XO SIP Trunk: Connecting Cisco Unified Customer Voice Portal 8.5 via the Cisco Unified Border Element 8.5 using SIP

June 18, 2012

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EDCS# 1169295 Rev # Initial version

Note: Testing was conducted in Tekvizion 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 free traffic to contact centers. This Application Note details the configuration used for connectivity to the XO SIP Trunk 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
- 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 and inter-site transfers using reINVITE
- CUCM-based intra-site and inter-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
- Mobile Agent
- SIP 302 Redirect responses sent to XO

Features Not Supported
- SIP over TLS and SRTP are not supported in the configuration tested
- Fax and video are not supported
- REFER with Replaces is not supported

Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
• CVP using H.323 is not supported
• DTMF based transfers are not supported (*8)

Caveats

• This Application Note is a result of functional testing only. The solution was not tested under load.
• XO SIP Trunk only accepts outbound calls from phone numbers (ANI) ported to XO. The validation is performed by looking at the following headers, in order of priority: Diversion, P-Asserted Identity, P-Preferred Identity, Remote Party ID, and From.
• 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 and 302 Redirect messages to be passed through CUBE and reach the service provider. If REFER and 302 Redirect 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.
• CUBE High Availability with media failover was not tested.
• CVP Standalone Model was not tested.
**Configuration**

*Configuring Cisco Unified Border Element*

```bash
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
!
!
!
```

*Note: Testing was conducted in Tekvition 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 (XO’s IP address)
ipv4 10.64.2.196
ipv4 10.64.1.37
address-hiding
allow-connections sip to sip
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
supplementary-service media-renegotiate
signaling forward none
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 Used when 302 Redirect consume is desired. If 302 Redirect pass-through is desired, use “supplementary-service sip moved-temporarily”. Note that 302 Redirect pass-through does not work when the survivability.tcl script is used.

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”.
voice class codec 2
  codec preference 1 g711ulaw

voice class codec 3
  codec preference 1 g729r8

voice class sip-profiles 20
  request INVITE sip-header Remote-Party-ID modify "<sip:(.*)@" "<sip:214XXXXX@""

voice class sip-profiles 3030
  request INVITE sip-header Remote-Party-ID modify "\"(.*)" <sip:(.*)@(.*)>" \1--CVP <sip:\2@\3>"

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 cv perror flash:cv perror.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

---

6 Used for outbound calls to XO.

7 XO SIP Trunks only accepts outbound calls from XO-owned numbers, as determined by the call Diversion, P-Asserted-Identity, P-Preferred-Identity, Remote-Party-ID or From header (in order of precedence). This sip-profile ensures all outbound calls will use the same P-Asserted-Identity, but other approaches that meet XO’s requirements could be used.

8 Used for calls to CVP when survivability.tcl is not configured in the inbound dial peer. 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.
vtp domain Corp1
vtp mode transparent
vtp version 2
username cisco secret 5 $1$1FQ1$wmWlMrRzjU8AnzJUnFFhe0

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
!
!
!
nls resp-timeout 1
cpd cr-id 1
!
!
control-plane
!
!
!
sc cp local GigabitEthernet0/1
sc c cm 10.64.2.194 identifier 1 version 4.0
sc c mp
!
sc c cm group 1
    associate c cm 1 priority 1
    associate profile 1 register XCODE123456
    keepalive retries 1
    keepalive timeout 10
    switchover method immediate
    switchback method immediate

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.
description Refer Passsthrough Dial Peer Match
destination-pattern 866
session protocol sipv2
session target ipv4:X.X.X.X
voice-class codec 1 offer-all
dtmf-relay rtp-nnte
no vad
!
dial-peer voice 2010 voip
description From XO

    service survivability

    session protocol sipv2
    incoming called-number 214XXXXXXXX
    voice-class codec 1 offer-all
    dtmf-relay rtp-nnte
    no vad
!
dial-peer voice 2010 voip
description To XO

    session protocol sipv2
    session target ipv4:X.X.X.X
    destination-pattern 1XXXXXXXX
    voice-class codec 1 offer-all
    voice-class sip profiles 20
    dtmf-relay rtp-nnte
    no vad
!
dial-peer voice 3010 voip
description To CVP w Survivability

destination-pattern 214XXXXXXX
    session protocol sipv2
    session target ipv4:10.64.2.195:5060
    session transport tcp
    voice-class codec 1 offer-all
    dtmf-relay rtp-nnte
    no vad
!

dial-peer voice 3030 voip

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 XO.

11 Survivability should not be used if REFER or 302 Redirect Pass-Through is desired.
12 If Survivability is not used, the dial peer to CVP should have a sip-profile manipulating the Remote-Party-ID header.
description To CVP wo Survivability
shutdown
destination-pattern 214XXXXXXX
session protocol sipv2
session target ipv4:10.64.2.195:5060
session transport tcp
voice-class codec 1 offer-all
voice-class sip-profile 3030
dtmf-relay rtp-nate
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
remote-party-id
!
!
!
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

"XO sends “P-Preferred-Identity” privacy headers, and CVP requires “Remote-Party-ID” instead."
exception crashinfo dump command sh controller g0/0
scheduler allocate 20000 1000
end
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
!
!
logging buffered 9999999
no logging console
logging monitor errors
enable secret 5 $XXXXXXXXXXXXXXXXXXXXXXXXXXXX
!
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
header-passing
!
!

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
!
service bootstrap flash:bootstrap.tcl
!
service handoff flash:handoff.tcl
!
!
xml 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
!
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!
!
!
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

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Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
no logging trap
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!
!
```
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

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

**SIP Profile Configuration**

#### SIP Profile Information
- **Name**: Default SIP Profile
- **Description**: Default SIP Profile
- **Default MTU Telephony Event Filter Type**: 101
- **Resource Priority Namespace List**: None
- **Early Offer for & Clear Calls**: Disabled
- **Send by Application**: Required
- **Erase early media on close**: Enabled
- **Dropping 1.15 IN/INTE include audio only**: Disabled
- **Enable AMR**: Enabled
- **Receivers GSP Inactive Exchange for Mid-Call Media Change**: Enabled

#### Parameters used in these:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires</td>
<td>100</td>
</tr>
<tr>
<td>Timer Register Expires</td>
<td>500</td>
</tr>
<tr>
<td>Timer Register Expires</td>
<td>500</td>
</tr>
<tr>
<td>Timer T1 (milliseconds)</td>
<td>500</td>
</tr>
<tr>
<td>Rdy Call INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Rdy Call INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Meet Me Invite Port</td>
<td>10344</td>
</tr>
<tr>
<td>Meet Me Invite Port</td>
<td>3776C</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>sip:тренерователно-виртуален</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>sip:тренерователно-виртуален</td>
</tr>
<tr>
<td>Meet Me Invite URI</td>
<td>sip:тренерователно-виртуален</td>
</tr>
<tr>
<td>User Info</td>
<td>Home</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Normal</td>
</tr>
<tr>
<td>Call Hold Ring Bath</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 Center</td>
<td>User</td>
</tr>
<tr>
<td>Telelink Level For 7950</td>
<td>Disabled</td>
</tr>
<tr>
<td>Timer Keep Alive Expires</td>
<td>120</td>
</tr>
<tr>
<td>Timer Keep Alive Expires</td>
<td>120</td>
</tr>
<tr>
<td>Maximum Registrations</td>
<td>70</td>
</tr>
<tr>
<td>Off Hold To Fast Digit Time (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>sip:тренерователно-виртуален</td>
</tr>
<tr>
<td>Speed Dial (Activated Only) URI</td>
<td>sip:тренерователно-виртуален</td>
</tr>
</tbody>
</table>

#### Specific Configuration

- Remote Incoming Request to new trunk based on: Never
- ISDN Class of Service: Local ISDN
- Send Conference Bridge Identifier: Disabled
- Early Offer support for voice and video calls (insert H223 if needed): Enabled
- Send send receive SIP in mid-call INVITE: Enabled

#### SIP Options

- **Enable ODFCNG Ping to monitor destination status for Trunks with Service Type "InfiniBand"**: Checked
- **Ping interval for InfiniBand and Fatally InfiniBand Trunks (seconds)**: 200
- **Ping interval for InfiniBand Trunks (milliseconds)**: 1000
- **Ping Relay Timer (milliseconds)**: 500
- **Ping Relay Count**: 5

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### SIP Trunk Security Profile

**SIP Trunk Security Profile Information**

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Device Security Mode</th>
<th>Incoming Transport Type</th>
<th>Outgoing Transport Type</th>
<th>Enable Digest Authentication</th>
<th>Secure Validity Time (ms)</th>
<th>SAML Subject Name</th>
<th>Enable Application-Level Authentication</th>
<th>Accept Presence Subscription</th>
<th>Accept Out-of-Binding INVITE</th>
<th>Accept Unspecified Invocation</th>
<th>Accept Replaces Header</th>
<th>Transmit Security Status</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Testing**

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

### Device Information

<table>
<thead>
<tr>
<th>Status</th>
<th>Device Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager dups/pubsub</td>
</tr>
<tr>
<td>IP Address</td>
<td>10.64.1.36</td>
</tr>
</tbody>
</table>

- **Device Name**: BlindFFR
- **Description**: BlindFFR
- **Device Pool**: Default
- **Common Device Configuration**: < None >
- **Calling Search Space**: < None >
- **Location**: Hub_None
- **User Locale**: < None >
- **Media Resource Group List**: Nvl_vml
- **Network Hold MOH Audio Source**: < None >
- **User Hold MOH Audio Source**: < None >
- **Use Trusted Relay Point**: Default
- **Calling Party Transformation CSS**: < None >
- **GsoLocation**: < None >

### Association Information

- **Line 1[1]: 2100 (no partition)**
- **Line 1[2]: add a new DN**

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

* indicates required item.
Note: Testing was conducted in Tekvizion Labs, an independent testing and certification facility, according to Cisco specifications.
**DN 2100**

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

**Directory Number Configuration**

**Status**

- Status: Ready

**Directory Number Information**

- Directory Number: 7100
- Route Partition: < None >
- Description: < None >
- Alerting Name: < None >
- ASCII Alerting Name: < None >
- Associated Devices: BlindTR

**Directory Number Settings**

- Voice Mail Profile: < None >
- Calling Search Space: < None >
- Presence Group: Standard Presence Group
- User MIM MIM Audio Source: < None >
- Network MIM MIM Audio Source: < None >

**AAR Settings**

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

[Retain this destination in the call forwarding history]

**Call Forward and Call Pickup Settings**

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

---

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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

<table>
<thead>
<tr>
<th>Default</th>
<th>G729</th>
<th>64 Kbps (G.711)</th>
<th>384</th>
<th>Use System Default</th>
</tr>
</thead>
<tbody>
<tr>
<td>Keep Current Setting</td>
<td>3</td>
<td>Keep Current Setting</td>
<td>3</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

* indicates required item.
Device Pools

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Media Termination Point

Transcoder

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

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Configuring the Cisco Unified Customer Voice Portal Software

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CVP Server

CVP Device Pool

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CVP Server Configuration

[Diagram of CVP Server Configuration]

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EDCS# 1169295 Rev # Initial version
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Configuring the Cisco Unified Contact Center Enterprise

UCCE Version

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.

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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.
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.
ICM Network VRU Label PSTN Destination

17 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”

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 Tekvizon Labs, an independent testing and certification facility, according to Cisco specifications.
ICM DN 2400 Mapping

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ICM DN Mapping

ICM Person List

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ICM Agent Explorer

ICM Agent Skill Group Support

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

ICM Label List

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18 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).
## 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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<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
</tr>
</thead>
<tbody>
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<td>Haarlerbergpark</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlerbergweg 13-19</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td>The Netherlands</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
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<td>Tel: 31 0 20 357 1000</td>
<td>Tel: 408 526-7660</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td>Fax: 31 0 20 357 1100</td>
<td>Fax: 408 527-0883</td>
<td>Tel: +65 317 7777</td>
</tr>
<tr>
<td>Fax: 408 526-4100</td>
<td></td>
<td>Fax: +65 317 7799</td>
<td></td>
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</tbody>
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