should deploy transcoders (controlled by either CUBE or CUCM). Alternatively, Verizon also offers the ability to disengage the IP IVR Media Server when agents dial *7 from their phones (CVP “dtmf labels” should not be used to send these digits because calls get disconnected). Once the Media Server is disengaged, calls may renegotiate to different codecs.
- CUCM default hold method is not compatible with Verizon IP IVR, but interoperability can be achieved by using either sip-profile manipulation in CUBE (preferred) or MTP resources controlled by CUCM. There are no interoperability issues when Verizon VoIP Inbound is used instead.
- Verizon Network Call Redirection (NCR) is the feature that enables redirection if the destination is either busy or does not answer (ring no answer). NCR is optional and can be activated with VoIP Inbound and IP-IVR. This feature is incompatible with the Survivability script on CUBE, because the script also tries to deal with downstream element failures. For example, if CUBE sends the call to a busy destination, the script will redirect the call, or it will answer the call and play an error message –in both cases, the service provider is never made aware that the destination was busy. Customers may choose to deploy NCR or the survivability script, but both together may not lead to the intended outcome. When NCR is deployed without the survivability script, the Remote-Party-ID header must be used and manipulated to include “--CVP” at the end of the display name.
- 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.
- The version of CUBE tested does not support midcall codec negotiation (including midcall DSP insertion/removal) triggered by REFER requests (where REFER is not passed through the Service Provider). This feature requires IOS 15.2.1T or greater. Midcall codec negotiated triggered by reINVITE requests is supported and tested with the version of CUBE used.
- DTMF transfers rely on CUBE converting SIP INFO messages from CVP into RFC2833 RTP events that can be understood by Verizon. This feature has the following caveats:
- CUBE requires a release that contains a fix for bug CSCtj93573 (validated during testing).
- INFO-to-RFC2833 interoperability does not work when “Remote-Party-ID” is also configured in CUBE (bug CSCty35807). DTMF transfers were successfully validated with Remote-Party-ID off. Note that some CVP features rely on the Remote-Party-ID being there (e.g., Locations-Based Access Control, and the ability to send busy/redirect-no-answer responses to CUBE).
- CVP 8.5 Engineering Specials 4 and 6 (ES4 and ES6) both contain an enhancement that controls how CVP disconnects the call after outpulsing the DTMF digits. Applying either ES is highly recommended if DTMF transfers are used.
- The current release of CUBE(Ent) on the ASR platform (version 3.6) does not support this feature.

- CUBE High Availability with media failover was not tested.
- CVP Standalone Model was not tested.
Configuration

Configuring Cisco Unified Border Element

version 15.1
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CUBE
!
boot-start-marker
boot system flash c3900e-universalk9-mz.SSA-eng-sp-151-2T3.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
!
!
! 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 (Verizon’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-renegotiate⁴
signaling forward none⁵

h323
sip
rel1xx disable
header-passing
early-offer forced
midcall-signaling passthru
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!

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 Verizon, and instructs CUBE to redirect calls internally instead (Verizon 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”.

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.

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

voice class sip-profiles 10
  request REFER sip-header Refer-To modify "sip:(.*)@(.*)" "<sip:+1\1@2>" 7
!

voice class sip-profiles 20
  request REFER sip-header Route modify "Route: (.*)(.*)" "Route: \2,\1"
  response ANY sip-header Audio-Attribute modify "sendonly" "sendrecv" 10
  request ANY sip-header Route modify "Route: (.*)(.*)" "Route: \2,\1"

voice class sip-profiles 3030
  request INVITE sip-header Remote-Party-ID modify "<sip:(.*)@(.*)>" "--CVP <sip:\1@2>

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

voice translation-profile removeplus
  translate called 2
!

1 Used for calls from Verizon VoIP Inbound service.
7 Adds “+1” in front of the Refer-To header provided by CVP. Verizon requires the +1 format if calls can be REFER’ed to PSTN destinations.
8 Used for calls from Verizon IP IVR service.
9 Used as a workaround to address CSCtx17323 (CUBE not writing Route headers in the correct order when sending REFER messages). This sip-profile is not very robust, as it assumes there are always two Routes used, which may not be the case in every deployment. This sip-profile is only needed when Verizon IP-IVR is used.
10 Solves an interoperability issue between the “hold” messages sent by CUCM and Verizon IP-IVR. Not required when Verizon IP-IVR is not used.
11 Used for calls to CVP when survivability.tcl is not configured in the inbound dial peer (as is the case when Verizon Network Call Redirection, NCR, is on). Appending “--CVP” to the display name of the “Remote-Party-ID” header instructs CVP to send busy and ring-no-answer responses back to CUBE, instead of handling them itself. It also enables the Locations-Based Access Control feature.
application
service survivability flash:survivability.tcl
  paramspace english language en
  paramspace english index 0
  paramspace english location flash
  paramspace english prefix en
  param ccb id:10.64.2.194;loc:lab;trunks:1
!

service cvperror flash:cvperror.tcl
  paramspace english index 0
  paramspace english language en
  paramspace english location flash
  paramspace english prefix en
!
!
license udi pid C3900-SPE200/K9 sn FOC1403425C
!
!
hw-module pvdm 0/0
!
vtp domain Corp1
vtp mode transparent
vtp version 2
username cisco secret 5 $1$1FQ1$wmW1MrRzjU8AnzJUnFFhe0
!
redundancy
!
!
!
!
!
!
!
!
!
!
!

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
!
!

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
nls resp-timeout 1
cpd cr-id 1
!

control-plane
!
!
!
scpp local GigabitEthernet0/1
scpp ccm 10.64.2.194 identifier 1 version 4.0
scpp
!
scpp 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
  codec g729br8
  codec g711ulaw
  maximum sessions 5
  associate application SCCP
  shutdown
!

dial-peer voice 9292 voip
description SIP error dial-peer
description cvperror
  incoming called-number 92929292
description voice-class codec 2
description dtmf-relay rtp-nte
  no vad
!
dial-peer voice 400 voip
description To CUCM
description destination-pattern 1...
description session protocol sipv2
description session target ipv4:10.70.18.2

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
session transport tcp
voice-class codec 1 offer-all\(^\text{12}\)
dtmf-relay rtp-nte
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\(^\text{Error! Bookmark not defined.}\)
dtmf-relay rtp-nte
no vad
!
\textbf{dial-peer voice 500 voip}\(^\text{13}\)
description Refer Passthrough Dial Peer Match
destination-pattern 1972
session protocol sipv2
session target ipv4:10.70.18.2
voice-class codec 1 offer-all\(^\text{Error! Bookmark not defined.}\)
dtmf-relay rtp-nte
no vad
!
dial-peer voice 2010 voip
description From Verizon VoIP Inbound
service survivability
session protocol sipv2
incoming called-number 8666735...
voice-class codec 1 offer-all\(^\text{Error! Bookmark not defined.}\)
voice-class sip-profile 10
dtmf-relay rtp-nte
dtmf-interworking rtp-nte
no vad
!
dial-peer voice 2020 voip

\(^{12}\)“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.

\(^{13}\)Prior to IOS 15.2.1T, CUBE requires a dial peer to match the “Refer-To” header in order to process REFER passthrough messages. CUBE then replaces used in the Refer-To header with its own IP address. In later releases, this dial peer is not necessary if “referto-passing” is configured in the voice class or in the dial peers used by the call. “referto-passing” instructs CUBE to relay the Refer-to payload unchanged, therefore the exact payload selected by CVP reaches Verizon.
description From Verizon IP-IVR
service survivability
session protocol sipv2
incoming called-number 8666733...
voice-class codec 1 offer-all
voice-class sip-profile 20
dtmf-relay rtp-npe
dtmf-interworking rtp-npe
no vad
!
dial-peer voice 2030 voip

description From Verizon VoIP Inbound NCR
session protocol sipv2
incoming called-number 8666747...
voice-class codec 1 offer-all
voice-class sip-profile 10
dtmf-relay rtp-npe
dtmf-interworking rtp-npe
no vad
!
dial-peer voice 3010 voip

description To CVP No NCR
translation-profile outgoing removeplus
preference 1
destination-pattern 866673....
session protocol sipv2
session target ipv4:10.64.2.195:5060
session transport tcp
voice-class codec 1 offer-all
dtmf-relay rtp-npe
dtmf-interworking rtp-npe
no vad
!
dial-peer voice 3030 voip

description To CVP NCR Enabled
translation-profile outgoing removeplus
preference 1
destination-pattern 866674....
session protocol sipv2
session target ipv4:10.64.2.195:5060
session transport tcp

Note the survivability script is not used when Verizon NCR is enabled.
voice-class codec 1 offer-all
voice-class sip-profile 3030
dtmf-relay rtp-nte
dtmf-interworking rtp-nte
no vad
!

gateway
timer receive-rtp 1200
!
sip-ua
remote-party-id\[1\]
!
!

telephony-service
sdspfarm units 1
sdspfarm transcode sessions 10
sdspfarm 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
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

\[1\] Verizon sends “P-Asserted-Identity” privacy headers, and CVP requires “Remote-Party-ID” instead. Note this should not be used if CSCty35807 is hit.
Configuring Cisco VXML Browser Gateway

version 15.1
service timestamps debug datetime msec localtime show-timezone year
service timestamps log datetime msec localtime show-timezone year
no service password-encryption
service internal
!
hostname c3845
!
boot-start-marker
boot system flash:c3845-adventerprisek9_ivs-mz.151-3.T2.bin
boot-end-marker
!
!
card type t1 1 1
! card type command needed for slot/vwic-slot 2/1
logging buffered 9999999
no logging console
logging monitor errors
enable secret 5 $XXXXXXXXXXXXXXXXXXXXXXXXXXXX
!
no aaa new-model
!
clock timezone CST -6 0
no network-clock-participate slot 1
no network-clock-participate slot 2
!
dot11 syslog
ip source-route
!
ip cef
!
!
!
no ipv6 cef
!
multilink bundle-name authenticated
!
!
!
voice-card 1

voice-card 2

voice service voip
ip address trusted list
ipv4 X.X.X.X
allow-connections sip to sip
no supplementary-service sip moved-temporarily
signaling forward unconditional
sip
rel1xx disable
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
paramspace english language en
paramspace english index 0
paramspace english location flash
paramspace english prefix en

service ringtone flash:ringtone.tcl
paramspace english language en
paramspace english index 0
paramspace english location flash
paramspace english prefix en
!
service cvperror flash:cvperror.tcl
paramspace english index 0
paramspace english language en
paramspace english location flash
paramspace english prefix en
!
service bootstrap flash:bootstrap.tcl
param cvpserverport 8000
param cvpserverbackup 10.64.2.195
param cvpserverhost 10.64.2.195
!
service handoff flash:handoff.tcl
paramspace english language en
paramspace english index 0
paramspace english location flash
paramspace english prefix en
!
!
! 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
!
!
!
ip ssh time-out 60
ip ssh authentication-retries 2
ip ssh version 1
!
!
!
!
!
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
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-nre
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-nre
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
dtmf-relay rtp-nre
voice-class codec 1
no vad
!
!
gateway
timer receive-rtp 1200
!
!

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

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

**SIP Profile Information**

- **Name**: Default SIP Profile
- **Description**: Default SIP Profile
- **Default HTTP Telephony Event Purged Type**: 10
- **Resource Priority Name/Path List**: None
- **Early Offer for A-Channel Calls**: Disabled
- **Readied for Application**: False
- **Invoke early media on SIP**: False
- **Ordering 7.26 INVITE include audio mime**: False
- **Execute SMF**: False
- **Receive SDP Inactive Exchange for Mid-Call Media Change**: False

**Parameters used in These**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Frames (seconds)</td>
<td>100</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>5000</td>
</tr>
<tr>
<td>Timer T2 (ms)</td>
<td>5000</td>
</tr>
<tr>
<td>Early INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Early INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Max Media Ports (bits)</td>
<td>1234</td>
</tr>
<tr>
<td>Max Media Ports (bits)</td>
<td>32768</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>sip:<a href="mailto:admin@cisco.com">admin@cisco.com</a></td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>sip:<a href="mailto:admin@cisco.com">admin@cisco.com</a></td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>sip:<a href="mailto:meetme@cisco.com">meetme@cisco.com</a></td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Normal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Centre</td>
<td>On</td>
</tr>
<tr>
<td>Telnet Level for 7945 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Timer Keep A Channel Expires</td>
<td>15000</td>
</tr>
<tr>
<td>Timer Keep A Channel Delta</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To Post Digit Time</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>sip:<a href="mailto:admin@cisco.com">admin@cisco.com</a></td>
</tr>
<tr>
<td>Speed Dial (Alphanumeric Dial URI)</td>
<td>sip:<a href="mailto:admin@cisco.com">admin@cisco.com</a></td>
</tr>
</tbody>
</table>

**SIP Specific Configuration**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Random Incoming Request to new Trunk based on</td>
<td>Never</td>
</tr>
<tr>
<td>Keep Date To local ntp</td>
<td>Local NTP</td>
</tr>
<tr>
<td>SIP Fail UX Options</td>
<td>Disabled</td>
</tr>
<tr>
<td>Early Offer for A-Channel Calls</td>
<td>None</td>
</tr>
<tr>
<td>Send and receive SDP for mid-call UMTS</td>
<td>False</td>
</tr>
</tbody>
</table>

**SIP Options**

- **Enable OPTIONS Ping to monitor destination status for Trunk with Service Type “None (Default)”**: False
- **Ping Interval for Unreachable and Partially Unreachable Trunks (seconds)**: 500
- **Ping Retry Timeout for Unreachable Trunks (seconds)**: 120
- **Ping Retry Count**: 5

---

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

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 Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
**CTI Route Point**

![Cisco Unified CM Administration](image)

**Device Information**

<table>
<thead>
<tr>
<th>Registration</th>
<th>Registered with Cisco Unified Communications Manager dus3/customer</th>
<th>10.64.1.10</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address</td>
<td>10.64.1.30</td>
<td></td>
</tr>
<tr>
<td>Device Name</td>
<td>BlindTFR</td>
<td></td>
</tr>
<tr>
<td>Description</td>
<td>BlindTFR</td>
<td></td>
</tr>
<tr>
<td>Device Pool</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
<td></td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>IT_RL_VUHL</td>
<td></td>
</tr>
<tr>
<td>Network Hold MOH Audio Source</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>User Hold MOH Audio Source</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Use Trusted Relay Point</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
<tr>
<td>Gaslocation</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

**Association Information**

- Line 11: 2100 (no partition)
- Line 12: add a new DN

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

<table>
<thead>
<tr>
<th>Media Termination Point Configuration</th>
<th>Related Links</th>
<th>New</th>
<th>Edit</th>
<th>Delete</th>
<th>Copy</th>
<th>Reset</th>
<th>Apply Config</th>
<th>Add New</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Media Termination Point Information</strong></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Registration: Unknown</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>IP Address: Unknown</td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Media Termination Point Type: Cisco CM Enhanced Software Media Termination Point</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
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<td>Description:</td>
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<td></td>
<td></td>
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<td></td>
<td></td>
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</tr>
<tr>
<td>Default Port: Default</td>
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<td></td>
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</tr>
<tr>
<td>Trusted Relay Point</td>
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<td></td>
<td></td>
<td></td>
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</table>

* indicates required field.

---

### Transcoder

<table>
<thead>
<tr>
<th>Transcoder Configuration</th>
<th>Related Links</th>
<th>New</th>
<th>Edit</th>
<th>Delete</th>
<th>Copy</th>
<th>Reset</th>
<th>Apply Config</th>
<th>Add New</th>
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</thead>
<tbody>
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<td><strong>Transcoder Information</strong></td>
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</tr>
<tr>
<td>Transcoder: Engineering (CMX, peer transcoding)</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Registration: Unknown</td>
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<td></td>
<td></td>
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<tr>
<td>IP Address: Unknown</td>
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<td></td>
<td></td>
</tr>
<tr>
<td>VSN Transcoder Info</td>
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<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Transcoder Type: Cisco CM Enhanced Media Termination Point</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Description:</td>
<td></td>
<td></td>
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<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Default Port: Default</td>
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<td></td>
<td></td>
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<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Trusted Relay Point</td>
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</tr>
</tbody>
</table>

* indicates required field.

---

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

CVP needs Local Static Routes configured when SIP REFER is used. The Refer-To domain will be the one that matches the table. As footnote Error! Bookmark not defined. describes, this domain may be forwarded to Verizon if CUBE uses an IOS release that supports that refer-to-passing parameter at a dial peer level.

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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.
PG Explorer – “CVP” Peripheral

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ICM Network VRU Label 8222222222 – Used by CUCM when sending calls to CVP

**Note:** Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
Dynamic labels may also be used.
**Network VRU Media Hello World Script**

**Network VRU Hold Script**

---

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ICM Call Type “cvp”

ICM Call Type “cvp_support”

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 DN 2100 Mapping

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

ICM DN IP-IVR 800 Service 866676XXXX script selector

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For CVP to send REFERs, either the call.user.sipREFERtransfer variable needs to be set or the label must begin with “rf” (it is not necessary to have both).
Note: Testing was conducted in Tekvizon Labs, an independent testing and certification facility, according to Cisco specifications.
### Acronyms

<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>
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