TELUS IP Trunking: Connecting Cisco Unified Communications Manager 8.6 via the Cisco Unified Border Element 10.0.1 [15.4(2)T] (Enterprise Edition) using SIP

December 2014

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EDCS# xxx Rev # <edcs revision number>

Note: Testing was conducted in Telus labs
Introduction

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

- This application note describes how to configure a Cisco Unified Communications Manager (CUCM) 8.6.2.25140-1 with a Cisco Unified Border Element (CUBE) 10.0.1 for connectivity to TELUS IP Trunking SIP trunk service. The deployment model covered in this application note is CPE to PSTN. This document does not address 911 emergency outbound calls. For 911 feature service details contact TELUS directly.

- Testing was performed in accordance to TELUS test plan and all features were verified. Key features verified are: Listed under features in this document.

- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between TELUS SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful and care must be taken, by the network administrator deploying CUBE, to ensure these commands are set per each dial-peer requiring to interoperate to TELUS SIP network.

- This application note does not cover the use of Calling Search Spaces (CSS) or Partitions on Cisco Unified Communications Manager. To understand and learn how to apply CSS and Partitions refer to the cisco.com link below:
  http://www.cisco.com/en/US/docs/voice_ip_comm/cucm/admin/6_0_1/ccmsys/a03ptcss.html
Network Topology

System Components

Hardware Components
- Cisco 2911/K9 (Cisco 2900 family router)
- CUCM cluster with (2) Cisco MCS 7800 Series server (Cisco Unified Communications Manager)
- 4 Cisco Unified IP Phones (7965)
- VG204XM MGCP gateway for fax
- Cisco 3560C powered Ethernet switch
- Unity connection for voice mail as well as Auto Attendant
- Fax machine G3

Software Requirements
The following software is required:
- Cisco Unified Communications Manager Release 8.6. This solution was tested with 8.6.2.25140-1
- Cisco Unity Connection Release 8.6. This solution was tested with 8.6.2ES144.25140-144
- Cisco Unified Border Element Release 10.0.1 with IOS version 15.4.2T release. This configuration was tested with c2900-universalk9-mz.SPA.154-2.T.bin
- Cisco VG204XM with IOS version 15.3.2T. This configuration was tested with vg20xxm-ipvoice-mz.153-2.T.bin
Features

Features Supported

- Basic Call using G.729 and G711 (inbound and outbound).
- Calling Party Number Presentation (CLIP).
- Calling Party Number Restriction (CLIR).
- Calling Name.
- Intra-site Call Transfer (Attended and Unattended).
- Intra-site Conference.
- Call Hold and Resume.
- Call Forward All, Busy and No Answer.
- Toll-free numbers.
- Long calls durations.
- DTMF (RFC2833).
- Fax using G.711 pass-through.
- Cisco Unity Connection Auto-attendant.
- Cisco Unity Connection Voice mail.
- Calling number privacy.

Features Not Supported

- Emergency 911 calls were not tested.
- Failover was not tested.
Configuration

Configuring Cisco Unified Border Element (CUBE)

Show Version

Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(2)T, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2014 by Cisco Systems, Inc. Compiled Wed 26-Mar-14 14:14 by prod_rel_team
ROM: System Bootstrap, Version 15.0(1r)M16, RELEASE SOFTWARE (fc1)
IPT-Cert-CUBE uptime is 4 hours, 15 minutes
System returned to ROM by reload at 10:08:11 PDT Fri Sep 12 2014
System restarted at 10:09:56 PDT Fri Sep 12 2014
System image file is "flash:c2900-universalk9-mz.SPA.154-2.T.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

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Cisco CISCO2911/K9 (revision 1.0) with 475136K/49152K bytes of memory.
Processor board ID FGL174711EA
4 Gigabit Ethernet interfaces
2 terminal lines
2 Channelized (E1 or T1)/PRI ports
1 Virtual Private Network (VPN) Module
1 Internal Services Module (ISM) with Services Ready Engine (SRE)
Cisco Unity Express 8.6.6 in slot/sub-slot 0/0
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
250880K bytes of ATA System CompactFlash 0 (Read/Write)

License Info:
License UDI:

<table>
<thead>
<tr>
<th>Device#</th>
<th>PID</th>
<th>SN</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>CISCO2911/K9</td>
<td>FGL174711EA</td>
</tr>
</tbody>
</table>

Technology Package License Information for Module:'c2900'

<table>
<thead>
<tr>
<th>Technology</th>
<th>Technology-package</th>
<th>Current</th>
<th>Type</th>
<th>Next reboot</th>
</tr>
</thead>
<tbody>
<tr>
<td>ipbase</td>
<td>ipbasek9</td>
<td>Permanent</td>
<td>ipbasek9</td>
<td></td>
</tr>
<tr>
<td>security</td>
<td>securityk9</td>
<td>Permanent</td>
<td>securityk9</td>
<td></td>
</tr>
<tr>
<td>ue</td>
<td>uck9</td>
<td>Permanent</td>
<td>uck9</td>
<td></td>
</tr>
<tr>
<td>data</td>
<td>datak9</td>
<td>Permanent</td>
<td>datak9</td>
<td></td>
</tr>
<tr>
<td>NewkEss</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>CollabPro</td>
<td>None</td>
<td>None</td>
<td>None</td>
<td>None</td>
</tr>
</tbody>
</table>

Configuration register is 0x2102

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Show Running-Configuration

Current configuration : 4014 bytes

! Last configuration change at 14:19:18 PDT Fri Sep 12 2014 by YYYY
! NVRAM config last updated at 14:22:23 PDT Fri Sep 12 2014 by YYYY
!
version 15.4
!
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service password-encryption
service sequence-numbers
!
hostname IPT-Cert-CUBE
!
boot-start-marker
boot system flash c2900-universalk9-mz.SPA.154-2.T.bin
boot-end-marker
!
asm-register-fnf
!
card type command needed for slot/vwic-slot 0/0
logging buffered 1000000
no logging rate-limit
no logging console
!
no aaa new-model
!
clck timezone EST -8 0
!
no ip domain lookup
ip domain name cube.ipt.local
ip cef
no ipv6 cef
!
multilink bundle-name authenticated
!
voice-card 0
dsfarm
dsp services dsfarm
!
voice service voip
!
no ip address trusted authenticate
mode border-element license capacity 25
allow-connections sip to sip
redirect ip2ip
fax protocol pass-through g711ulaw h323
sip
refer-to-passing
asserted-id pai
asymmetric payload dmf
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all

1 This introduces the mode border-element command to distinguish between Cisco Unified Communications Manager Express and Cisco UBE configuration.
2 Enables the P-Asserted-Identity (PAI) privacy header in incoming and outgoing SIP requests or response messages.
voice class codec 1
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice class sip-profiles 150
request INVITE sip-header From modify "Anonymous" "IPT-Cert-Phone3-Line-1"
request INVITE sip-header From modify "<sip:anonymous@anonymous.invalid>" "<sip:+15877569936@Z.Z.Z.Z>
license udi pid CISCO2911/K9 sn FGL174711EA
hw-module ism 0
!
hw-module pvdm 0/0
!
username YYYYY privilege 15 secret XXXXXXXXXXXXX
!
redundancy
!
interface Embedded-Service-Engine0/0
  no ip address
  shutdown
!
interface GigabitEthernet0/0
  description OUTSIDE CUBE Interface
  ip address X.X.X.X 255.255.255.252
  duplex auto
  speed auto
!
interface ISM0/0
  no ip address
  shutdown
  !Application: CUE Running on ISM
!
interface GigabitEthernet0/1
  description INSIDE CUBE Interface
  ip address X.X.X.X 255.255.255.0
  duplex auto
  speed auto
!
interface GigabitEthernet0/2
  no ip address
  shutdown
  duplex auto
  speed auto
!
interface ISM0/1
  description Internal switch interface connected to Internal Service Module
  no ip address
!
interface Vlan1
  no ip address
!
  ip forward-protocol nd
!
  no ip http server
  no ip http secure-server
!
  ip route X.X.X.X 255.255.255.192 Y.Y.Y.Y

3 Specifies that the asymmetric payload support is dual-tone multi-frequency (DTMF) only.
4 To force a Cisco Unified Border Element (Cisco UBE) to send a SIP invite with Early-Offer (EO) on the Out-Leg (OL), use the early-offer command in SIP or dial peer configuration mode.
5 Passes SIP messages that involve media-change from one IP leg to another IP leg.
6 Passes the privacy values from the received message to the next call leg.
! logging trap debugging
logging host X.X.X.X

! control-plane

! sccp local GigabitEthernet0/1
sccp ccm Z.Z.Z.Z identifier 1 version 7.0
sccp

! sccp ccm group 1
bind interface GigabitEthernet0/1
associate ccm 1 priority 1
associate profile 2 register MTP24e9b3735829
associate profile 1 register CFB24e9b3735829

! dspfarm profile 2 transcode
codec g'729hr8
codec g'729r8
codec g'711ulaw
codec g'711law
codec g'729ar8
codec g'729abr8
maximum sessions 3
associate application SCCP

! dspfarm profile 1 conference
codec g'711ulaw
codec g'711law
codec g'729ar8
codec g'729abr8
maximum sessions 3
associate application SCCP

! dial-peer voice 9900 voip
description "IPT-Cert-CUCM-DID-AB"
destination-pattern 58775093[5-9]
signaling forward unconditional
session protocol sipv2
session target ipv4: A.A.A.A

! dial-peer voice 2000 voip
description "TELUS LAB SIP Trunk - NA calls"
preference 1
destination-pattern 1:
signaling forward unconditional
session protocol sipv2
session target ipv4: B.B.B.B

7 Cisco Unified Communication System
8 Tunnels Generic Transparency Descriptor (GTD), payload along with QSIG or Q.931 message bodies.
9 Telus Main SBC IP address
!
!
dial-peer voice 2001 voip
   description "TELUS LAB SIP Trunk - International"
   preference 1
   destination-pattern 011T
   signaling forward unconditional
   session protocol sipv2
   session target ipv4: B.B.B.B
   voice-class codec 1
   voice-class sip g729 annexb all
   voice-class sip early-offer forced
   voice-class sip profiles 150
   dtmf-relay rtp-nte
   fax-relay sg3-to-g3
   fax protocol pass-through g711ulaw
   no vad
!
!
dial-peer voice 2002 voip
   description "TELUS LAB SIP Trunk - Operator"
   preference 1
   destination-pattern 0T
   signaling forward unconditional
   session protocol sipv2
   session target ipv4: B.B.B.B
   voice-class codec 1
   voice-class sip g729 annexb all
   voice-class sip early-offer forced
   dtmf-relay rtp-nte
   fax-relay sg3-to-g3
   fax protocol pass-through g711ulaw
   no vad
!
!
!
! gatekeeper
   shutdown
!
   line con 0
   line aux 0
   line 2
   no activation-character
   no exec
   transport preferred none
   transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
   stopbits 1
   line 131
   no activation-character
   no exec
   transport preferred none
   transport input all
   transport output lat pad telnet rlogin lapb-ta mop udptn v120 ssh
   stopbits 1
   line vty 0 4
   exec-timeout 0 0
   login local
   transport input ssh
!
!
scheduler allocate 20000 1000
ntp source GigabitEthernet0/1
ntp master
!
end
Configuring the Cisco Unified Communications Manager

System Version

Cisco Unified CM Administration
System version: 8.6.2.25140-1

License Warnings:
System is operating on trial license. Please upload relevant license files;
Please visit the License Report Page for more details.
VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU ES-2643 0 @ 3.30GHz, disk 1: 80Gbytes, 4096Mbytes RAM

Last Successful Logon: Dec 3, 2014 11:42:21 AM

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

For Cisco Technical Support please visit our Technical Support web site.

Region

Region Configuration
Region Information
Name: Toronto_Rg

Region Relationships
Region | Max Audio Bit Rate | Max Video Call Bit Rate (Includes Audio) | Link Loss Type
Default | 6 kbps (0.729) | 384 | Use System Default
Toronto_Rg | 8 kbps (0.729) | 384 | Use System Default

Modify Relationship to other Regions
Regions | Max Audio Bit Rate | Max Video Call Bit Rate (Includes Audio) | Link Loss Type
Default | Keep Current Setting | Keep Current Setting | Keep Current Setting
Toronto_Rg | Use System Default | Use System Default | Use System Default

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Device Pool

### Device Pool Configuration

- **Status**: Ready

### Device Pool Information

- **Device Pool**: Toronto (0 members)**

### Device Pool Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Pool Name</td>
<td>Toronto</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Group</td>
<td>Toronto_Group</td>
</tr>
<tr>
<td>Calling Search Space for Auto-registration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Adjunct CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Reverted Call Focus Priority</td>
<td>Default</td>
</tr>
<tr>
<td>Local Route Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Intercompany Media Services Enrolled Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Roaming Sensitive Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Date/Time Group</td>
<td>Toronto_Time</td>
</tr>
<tr>
<td>Region</td>
<td>Toronto_Rg</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Network Locale

- **Network Locale**: < None >

### SRST Reference

- **SRST Reference**: Disable

### Connection Monitor Duration

- **Connection Monitor Duration**: Default

### Single Button Barge

- **Single Button Barge**: Default

### Join Across Lines

- **Join Across Lines**: Default

### Physical Location

- **Physical Location**: < None >

### Device Mobility Group

- **Device Mobility Group**: < None >

### Device Mobility Related Information

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

### Geolocation Configuration

- **Geolocation**: < None >
- **Geolocation Filter**: < None >
## Call Routing Information

### Incoming Calling Party Settings

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

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

### Incoming Called Party Settings

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

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Default Prefix Settings</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>0</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>0</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td>Default</td>
<td></td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>0</td>
<td>Default</td>
<td></td>
</tr>
</tbody>
</table>

### Connected Party Settings

Connected Party Transformation CSS: None

### Redirecting Party Settings

Redirecting Party Transformation CSS: None
## Trunk Configuration

### Device Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Product</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>IPT-Cert-CUBE</td>
</tr>
<tr>
<td>Description</td>
<td>IPT-Cert-CUBE</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunneled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
</tbody>
</table>

- Media Termination Point Required
- Retry Video Call as Audio
- Path Replacement Support

- Transmit UTF-8 for Calling Party Name
- Transmit UTF-8 Names in QSIG APDU
- Unattended Port
- SRTP Allowed: When this flag is checked, encrypted TLS needs to be configured in the network to provide end-to-end security. Failure Consider Traffic on This Trunk Secure
- Route Class Signaling Enabled
- Use Trusted Relay Point
- PSTN Access
- Run On All Active Unified CM Nodes

### Intercompany Media Engine (IME)

| E.164 Transformation Profile   | < None >                |

### Multilevel Precedence and Preemption (MLPP) Information

| MLPP Domain                   | < None >                |

### Call Routing Information

- Remote-Party-Id
- Asserted-Identity
- Asserted-Type (TAP)
### Inbound Calls

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits</td>
<td>3</td>
</tr>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td>1</td>
</tr>
</tbody>
</table>

### Incoming Calling Party Settings

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

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

### Outbound Calls

- **Called Party Transformation CSS**: < None >
- **Use Device Pool Called Party Transformation CSS**
- **Calling Party Transformation CSS**: < None >
- **Use Device Pool Calling Party Transformation CSS**
- **Calling Party Selection**: Originator
- **Calling Line ID Presentation**: Default
- **Calling Name Presentation**: Default
- **Caller ID DN**: 
- **Caller Name**: 
- **Redirecting Diversion Header Delivery - Outbound**
- **Redirecting Party Transformation CSS**: < None >
- **Use Device Pool Redirecting Party Transformation CSS**

### SIP Information

- **Destination**
  - **Destination Address is an SRV**
SIP PRACK for early-media negotiation

Telus Mobility requirement for compatibility with Cisco CUCM is to modify the service parameters related to two timers and enable the PRACK on SIP trunk profile in order to acknowledge SDP messages.
### Trunk Specific Configuration

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reroute Incoming Request to new Trunk based on</td>
<td>Never</td>
</tr>
<tr>
<td>RSVP Over SIP</td>
<td>Local RSVP</td>
</tr>
<tr>
<td>Fall back to local RSVP</td>
<td></td>
</tr>
<tr>
<td>SIP Req1XX Options</td>
<td>Send PRACK if 1xx Contains SDP</td>
</tr>
<tr>
<td>Deliver Conference Bridge Identifier</td>
<td></td>
</tr>
<tr>
<td>Early Offer support for voice and video calls (insert MTP if needed)</td>
<td></td>
</tr>
<tr>
<td>Send send-receive SDP in mid-call INVITE</td>
<td></td>
</tr>
<tr>
<td>Allow Presentation Sharing using BFCP</td>
<td></td>
</tr>
</tbody>
</table>

### Route Group (SIP Trunk)

#### Route Group Configuration

- Save
- Delete
- Add New
- Start/Stop Ready

#### Route Group Information

- **Route Group Name**: RG-Cert
- **Distribution Algorithm**: Circular

#### Route Group Member Information

- **Find Devices to Add to Route Group**
  - **Device Name contains**: IPT-Cert-CUBE
  - **Port(s)**: None Available

- **Current Route Group Members**
  - **Selected Devices (ordered by priority)**: IPT-Cert-CUBE (All Ports)
  - **Removed Devices**: IPT-Cert-CUBE
- **Reverse Order of Selected Devices**

### Route Group Members

- **IPT-Cert-CUBE**
Route List (SIP Trunk)

- **Route List Information**
  - Registration:
  - IPv4 Address:
  - Device is trusted
  - Name:
  - Description:
  - Cisco Unified Communications Manager Group:
  - Enable this Route List (change effective on Save; no reset required)
  - Run On All Active Unified CM Nodes

- **Route List Member Information**

- **Route List Details**
  - RG-Cert

- **Route Pattern (SIP Trunk)**

Please note that Digits 9. Predot will be discarded on all patterns on the outgoing SIP trunk. 9 was used only for route selection.
## Route Pattern Configuration

### Pattern Definition

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Description</td>
<td>NA Calls</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>NLPP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td></td>
</tr>
<tr>
<td>Route Class 8</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List 8</td>
<td>R2-Crop</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Class 7</td>
</tr>
</tbody>
</table>

### Calling Party Transformations

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Calling Party’s External Phone Number Mask</td>
<td>Yes</td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Calling Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>
### Phone Configuration

**Status**
- Update successful

#### Association Information

<table>
<thead>
<tr>
<th>Line 1 (11-1935) [no partition]</th>
<th>Modify Button Items</th>
</tr>
</thead>
<tbody>
<tr>
<td>Line 2 - Add a new DN</td>
<td></td>
</tr>
<tr>
<td>Line 3 - Add a new SD</td>
<td></td>
</tr>
<tr>
<td>Line 4 - Add a new SD</td>
<td></td>
</tr>
<tr>
<td>Line 5 - Add a new SD</td>
<td></td>
</tr>
<tr>
<td>Line 6 - Add a new SD</td>
<td></td>
</tr>
<tr>
<td>Line 7 - Unassigned Associated Items</td>
<td></td>
</tr>
<tr>
<td>Line 8 - Add a new BLF SD</td>
<td></td>
</tr>
<tr>
<td>Line 9 - Add a new BLF SD</td>
<td></td>
</tr>
</tbody>
</table>

#### Phone Type
- **Product Type:** Cisco 7965
- **Device Protocol:** SCCP

#### Device Information
- Registered with Cisco Unified Communications Manager CUCMv6.1-PUB
- SCCP: 15.5-3-1584-15
- Device ID: 263407F7D1B20
- IPT-Cert-Phone: L

- **Device Pool:** Default
- **Common Device Configuration:** None
- **Phone Button Template:** Standard 7965 SCCP
- **Softkey Template:** None

#### Common Phone Profile
- Calling Search Space
- AAR Calling Search Space
- Media Resource Group List
- User Hold MCH Audio Source
- Network Hold MCH Audio Source
- Location
- AAR Group
- User Locale
- Network Locale
- Built-In Bridge
- Privacy
- Device Mobility Mode
- Owner User ID
- Phone Personalization
- Services Provisioning
- Phone Load Name

#### Standard Common Phone Profile
- Default

### Status Bar
- Save
- Delete
- Copy
- Reset
- Apply Config
- Add New

---

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### Quality Reporting Tool
- 28: Quality Reporting Tool
- 29: Radial
- 30: Remove Last Participant
- 31: Transfer
- 32: Video Mode
- 33: Privacy
- 34: None

### Use Trusted Relay Point
- Default

### BLF Audible Alert Setting (Phone Idle)
- Default

### BLF Audible Alert Setting (Phone Busy)
- Default

### Always Use Prime Line
- Default

### Always Use Prime Line for Voice Message
- Default

### Calling Party Transformation CSS
- < None >

### Geolocation
- < None >

### Use Device Pool Calling Party Transformation CSS

### Retry Video Call as Audio

### Ignore Presentation Indicators (internal calls only)

### Allow Control of Device from CTI

### Logged Into Hunt Group

### Remote Device

### Protected Device

### Hot line Device

### Protocol Specific Information
- Packet Capture Mode: None
- Packet Capture Duration: 0

### Presence Group
- Standard Presence group

### Device Security Profile
- Cisco 7965 - Standard SCCP Non-Secure Profile

### SUBSCRIBE Calling Search Space
- < None >

### Unattended Port
- Require DTMF Reception
- RFC2833 Disabled

### Certification Authority Proxy Function (CAPF) Information
- Certificate Operation: No Pending Operation
- Authentication Mode: By Null String
- Authentication String

### Key Size (Bits)
- 1024

### Operation Completes By
- 2014 12 21 12 (YYYY:MM:DD:HH)

### Certificate Operation Status: None

Note: Security Profile Contains Additional CAPF Settings.

### Expansion Module Information
- Module 1: < None >
## Module 2 Load Name

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

### External Data Locations Information (Leave blank to use default)

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

### Extension Information

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

### MLPP Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>MLPP Domain</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>MLPP Indication</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>MLPP Preemption</strong></td>
<td>Default</td>
</tr>
</tbody>
</table>

### Do Not Disturb

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

Secure Shell User: 
Secure Shell Password: 

## Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Feature</th>
<th>Param</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td>Disabled</td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td>Disabled</td>
</tr>
<tr>
<td>Forwarding Delay*</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Settings Access*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Gratuitous ARP*</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Voice VLAN Access*</td>
<td>Enabled</td>
</tr>
<tr>
<td>Video Capabilities*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Auto Line Select*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Web Access*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Days Display Not Active</td>
<td>Sunday</td>
</tr>
</tbody>
</table>

### Additional Settings

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display On Time</td>
<td>07:30</td>
</tr>
<tr>
<td>Display On Duration</td>
<td>10:30</td>
</tr>
<tr>
<td>Display Idle Timeout</td>
<td>01:00</td>
</tr>
<tr>
<td>Enable Power Save Plus</td>
<td>Sunday</td>
</tr>
<tr>
<td>Phone On Time</td>
<td>00:00</td>
</tr>
<tr>
<td>Phone Off Time</td>
<td>24:00</td>
</tr>
<tr>
<td>Phone Off Idle Timeout*</td>
<td>60</td>
</tr>
<tr>
<td>Enable Audible Alert</td>
<td></td>
</tr>
<tr>
<td>EnergyWise Domain</td>
<td></td>
</tr>
<tr>
<td>EnergyWise Endpoint Security Secret</td>
<td></td>
</tr>
<tr>
<td>Allow EnergyWise Overrides</td>
<td></td>
</tr>
<tr>
<td>Span to PC Port*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Logging Display*</td>
<td>PC Controlled</td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
</tr>
<tr>
<td>Recording Tone*</td>
<td>Disabled</td>
</tr>
<tr>
<td>Recording Tone Local Volume*</td>
<td>100</td>
</tr>
<tr>
<td>Recording Tone Remote Volume*</td>
<td>50</td>
</tr>
<tr>
<td>Feature</td>
<td>Value</td>
</tr>
<tr>
<td>----------------------------------------------</td>
<td>---------</td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
</tr>
<tr>
<td>Display On When Incoming Call</td>
<td>Disabled</td>
</tr>
<tr>
<td>RTCP</td>
<td>Disabled</td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>5</td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Enabled</td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Advertise G.722 Codec</td>
<td></td>
</tr>
<tr>
<td>Wideband Headset UI Control</td>
<td>Enabled</td>
</tr>
<tr>
<td>Wideband Headset</td>
<td>Enabled</td>
</tr>
<tr>
<td>Peers Firmware Sharing</td>
<td>Enabled</td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port</td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port</td>
<td></td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td>Unknown</td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td></td>
</tr>
<tr>
<td>Wireless Headset Hookswitch Control</td>
<td></td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>IPv6 Log Server</td>
<td></td>
</tr>
<tr>
<td>802.1x Authentication</td>
<td></td>
</tr>
<tr>
<td>Detect Unified CM Connection Failure</td>
<td></td>
</tr>
<tr>
<td>Minimum Ring Volume</td>
<td></td>
</tr>
<tr>
<td>Headset Sidetone Level</td>
<td></td>
</tr>
<tr>
<td>Headset Send Gain</td>
<td></td>
</tr>
<tr>
<td>HTTPS Server</td>
<td></td>
</tr>
<tr>
<td>Handset/Headset Monitor</td>
<td></td>
</tr>
<tr>
<td>Handset Recording</td>
<td></td>
</tr>
<tr>
<td>Enbloc Dialing</td>
<td></td>
</tr>
<tr>
<td>Switch Port Remote Configuration</td>
<td></td>
</tr>
<tr>
<td>PC Port Remote Configuration</td>
<td></td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td></td>
</tr>
<tr>
<td>SSH Access</td>
<td></td>
</tr>
<tr>
<td>LOGIN Access</td>
<td></td>
</tr>
<tr>
<td>FIPS Mode</td>
<td></td>
</tr>
<tr>
<td>80-bit SRTCP</td>
<td></td>
</tr>
</tbody>
</table>
IP phone DN configuration

Directory Number Configuration

Status
Status: Ready

Directory Number Information
Directory Number 1935
Route Partition < None >
Description IPT-Cert Phone1-Line-1-IPT
Alerting Name IPT-Cert-Phone1-Line-1-IPT
ASCII Alerting Name IPT-Cert-Phone1-Line-1-IPT
Allow Control of Device from CTI
Associated Devices 604685834a1
SEP203A07FD1B29

Dissociate Devices

Directory Number Settings
Name Profile CUC_VM_Profile
Calling Search Space < None >
Presence Group Standard Presence group
User Hold MOH Audio Source < None >
Network Hold MOH Audio Source < None >
Auto Answer Auto Answer Off

AAR Settings

<table>
<thead>
<tr>
<th>AAR</th>
<th>Voice Mail</th>
<th>AAR Destination Mask</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

| Retain this destination in the call forwarding history |

Call Forward and Call Pickup Settings

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space Activation Policy</td>
<td></td>
</tr>
<tr>
<td>Forward All</td>
<td>or</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td></td>
</tr>
</tbody>
</table>

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Forward Busy Internal ✓ or
Forward Busy External ✓ or
Forward No Answer Internal ✓ or
Forward No Answer External ✓ or
Forward No Coverage Internal □ or
Forward No Coverage External □ or
Forward on CTI Failure □ or
Forward Unregistered Internal ✓ or
Forward Unregistered External ✓ or

No Answer Ring Duration (seconds) 8
Call Pickup Group < None >

---

**Park Monitoring**

<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Park Monitoring Forward No Retrieve
Destination External □ or

Park Monitoring Forward No Retrieve
Destination Internal □ or

Park Monitoring Reversion Timer

---

**MLPP Alternate Party Settings**

Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)

---

**Line Settings for All Devices**

Hold Reversion Ring Duration (seconds)
Hold Reversion Notification Interval (seconds)
Party Entrance Tone* Default

---

**Line 1 on Device SEP203A07FD1B29**

Display (Internal Caller ID) IPT-Cert-Phone1-Line-1-IPT
ASCII Display (Internal) IPT-Cert-Phone1-Line-1-IPT

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<table>
<thead>
<tr>
<th>Line Text Label</th>
<th>1935</th>
</tr>
</thead>
<tbody>
<tr>
<td>ASCII Line Text Label</td>
<td>1935</td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td>5877569935</td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy</td>
<td>Use System Policy</td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy</td>
<td>Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Idle)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Recording Option*</td>
<td>Call Recording Disabled</td>
</tr>
<tr>
<td>Recording Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Monitoring Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>✔ Log Missed Calls</td>
<td></td>
</tr>
</tbody>
</table>

### Multiple Call/Call Waiting Settings on Device SEP203A07FD1B29

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

<table>
<thead>
<tr>
<th>Maximum Number of Calls*</th>
<th>4</th>
</tr>
</thead>
<tbody>
<tr>
<td>Busy Trigger*</td>
<td>1</td>
</tr>
</tbody>
</table>

### Forwards Call Information Display on Device SEP203A07FD1B29

- ✔ Caller Name
- □ Caller Number
- □ Redirected Number
- ✔ Dialed Number

### Users Associated with Line

[Associate End Users]

- Save
- Delete
- Reset
- Apply Config
- Add New
Privacy configuration

CUCM configuration privacy

Calling Line and Calling Name should be set to Restricted, this will enable the privacy: id

<table>
<thead>
<tr>
<th>Outbound Calls</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
<td>✓</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>✓</td>
</tr>
<tr>
<td>Calling Party Selection*</td>
<td>Originator</td>
</tr>
<tr>
<td>Calling Line ID Presentation*</td>
<td>Restricted</td>
</tr>
<tr>
<td>Calling Name Presentation*</td>
<td>Restricted</td>
</tr>
<tr>
<td>Calling and Connected Party Info Format*</td>
<td>Deliver DN only in connected party</td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Outbound</td>
<td></td>
</tr>
<tr>
<td>Redirecting Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Use Device Pool Redirecting Party Transformation CSS</td>
<td>✓</td>
</tr>
</tbody>
</table>

CUBE configuration privacy

SIP header should be modified in order to replace the anonymous with valid information as the SBC expect valid information in the INVITE header. If a valid header is not present then the SBC will use the Parent DN when the call is sent out to other switches (for tracking purposes). PAI header is being used for billing and routing on the SBC.

```
voice class sip-profiles 150
request INVITE sip-header From modify "Anonymous" "IPT-Cert-Phone3-Line:1"
request INVITE sip-header From modify "<sip:anonymous@anonymous.invalid>" "<sip:+15877569937@172.24.0.120>"
dial-peer voice 2000 voip
   voice-class sip profiles 150
```

Configuring the Cisco Unity Connection

System Version

![Cisco Unity Connection Administration](image)
### User configuration

<table>
<thead>
<tr>
<th>Alias</th>
<th>Extension</th>
<th>First Name</th>
<th>Last Name</th>
<th>Display Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>admin</td>
<td></td>
<td></td>
<td></td>
<td>admin</td>
</tr>
<tr>
<td>ashley</td>
<td>1936</td>
<td>Ashley</td>
<td>Williams</td>
<td>Ashley Williams</td>
</tr>
<tr>
<td>gemus</td>
<td>1935</td>
<td>Gennus</td>
<td>Vakaran</td>
<td>Gennus Vakaran</td>
</tr>
<tr>
<td>james</td>
<td>1927</td>
<td>James</td>
<td>Vega</td>
<td>James Vega</td>
</tr>
<tr>
<td>operator</td>
<td>99990</td>
<td></td>
<td></td>
<td>Operator</td>
</tr>
<tr>
<td>replication</td>
<td></td>
<td>Replication</td>
<td>Agent</td>
<td>Replication Agent (ccw)</td>
</tr>
<tr>
<td>teli</td>
<td>1930</td>
<td>Teli</td>
<td>Zoroh</td>
<td>Teli Zoroh</td>
</tr>
<tr>
<td>undeliverablemessagesmailbox</td>
<td>99999</td>
<td></td>
<td></td>
<td>Undeliverable Messages</td>
</tr>
<tr>
<td>UnityConnection</td>
<td></td>
<td></td>
<td></td>
<td>Cisco Unity Connection</td>
</tr>
</tbody>
</table>

Rows per Page: 25
### Call Handler for Auto Attendant

<table>
<thead>
<tr>
<th>Call Handler</th>
<th>Edit</th>
<th>Refresh</th>
<th>Help</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="51x719.png" alt="Image" /></td>
<td><img src="131x764.png" alt="Image" /></td>
<td><img src="54x340.png" alt="Image" /></td>
<td><img src="522x668.png" alt="Image" /></td>
</tr>
</tbody>
</table>

#### Status

⚠️ It is recommended you backup report data prior to renaming the Call Handler Display Name.

#### Call Handler

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display Name</td>
<td>AA</td>
</tr>
<tr>
<td>Creation Time</td>
<td>2014-09-18 14:34:02.983</td>
</tr>
<tr>
<td>Phone System</td>
<td>PhoneSystem</td>
</tr>
<tr>
<td>Active Schedule</td>
<td>All Hours</td>
</tr>
<tr>
<td>Use System</td>
<td>Default Time Zone</td>
</tr>
<tr>
<td>Time Zone</td>
<td>(GMT-08:00) America/Vancouver</td>
</tr>
<tr>
<td>Language</td>
<td>Inherit Language from Caller</td>
</tr>
<tr>
<td>Extension</td>
<td>3334</td>
</tr>
<tr>
<td>Partition</td>
<td>CUC10 Partition</td>
</tr>
<tr>
<td>Recorded Name</td>
<td>Play/Record</td>
</tr>
</tbody>
</table>

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Call Routing for user extension

Cisco Unity Connection Ports

Port (1 - 2 of 2)
### Cisco Unity Connection SIP Trunk

#### Trunk Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Device Information</strong></td>
<td></td>
</tr>
<tr>
<td>Product</td>
<td>SIP Trunk</td>
</tr>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Trunk Service Type</td>
<td>None (Default)</td>
</tr>
<tr>
<td>Device Name</td>
<td>CUC_SIP_Trunk</td>
</tr>
<tr>
<td>Description</td>
<td>SIP trunk for Cisco Unity Connection</td>
</tr>
<tr>
<td>Device Pool</td>
<td>Default</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_None</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Tunneled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>No Changes</td>
</tr>
<tr>
<td>ASN.1 ROSE OID Encoding</td>
<td>No Changes</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>None</td>
</tr>
</tbody>
</table>

#### Packet Capture Duration

- [ ] Media Termination Point Required
- [ ] Retry Video Call as Audio
- [ ] Path Replacement Support
- [ ] Transmit UTF-8 for Calling Party Name
- [ ] Transmit UTF-8 Names in QSIG APDU
- [ ] Unattended Port

#### Intercompany Media Engine (IME)

- [ ] PSTN Access
- [ ] Run On All Active Unified CM Nodes

#### E.164 Transformation Profile

- [ ] < None >

#### Multilevel Precedence and Preemption (MLPP) Information

- [ ] < None >
### Call Routing Information

- Remote-Party-Id
- Asserted-Identity

**Asserted-Type***
- Default

**SIP Privacy***
- Default

### Inbound Calls

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits</td>
<td>3</td>
</tr>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td>1</td>
</tr>
</tbody>
</table>

- Redirecting Diversion Header Delivery - Inbound

### Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level unless the field is empty in which case there is no prefix assigned.

### Connected Party Settings

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

### Outbound Calls

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Called Party Transformation CSS</td>
<td><code>&lt; None &gt;</code></td>
</tr>
<tr>
<td>Use Device Pool Called Party Transformation CSS</td>
<td>✔️</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td><code>&lt; None &gt;</code></td>
</tr>
<tr>
<td>Use Device Pool Calling Party Transformation CSS</td>
<td>✔️</td>
</tr>
<tr>
<td>Calling Party Selection</td>
<td>Originator</td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>

- Redirecting Diversion Header Delivery - Outbound
- Redirecting Party Transformation CSS: `< None >`
- Use Device Pool Redirecting Party Transformation CSS

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**Configuring Cisco Voice Gateway VG204XM**

**Show Version**
Cisco IOS Software, VG20XXM Software (VG20XXM-IPVOICE-M), Version 15.3(2)T, RELEASE SOFTWARE (fc3)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2013 by Cisco Systems, Inc.
Compiled Thu 28-Mar-13 14:21 by prod_rel_team

ROM: System Bootstrap, Version 12.4(20r)YA2, RELEASE SOFTWARE (fc1)

VG204XM uptime is 1 hour, 1 minute
System returned to ROM by power-on
System image file is "flash:vg20xxm-ipvoice-mz.153-2.T.bin"
Last reload type: Normal Reload
Last reload reason: power-on

Cisco VG204XM (MPC8300) processor (revision 0x100) with 249856K/12288K bytes of memory.
Processor board ID FCH1807S0N8
MPC8300 CPU Rev: Part Number 0x8062, Revision ID 0x11
2 FastEthernet interfaces
4 Voice FXS interfaces
256K bytes of non-volatile configuration memory.
125496K bytes of ATA CompactFlash (Read/Write)

**Show Running-Configuration**
Building configuration...

Current configuration : 1942 bytes

Last configuration change at 18:47:02 UTC Sun Mar 3 2002
version 15.3
no service pad
service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
!
hostname VG204XM
!
boot-start-marker
boot-end-marker
!
enable secret 5 YYYYY
enable password YYYYY
!
no aaa new-model
!
ip domain name ipt.local
ip cef
no ipv6 cef
!
voice-card 0
!
interface FastEthernet0/0
ip address dhcp
duplex auto
speed auto
!
interface FastEthernet0/1
ip address dhcp
shutdown
speed auto
half-duplex
!
ip forward-protocol nd
!
no ip http server
!
control-plane
!
voice-port 0/0
cptone CA
!
voice-port 0/1
cptone CA
!
voice-port 0/2
cptone CA
!
voice-port 0/3
cptone CA
!
ccm-manager redundant-host Y.Y.Y.Y
ccm-manager mgcp
no ccm-manager fax protocol cisco
ccm-manager music-on-hold
ccm-manager config server X.X.X.X
ccm-manager config
!
mgcp
mgcp call-agent X.X.X.X 2427 service-type mgcp version 0.1
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp package-capability rtp-package
mgcp package-capability sip-package
no mgcp package-capability res-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp bind control source-interface FastEthernet0/0
mgcp bind media source-interface FastEthernet0/0
mgcp behavior rtp-range sgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 1 pots
  service mgcapp
  port 0/0
!
dial-peer voice 2 pots
  service mgcapp
  port 0/1
!
dial-peer voice 3 pots
  service mgcapp
  port 0/2
!
dial-peer voice 4 pots
  service mgcapp
  port 0/3
!
line con 0
  no modem enable
line aux 0
line vty 0 4
  password YYYY
login
transport input all
!
end

Fax mode
Pass-through mode

Dial-peer or global configuration: fax protocol pass-through g711ulaw
## Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CODEC</td>
<td>Coder-Decoder (in this document a device used to digitize and un-digitize voice signals)</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>DN</td>
<td>Directory Number</td>
</tr>
<tr>
<td>IP</td>
<td>Internet Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>MPLS</td>
<td>Multiprotocol Label Switching</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public switched telephone network</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>SP</td>
<td>Service Provider</td>
</tr>
<tr>
<td>TDM</td>
<td>Time-division multiplexing</td>
</tr>
<tr>
<td>VG</td>
<td>Voice Gateway</td>
</tr>
<tr>
<td>VPN</td>
<td>Virtual Private Network</td>
</tr>
</tbody>
</table>
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### Corporate Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

### European Headquarters
Cisco Systems International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

### Americas Headquarters
Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

### Asia Pacific Headquarters
Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

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