Bell Canada SIP Trunking Service: Connecting Cisco Unified Communications Manager 10.5 via the Cisco Unified Border Element 10.5 [IOS 15.4(3)M] using SIP

December 2014

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Note: Testing was conducted in (Bell Canada) labs
Introduction

Service Providers today, such as Bell Canada, 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. Bell Canada 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) 10.5.1 with a Cisco Unified Border Element (CUBE) for connectivity to Bell Canada SIP Trunking Service SIP trunk service. The deployment model covered in this application note is Cisco CUCM 10.5.1 / CUBE to PSTN Bell Canada SIP trunk service. This document does not address 911 emergency outbound calls. For 911 feature service details contact Bell Canada directly.

- Testing was performed in accordance to Bell Canada SIP Trunking Test plan and all features were verified. Key features verified are:
  - Inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, DNIS translations, CODEC negotiation, advanced 8XX call prompter, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), auto-attendant, fax using G.711 (G3 and SG3 speeds), teleconferencing, Simultaneous and Sequential Ring, failover of unresponsive SIP network to PSTN and outbound/inbound calls to/from TDM networks.
  - 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 Bell Canada’s 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 interoperating with Bell Canada’s 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

Figure 1. Basic Call Setup

System Components

Hardware Components

Hardware Components

- CUCM V 10.5.1
- CUBE 10.5
- VG224

Software Requirements

Bell service is compatible with CUCM v10.5.1 + CUBE on Cisco IOS Software, C2900 Software (c2900-universalk9-mz.SPA.154-3.M.bin), Version 15.4(3) M, and above. The following list summarizes the version and patch levels of the hardware that has been validated as compatible in the Bell lab:

CUCM v 10.5.1
CUBE on IOS 15.4(3) M
Features

Features Supported

- Basic Call using G.729 or G.711ulaw
- Calling Party Number Presentation and Restriction
- Calling Name
- Intra-site Call Transfer
- Intra-site Conference
- Advanced 8XX Call Prompter
- Call Hold and Resume
- Call Forward All, Busy and No Answer
- Fax using G.711 pass-through
- CUBE: performs Early-Offer-to initial SIP INVITE with SDP
- Incoming DNIS Translation and Routing
- Outbound calls to IP and TDM networks
- Auto-attendant transfer-to service
- CPE voicemail managed service (Cisco Unity Connection), leave and retrieve voice message via incoming Bell SIP Trunk
- Auto-attendant transfer-to service
- Network-based Call Forward Unconditional, Busy, No Answer and Not Reachable

Features Not Supported

- T.38 Fax protocol
- Modem transmissions
Caveats

- It is recommended to have a transponder resource if the customer network will support more than one codec.
- When a PSTN to CPE call is transferred by the CPE to a second PSTN number, the Caller ID displayed on the transfer target is the CPE DID number. It does not update to the original PSTN calling party number when the transfer is completed.
- BELL SIP Trunking Service only supports G711ulaw codec for FAX calls. G.711 fax passthrough is the supported FAX transport method for the BELL SIP Trunking Service. See configuration section for details.
Configuration

Configuring Cisco Unified Border Element (CUBE)

This section covers the base configuration of the customer CUBE required to set it up as the customers interface for Bell Canada's SIP Trunking service. The CUBE acts as the mediation point between the customer’s CUCM PBX and the session border controllers that provide access to Bell Canada’s SIP Trunking Service.

The “show version” and “show running config” output from the CUBE in Bell’s verification lab is shown below for reference. The running configuration excerpt has been truncated and abbreviated for clarity, and explanatory annotations have been added to explain the intent of each important entry.

**NOTE:** The recommended version for the CISCO IOS load on the customer CUBE is 15.4(3) M or later. Earlier versions might not support some of the configuration entries displayed below. Pay special attention to the bolded entries, and their annotation numbers, which map to the explanatory footnotes at the end of the configuration. Extraneous empty lines (marked with “!”) have been added to the excerpt to keep configuration sections in one piece for ease of reading.

```
BITS_SIP_TRUNK_2921#show version
Cisco IOS Software, C2900 Software (C2900-UNIVERSALK9-M), Version 15.4(3)M, RELEASE SOFTWARE (fc1)
Technical Support: http://www.cisco.com/techsupport
Copyright (c) 1986-2014 by Cisco Systems, Inc.

ROM: System Bootstrap, Version 15.0(1r)M9, RELEASE SOFTWARE (fc1)

BITS_SIP_TRUNK_2921 uptime is 7 weeks, 3 days, 31 minutes
System returned to ROM by reload at 11:15:42 EDT Thu Oct 9 2014
System image file is "flash:c2900-universalk9-mz.SPA.154-3.M.bin"
Last reload type: Normal Reload
Last reload reason: Reload Command

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

A summary of U.S. laws governing Cisco cryptographic products may be found at: http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

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

Cisco CISCO2921/K9 (revision 1.0) with 1519616K/53248K bytes of memory.
Processor board ID FHK1448F0UZ
6 Gigabit Ethernet interfaces
3 terminal lines
1 Virtual Private Network (VPN) Module
1 Internal Services Module (ISM) with Services Ready Engine (SRE)
DRAM configuration is 64 bits wide with parity enabled.
255K bytes of non-volatile configuration memory.
2048256K bytes of ATA System CompactFlash 0 (Read/Write)
```

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voice-card 0
dspfarm
dsp services dspfarm
!
voice service voip
address-hiding
mode border-element
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
fax protocol pass-through g711ulaw
sip
asserted-id pai
localhost dns:simcoe-siptcube.on.bell.ca
options-ping 60
early-offer forced
midcall-signaling passthru
privacy-policy passthru
g729 annexb-all
!
voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
!
voice class sip-profiles 416
request INVITE sip-header To modify "@.*>" "@siptrunking.bell.ca>"
request INVITE sip-header From modify "(@.*)>" "@simcoe-siptcube.on.bell.ca;user=phone>"
request INVITE sip-header Diversion modify "<sip:1.(.*)@(..*)>" "<sip:1@simcoe-siptcube.on.bell.ca;user=phone>"
request INVITE sip-header SIP-Req-URI modify "10.220.254.228" "siptrunking.bell.ca"
request INVITE sip-header P-Asserted-Identity modify "(@.*)>" "@simcoe-siptcube.on.bell.ca;user=phone>"
request INVITE sip-header Contact modify "@" ;tgrp=simcoe_01_4163532154_admin;trunk-context=siptrunking.bell.ca"

! voice translation-rule 1
rule 1 /\^9/ //

! voice translation-rule 2
rule 1 /\^\+1/ //

! voice translation-profile remove+1
 translate calling 2

! voice translation-profile remove9
 translate called 1

! sccp local GigabitEthernet0/2
 sccp ccm 10.220.22.161 identifier 1 version 7.0
 sccp

! sccp ccm group 1
 bind interface GigabitEthernet0/2
 associate ccm 1 priority 1
 associate profile 2 register Transcoder
 associate profile 1 register Conference

! !

! dspfarm profile 2 transcode
 codec g729abr8
 codec g729ar8
 codec g711alaw
 codec g711ulaw
 maximum sessions 20
 associate application SCCP

! dspfarm profile 1 conference
codec g729br8
codec g729r8
codec g729abr8
codec g729ar8
codec g711alaw
codec g711ulaw
maximum sessions 4
associate application SCCP
!
dial-peer voice 41635321 voip
description outgoing leg to SIP trunk for default trunk
translation-profile outgoing remove9
destination-pattern 9T
session protocol sipv2
session target ipv4:10.220.254.228
voice-class codec 1
no voice-class sip localhost
voice-class sip profiles 416
voice-class sip options-keepsalive down-interval 20
dtmf-relay rtp-nr
!
dial-peer voice 41635322 voip
description outgoing leg to SIP trunk for Load Balancer
translation-profile outgoing remove9
preference 1
destination-pattern 9T
session protocol sipv2
session target ipv4:10.220.254.231
voice-class codec 1
no voice-class sip localhost
voice-class sip profiles 416
voice-class sip options-keepsalive down-interval 20
!
dial-peer voice 41635323 voip
description outgoing leg to SIP trunk for Load Balancer
translation-profile outgoing remove9
preference 2
destination-pattern 9T
session protocol sipv2
session target ipv4:10.220.254.234
voice-class codec 1
no voice-class sip localhost
voice-class sip profiles 416
voice-class sip options-keepalive down-interval 20
dtmf-relay rtp-nte
!
dial-peer voice 416353 voip
description outgoing leg to CUCM-10.5 for 41635321..
translation-profile incoming remove+1
answer-address +1T
destination-pattern 41635321..
session protocol sipv2
session target ipv4:10.220.22.161
voice-class codec 1
dtmf-relay rtp-nte
!
sip-ua
keepalive target dns:simcoe-siptcube.on.bell.ca
authentication username Bell password 7 123B001B1E5A5BB7E
no remote-party-id
retry invite 2
retry response 3
retry bye 3
retry prack 6
<<CUBE Configuration excerpt END>>
Configuration Annotations:

- These settings are global SIP settings. Depending on your specific requirements, these parameters may also be provisioned specifically in dial-peer entries for more refined control granularity.

- The company domain agreed upon during service setup with Bell Canada. This parameter is used in the From: and P-Asserted-Identity: headers as the host part of the address URI.

- “early-offer-forced” is used to force an early offer of media codec support (via SDP message body) in outgoing INVITE messages. Bell Canada SIP Trunking requires this early offer.

- This privacy-policy setting (passthru) is used to indicate that the session target is trusted and it will maintain privacy requests. This setting is new in IOS version 15 for the CUBE.

- Voice codec class setup, this example shows that both G711ulaw and g729 are defined for this class. Use of this voice class will supply both codec in the offer during media negotiation.

- Modification of Diversion header to allow for Call Forwarding from CUCM v10.5.1. The modification simply replaces the host portion of the URI with the Bell Canada SIP Trunking service domain. It also adds the required (mandatory) “user=phone” parameter.

- The contact header is modified to supply a tgrp parameter (SIP RFC 4904) to select a specific trunk for outbound calls.

- A further addition of mandatory user=phone parameter to the From header.

- A further addition of mandatory user=phone parameter to the P-Asserted Identity header.

- Specifies the use of the codec class defined in “codec class 1”

- Specifies the sip profile to use. “The specified profile provides trunk selection via tgrp specification.

- Specifies DTMF signaling type for SIP Trunking service.

- This Dial peer is used as a default outgoing dial peer for outgoing calls.

- Another Dial peer Specifies the address of the customer’s CUCM v10.5.1 PBX. This session target is the target to be used for inbound calls, which obviously need to terminate on the customer’s CUCM.

- This Dial peer is used as a default incoming dial peer from the customer’s SIP Trunking service. The E.164 prefix “+1” will be stripped before sending the call to the CUCM v10.5.1.

- Specifies the DIGEST credentials to be used for authentication challenges to outbound SIP requests. These credentials are provided by Bell Canada.

- Bell Canada SIP Trunking only supports UDP transport.
Configuring the Cisco Unified Communications Manager

This section covers the base configuration of the CUCM to communicate with a Cisco Unified Border Element (CUBE). The CUBE acts as the customer interface to Bell Canada’s SIP Trunking Service.

The customer administrator will use the CUCM Administration Web interface to implement the following CUCM configuration examples:

**Configure a SIP Profile**

Navigate to Device ➔ Device Settings ➔ SIP Profile

- Press the Find button to display existing SIP Profile entries.
- Copy the existing Default SIP Profile using the Copy icon on the right hand side (See NOTE below).
- Name the New SIP Profile.
- There are no changes made to the settings of the CUCM Default SIP Profile for the CUCM SIP Profile.
- Save the configuration

**NOTE:** Cisco strongly suggests that each configuration profile component be created as a copy of the default version, rather than using the default version as is, or altering the default profile and re-saving it. The default profile provides a clean base reference point for future profiles, or to revert to in the event that a configuration becomes corrupted.
### Do Not Disturb Configurations

<table>
<thead>
<tr>
<th>Configuration</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>User</td>
<td></td>
</tr>
<tr>
<td>Disabled</td>
<td></td>
</tr>
</tbody>
</table>

### Resource Priority Namespace

<table>
<thead>
<tr>
<th>Namespace Name</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### Timer Settings

<table>
<thead>
<tr>
<th>Timer Type</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Keep Alive Delta (seconds)</td>
<td>6</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Hook Time To First Digit (ms)</td>
<td>15000</td>
</tr>
</tbody>
</table>

### Speed Dial Configurations

<table>
<thead>
<tr>
<th>Dial Type</th>
<th>Number</th>
</tr>
</thead>
<tbody>
<tr>
<td>Abbreviated Dial</td>
<td>100-serviceun-divibel</td>
</tr>
<tr>
<td>Abbreviated Dial 600</td>
<td>100-serviceun-600dial</td>
</tr>
</tbody>
</table>

### Normalization Script

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

### SIP Options Ping

<table>
<thead>
<tr>
<th>Option</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>20</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)</td>
<td>1000</td>
</tr>
<tr>
<td>Ping Retry Count</td>
<td>6</td>
</tr>
</tbody>
</table>
Configure a SIP Trunk

Navigate to the Device → Trunk page in the CUCM portal.

- Click on “Add New”
- Select “SIP Trunk” as the trunk type.
- The Device Protocol will automatically be populated with “SIP”, just click Next
- Configure as below screen shot.
- Save the configuration.
### Trunk Configuration

**Status**
- Status: Ready

**Service Status**
- Service Status: Full Service
- Duration: Time In Full Service: 8 days 21 hours 52 minutes

### 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(Calls Only)</td>
</tr>
<tr>
<td>Description</td>
<td>SEL SIP TRUNK</td>
</tr>
<tr>
<td>Device Port</td>
<td>SEL SIP DP</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>(None)</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Location</td>
<td>(None)</td>
</tr>
<tr>
<td>AAR Group</td>
<td>(None)</td>
</tr>
<tr>
<td>Tunnedled Protocol</td>
<td>None</td>
</tr>
<tr>
<td>QSIG Variant</td>
<td>(None)</td>
</tr>
<tr>
<td>ASN.1 ROSE OIDs Encoding</td>
<td>(None)</td>
</tr>
<tr>
<td>Packet Capture Mode</td>
<td>(None)</td>
</tr>
<tr>
<td>Packet Capture Duration</td>
<td>0</td>
</tr>
<tr>
<td>Media Termination Point Requ</td>
<td></td>
</tr>
</tbody>
</table>

### Intercompany Media Engine (IME)

- 164 Transformation Profile: <None>

### MLPP and Confidential Access Level Information

- MLPP Domain: <None>
- Confidential Access Mode: <None>
- Confidential Access Level: <None>

### Call Routing Information

- Remote-Party-ID
- Assisted-Identity
- Asserted-Identity: PA
- SIP Privacy: Default
### Inbound Calls

<p>| | | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits*</td>
<td>All</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Connected Line ID Presentation*</td>
<td>Default</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Connected Name Presentation*</td>
<td>Default</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Prefix DN</td>
<td></td>
<td></td>
<td></td>
<td></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 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></th>
<th></th>
<th></th>
<th></th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number Type</td>
<td>Prefix</td>
<td>Strip Digits</td>
<td>Calling Search Space</td>
<td></td>
</tr>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>1</td>
<td>&lt; None &gt;</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></th>
<th></th>
<th></th>
<th></th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Number Type</td>
<td>Prefix</td>
<td>Strip Digits</td>
<td>Calling Search Space</td>
<td></td>
</tr>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>1</td>
<td>&lt; None &gt;</td>
<td></td>
</tr>
</tbody>
</table>

### Connected Party Settings

- **Connected Party Transformation CSS** | < None > |
- Use Device Pool Connected Party Transformation CSS

### Outbound Calls

- **Called Party Transformation CSS** | < None > |
- Use Device Pool Called Party Transformation CSS
- **Calling Party Transformation CSS** | < None > |
- Use Device Pool Calling Party Transformation CSS
- **Calling Party Selection** | Originator |
- **Calling Line ID Presentation** | Allowed |
- **Calling Name Presentation** | Default |
- **Calling and Connected Party Info Format** | Deliver DN only in connected party |
- **Redirecting Diversion Header Delivery - Outbound**
- **Redirecting Party Transformation CSS** | Diversion-CSS |
- Use Device Pool Redirecting Party Transformation CSS

### Caller Information

- **Caller ID DN**
- **Caller Name**

- **Maintain Original Caller ID DN and Caller Name in Identity Headers**
## SIP Information

<table>
<thead>
<tr>
<th>Destination</th>
<th>Destination Address</th>
<th>Destination Address IPv6</th>
<th>Destination Port</th>
<th>Status</th>
<th>Status Reason</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>192.168.1.1</td>
<td></td>
<td>5060</td>
<td>up</td>
<td></td>
</tr>
</tbody>
</table>

- **NTP Preferred Originating Codes**: 0728/0729m
- **BIF Presence Group**: Standard Presence group
- **SIP Trunk Security Profile**: Dell SIP Trunk Profile
- **Re-routing Calling Search Space**: x None x
- **Out-of-Dialog Refer Calling Search Space**: x None x
- **SUBSCRIBE Calling Search Space**: x None x
- **SIP Profile**: Dell SIP Profile
- **DTMF Signaling Method**: RFC 2833

### Normalization Script

<table>
<thead>
<tr>
<th>Normalization Script</th>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
</table>

- **Enable Trace**: Check box

### Recording Information

- **None**
- **This trunk connects to a recording-enabled gateway**
- **This trunk connects to other clusters with recording-enabled gateways**

### Geolocation Configuration

- **Geolocation**: x None x
- **Geolocation Filter**: x None x
- **Send Geolocation Information**: Check box

---

* indicates required item.

** Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
Trunk Configuration

- **Status**
  - Status: Ready

- **SIP Trunk Status**
  - Service Status: Full Service
  - Duration: Time In Full Service: 0 days 23 hours 30 minutes

- **Device Information**
  - Product:
  - Device Protocol:
  - Trunk Service Type:
  - Device Name:
  - Description:
  - Device Port:
  - Common Device Configuration:
  - Call Classification:
  - Media Resource Group List:
  - Location:
  - AAR Group:
  - Tunnelled Protocol:
  - QSIG Variant:
  - ASN.1 ROSE CID Encoding:
  - Packet Capture Mode:
  - Packet Capture Duration:
  - Media Termination Point Required:

- **Intercom Media Engine (IME)**
  - E.164 Transformation Profile:

- **MLPP and Confidential Access Level Information**
  - MLPP Domain:
  - Confidential Access Mode:
  - Confidential Access Level:

- **Call Routing Information**
  - Remote-Party Id:
  - Asserted-Identity:
  - Assessed-Type:
  - SIP Privacy:
Configure a SIP Trunk Security Profile

- Navigate to the System → Security Profile → SIP Trunk Security Profile page in the portal (see screen capture below):
- Copy the existing Non-secure SIP Trunk Profile.
- Name the New SIP Trunk Profile.
- Save the configuration.
Configure a Region

Navigate to the System → Region Information → Region page in the CUCM portal.

- Click on the “Add New” button
- Name the new region.
- Click on the newly created region and choose Audio Codec Preference List and Maximum Audio Bit Rate to 64 kbps (G.722, G.711)
- Select other settings to be consistent with the customer’s environment.
- Save the configuration.
### Region Configuration

#### Region Information
- **Name**: **BELL SIP REGION**

#### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
<th>Maximum Session Bit Rate for Immersive Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>BELL SIP REGION</strong></td>
<td>Use System Default (Factory Default low loss)</td>
<td>64 kbps (0.722, 0.711)</td>
<td>304 kbps</td>
<td>2147483647 kbps</td>
</tr>
<tr>
<td><strong>BELL SIP REGION</strong></td>
<td>Use System Default (Factory Default low loss)</td>
<td>64 kbps (0.722, 0.711)</td>
<td>304 kbps</td>
<td>2147483647 kbps</td>
</tr>
</tbody>
</table>

**NOTE:** Regions not displayed

*Use System Default* *Use System Default* *Use System Default* *Use System Default*

#### Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec Preference List</th>
<th>Maximum Audio Bit Rate</th>
<th>Maximum Session Bit Rate for Video Calls</th>
<th>Maximum Session Bit Rate for Immersive Video Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>BELL SIP REGION</strong></td>
<td>Use System Default (Factory Default low loss)</td>
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<td>304 kbps</td>
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<td><strong>BELL SIP REGION</strong></td>
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<td>304 kbps</td>
<td>2147483647 kbps</td>
</tr>
</tbody>
</table>

**NOTE:** Regions not displayed

*Use System Default* *Use System Default* *Use System Default* *Use System Default*
Configure a Device pool:

Navigate to the System → Device Pool page in the CUCM portal.

- Click on “Add New”
- Name the new Device Pool.
- Select desired Region created in the previous step)
- Fill in required fields with appropriate configurations for your customer environment.
- Save the configuration.
Configure Route groups, Route Lists, and Route Patterns

Due to the complexity and the wide variance in the individual needs of each enterprise, configuring enterprise route patterns requires detailed planning and consideration. For this reason, route patterns are not covered in any detail in this reference document. Please refer to the appropriate Cisco documentation for guidance in configuring Route Patterns suitable to your particular situation.

For the sake of a simple example, a simple Route Pattern that routes calls to a CUBE is defined here:

- Navigate to Call routing → Route/Hunt → Route Pattern
- Click on “Add New”
- Insert route-pattern expression
- Choose a route partition
- Give it a description describing the intention of this route pattern.
- Choose the Gateway/route list as per provisioned.
- Define any other trunk specific configurations.
- Save the configuration
### Find and List Route Patterns

<table>
<thead>
<tr>
<th>Pattern</th>
<th>Description</th>
<th>Partition</th>
<th>Route Filter</th>
<th>Associated Device</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>*67.5#</td>
<td>Block Calling Name and Number</td>
<td>PHONE-PT</td>
<td>SIP_AL</td>
<td></td>
<td></td>
</tr>
<tr>
<td>9.6</td>
<td></td>
<td>FAX-PT</td>
<td>FAX_AL</td>
<td></td>
<td></td>
</tr>
<tr>
<td>9.6</td>
<td></td>
<td>PHONE-PT</td>
<td>SIP_AL</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Route Pattern Configuration

**Status:** Ready

**Pattern Definition:**
- Route Pattern: 9.6
- Route Partition: PHONE-PT
- Description:
- Numbering Plan: NAP
- Route Filter: None
- MLPP Preference: Default
- Apply Call Blocking Percentage: No
- Resource Priority Namespace Network Domain: None
- Route Class: Default
- Gateway/Route List: SIP_AL (Edit)
- Route Option: Block this pattern: No
- Call Classification: Off
- External Control Profile: None
- Allow Device Override: Yes
- Provide Outside Dialed Tone: No
- Allow Overfow Sending: No
- Urgent Priority: No
- Require Forward Authentication Code: No
- Authorization Level: 0
- Require Client Header Code: No

### Calling Party Transformations
Route List Configuration
### Route List Configuration

#### Status
- Status: Ready

#### Route List Information
- **Registration:** Registered with Cisco Unified Communications Manager 10.220.22.161
- **IPv4 Address:** 10.220.22.161
- **Device is trusted:**
- **Name:** SIP RL
- **Description:**
- **Cisco Unified Communications Manager Group:** Default
- **Enable this Route List (change effective on Save; no reset required):**
- **Run On All Active Unified CM Nodes:**

#### Route List Member Information
- **Selected Groups:**
  - SIP RL
- **Removed Groups:**

#### Route List Details
- **SIP RL**

---

### Route List Configuration

#### Status
- Status: Ready

#### Route List Information
- **Registration:** Registered with Cisco Unified Communications Manager 10.220.22.161
- **IPv4 Address:** 10.220.22.161
- **Device is trusted:**
- **Name:**
- **Description:**
- **Cisco Unified Communications Manager Group:** Default
- **Enable this Route List (change effective on Save; no reset required):**
- **Run On All Active Unified CM Nodes:**

#### Route List Member Information
- **Selected Groups:**
  - FAX RL
- **Removed Groups:**

#### Route List Details
- **FAX RL**
## Route Group Configuration

### Route Group Information
- **Route Group Name**: SIP_RG
- **Distribution Algorithm**: Circular

### Route Group Member Information
#### Find Devices to Add to Route Group
- **Device Name contains**:
- **Available Devices**: SIP1, SIP2, SIP3, SIP4
- **Port(s)**: All

#### Current Route Group Members
- **Selected Devices (ordered by priority)**: SIP1, SIP2, SIP3, SIP4
- **Removed Devices**: None

### Route Group Members
- SIP1
- SIP2
- SIP3
- SIP4
## Calling Search Space Configuration

### Find and List Calling Search Spaces

- **Status:** 3 records found

<table>
<thead>
<tr>
<th>CSS Name</th>
<th>Description</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Diversion-CSS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Fax-CSS</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Phone-CSS</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

### Calling Search Space Configuration

- **Status:** Status: Ready

#### Calling Search Space Information

**Name:** Diversion-CSS

**Description:**

#### Route Partitions for this Calling Search Space

- **Available Partitions:**
  - Directory URI
  - Diversion-PT
  - Global Learned E164 Numbers
  - Global Learned E164 Patterns
  - Global Learned Enterprise Numbers

- **Selected Partitions:**
  - Diversion-PT

### Calling Search Space Configuration

- **Status:** Status: Ready

#### Calling Search Space Information

**Name:** Fax-CSS

**Description:**

#### Route Partitions for this Calling Search Space

- **Available Partitions:**
  - Directory URI
  - Diversion-PT
  - Global Learned E164 Numbers
  - Global Learned E164 Patterns
  - Global Learned Enterprise Numbers

- **Selected Partitions:**
  - FAX-PT
### Translation Pattern Configuration

<table>
<thead>
<tr>
<th>Pattern Definition</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Translation Pattern</td>
<td>41635321XX</td>
</tr>
<tr>
<td>Pattern Description</td>
<td></td>
</tr>
<tr>
<td>Numbering Plan</td>
<td></td>
</tr>
<tr>
<td>Route Filter</td>
<td></td>
</tr>
<tr>
<td>NLP Precedence</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Name</td>
<td></td>
</tr>
<tr>
<td>Route Class</td>
<td></td>
</tr>
<tr>
<td>Calling Search Space</td>
<td></td>
</tr>
<tr>
<td>Use Originator’s Calling Search Space</td>
<td></td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td></td>
</tr>
<tr>
<td>Route Option</td>
<td></td>
</tr>
<tr>
<td>Provide Outside Dial Tone</td>
<td></td>
</tr>
<tr>
<td>Urgent Priority</td>
<td></td>
</tr>
<tr>
<td>Do Not Wait For Interdigit Timeout On Subsequent Hops</td>
<td></td>
</tr>
<tr>
<td>Route Next Hop By Calling Party Number</td>
<td></td>
</tr>
</tbody>
</table>

### Calling Search Space Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Phone-COS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td></td>
</tr>
</tbody>
</table>

### Route Partitions for this Calling Search Space

<table>
<thead>
<tr>
<th>Available Partitions</th>
<th>Selected Partitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>Directory URI</td>
<td>PHONE-PT</td>
</tr>
<tr>
<td>Division-PT</td>
<td></td>
</tr>
<tr>
<td>Prefix-PT</td>
<td></td>
</tr>
<tr>
<td>Global Learned E164 Numbers</td>
<td></td>
</tr>
<tr>
<td>Global Learned E164 Patterns</td>
<td></td>
</tr>
</tbody>
</table>
Calling Party Transformation Pattern Configuration

Status
- Status: Ready

Pattern:
- Pattern: XXXX
- Partition: Division-PIT

Description:

Numbering Plan:
- < None >

Route Filter:
- < None >

Calling Party Transformations
- Use Calling Party's External Phone Number Mask
- Calling Party Transformation Mask:
- Prefix Digits:
- Calling Line ID Presentation:
- Calling Name Presentation:
- Calling Party Number Type:
- Calling Party Numbering Plan:

* - indicates required item.
Transcoder Configuration

### Transcoder Information
- **Transcoder:** Transcoder (CUBE Transcode)
- **Registration:** Registered with Cisco Unified Communications Manager 10.220.22.161
- **IPv4 Address:** 10.220.53.5

<table>
<thead>
<tr>
<th><strong>Parameter</strong></th>
<th><strong>Value</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Transcoder Type</strong></td>
<td>Cisco IOS Enhanced Media Termination Point</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>CUBE Transcode</td>
</tr>
<tr>
<td><strong>Device Name</strong></td>
<td>Transcoder</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>BELL SIP DP</td>
</tr>
<tr>
<td><strong>Common Device Configuration</strong></td>
<td>None</td>
</tr>
<tr>
<td><strong>Special Load Information</strong></td>
<td>Leave blank to use default</td>
</tr>
</tbody>
</table>

* Asterisk (*) indicates required item.

---

Media Termination Point

### Media Termination Point Configuration

**Status:** Ready

<table>
<thead>
<tr>
<th><strong>Parameter</strong></th>
<th><strong>Value</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Registration</strong></td>
<td>Registered with Cisco Unified Communications Manager 10.220.22.161</td>
</tr>
<tr>
<td><strong>IPv4 Address</strong></td>
<td>10.220.22.161</td>
</tr>
<tr>
<td><strong>Media Termination Point Type</strong></td>
<td>Cisco Media Termination Point Software</td>
</tr>
<tr>
<td><strong>Host Server</strong></td>
<td>10.220.22.161</td>
</tr>
<tr>
<td><strong>Media Termination Point Name</strong></td>
<td>MTP_12</td>
</tr>
<tr>
<td><strong>Description</strong></td>
<td>MTP_cu0m138</td>
</tr>
<tr>
<td><strong>Device Pool</strong></td>
<td>Default</td>
</tr>
</tbody>
</table>

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Conference Bridge Configuration

Software Conference Bridge Info
- Conference Bridge Name: CFB_2
- Description: CF_B_CONF105
- Device Pool: Default

Media Resource Group Configuration

Find and List Media Resource Groups
- Add New
- Select All
- Clear All
- Delete Selected

Status
- 2 records found

Media Resource Group (1 - 2 of 2)

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Multicast</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>BELL SIP MRG</td>
<td>BELL SIP MRG</td>
<td>false</td>
<td></td>
</tr>
<tr>
<td>Dummy MRG</td>
<td></td>
<td>false</td>
<td></td>
</tr>
</tbody>
</table>
Note: The Media Resource Group (MRG) contains media resources, such as Conference Bridge, Transcoder, MoH server, and Annunciator. It will be assigned to a Media Resource Group List (MRGL), used to allocate media resources to groups of devices through Device Pools, or individually by configuring a valid MRGL at the device configuration page.
<table>
<thead>
<tr>
<th>Secure Directory URL</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Secure Idle URL</td>
<td></td>
</tr>
<tr>
<td>Secure Information URL</td>
<td></td>
</tr>
<tr>
<td>Secure Messages URL</td>
<td></td>
</tr>
<tr>
<td>Secure Services URL</td>
<td></td>
</tr>
</tbody>
</table>

### Extension Information

- **Enable Extension Mobility**: 
- **Log Out Profile**: [Select or Use Current Device Setting]
- **Log in Time**: [None]
- **Log Out Time**: [None]

### MIFP and Confidential Access Level Information

- **MIFP Domain**: [None]
- **MIFP Indicator**: [None]
- **MIFP Password**: [None]
- **Confidential Access Mode**: [None]
- **Confidential Access Level**: [None]

### 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

- **Disable Speakerphone**: [Disabled]
- **Enable Speakerphone and Headset**: [Enabled]
- **Forwarding Delay**: [Disabled]
- **PC Port**: [Enabled]
- **Settings Access**: [Enabled]
- **Gratuity**: [Disabled]
- **PC Voice VLAN Access**: [Enabled]
- **Video Capabilities**: [Disabled]
- **Auto Line Select**: [Disabled]
- **Web Access**: [Disabled]
- **Enable Power Save Plus**: [Disabled]
- **Secure Power Save Plus**: [Sunday]
- **Phone On Time**: [00:00]
- **Phone Off Time**: [24:00]
- **Phone Off Idle Timeout**: [60]
- **Enable Audible Alert**: 
- **EnergyWise Domain**: 
- **EnergyWise Endpoint Security Secret**: 
- **Allow EnergyWise Override**: 

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<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allow EnergyWise Overrides</td>
<td>Disabled</td>
</tr>
<tr>
<td>Open 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>Recording Tone Duration</td>
<td></td>
</tr>
<tr>
<td>RTCP</td>
<td></td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td>Disabled</td>
</tr>
<tr>
<td>Auto Call Select</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
</tr>
<tr>
<td>Advertisement G.722 Codec</td>
<td></td>
</tr>
<tr>
<td>Widescreen Headset UE Control</td>
<td></td>
</tr>
<tr>
<td>Widescreen Headset</td>
<td></td>
</tr>
<tr>
<td>Poor Firmware Sharing</td>
<td></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</td>
<td></td>
</tr>
<tr>
<td>Discover (LDP-MBD): Switch Port</td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LDP): PC Port</td>
<td></td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority</td>
<td></td>
</tr>
<tr>
<td>Wireless Headset Mockswitch Control</td>
<td>Disabled</td>
</tr>
<tr>
<td>Display Refresh Rate</td>
<td>Normal</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>IPv6 Log Server</td>
<td>User Controlled</td>
</tr>
<tr>
<td>IPv6 Load Server</td>
<td></td>
</tr>
<tr>
<td>EAPOL Authentication</td>
<td></td>
</tr>
<tr>
<td>Dataset Unified CN 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>HTTP Server</td>
<td>http and https Enabled</td>
</tr>
<tr>
<td>Handset/Headset Monitor</td>
<td></td>
</tr>
<tr>
<td>Headset Recording</td>
<td>Disabled</td>
</tr>
<tr>
<td>Endpoint Dialing</td>
<td>Enabled</td>
</tr>
<tr>
<td>Switch Port Remote Configuration</td>
<td>Disabled</td>
</tr>
<tr>
<td>PC Port Remote Configuration</td>
<td>Disabled</td>
</tr>
<tr>
<td>Automatic Port Synchronization</td>
<td>Disabled</td>
</tr>
<tr>
<td>SM Access</td>
<td>Disabled</td>
</tr>
<tr>
<td>LOGIN Access</td>
<td>Enabled</td>
</tr>
<tr>
<td>FIPS Mode</td>
<td>Disabled</td>
</tr>
<tr>
<td>802.11s ERTCP</td>
<td>Disabled</td>
</tr>
</tbody>
</table>
### Secondary Calling Search Space for Forward All

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

**No Answer Ring Duration (seconds)**: 8

**Call Pickup Group**: < None >

### Park Monitoring

<table>
<thead>
<tr>
<th>Monitoring Type</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring Forward No Retrieve</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Destination External</td>
<td></td>
<td>A blank value means to call the parker’s line.</td>
<td></td>
</tr>
<tr>
<td>Park Monitoring Forward No Retrieve</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Destination Internal</td>
<td></td>
<td>A blank value means to call the parker’s line.</td>
<td></td>
</tr>
<tr>
<td>Park Monitoring Reversion Timer</td>
<td></td>
<td>A blank value will use value set in Park Monitoring Reversion Timer service parameter</td>
<td></td>
</tr>
</tbody>
</table>

### MLPP Alternate Party And Confidential Access Level Settings

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target (Destination)</td>
<td></td>
</tr>
<tr>
<td>MLPP Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLPP No Answer Ring Duration (Seconds)</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Confidential Access Mode</td>
<td></td>
</tr>
<tr>
<td>Confidential Access Level</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Control Agent Profile</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Line Settings for All Devices

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hold Reversion Ring Duration (seconds)</td>
<td></td>
</tr>
<tr>
<td>Hold Reversion Notification Interval (seconds)</td>
<td></td>
</tr>
<tr>
<td>Party Entrance Tone</td>
<td>Default</td>
</tr>
</tbody>
</table>

### Line 1 on Device 10P4ADD9055C04

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Display (Caller ID)</td>
<td>Gray Phone 1</td>
</tr>
<tr>
<td>ASCII Display (Caller ID)</td>
<td>Gray Phone 1</td>
</tr>
<tr>
<td>Line Text Label</td>
<td>1134</td>
</tr>
<tr>
<td>External Phone Number Mask</td>
<td>816333XXX</td>
</tr>
<tr>
<td>Visual Message Waiting Indicator Policy</td>
<td>Use System Policy</td>
</tr>
<tr>
<td>Audible Message Waiting Indicator Policy</td>
<td>Default</td>
</tr>
<tr>
<td>Ring Setting (Phone Idle)</td>
<td>Ring</td>
</tr>
<tr>
<td>Ring Setting (Phone Active)</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup Group Audio Alert Setting</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>
VG224 Analog Gateway (FAX application)
### VG224 Analog Gateway Directory Number Configuration

#### Directory Number Configuration

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

#### Directory Number Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Partition</td>
<td>2161</td>
</tr>
<tr>
<td>Description</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Alerting Name</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>ASCII Alerting Name</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>External Call Control Profile</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Associated Devices</td>
<td>AALN/S010/EBELL_VG224</td>
</tr>
</tbody>
</table>

#### Directory Number Settings

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

**Note**: (Choose <None> to use system default)
**Enterprise Alternate Number**

- Add Enterprise Alternate Number

**+E.164 Alternate Number**

- Add +E.164 Alternate Number

**Directory URLs**

<table>
<thead>
<tr>
<th>Primary</th>
<th>URL</th>
<th>Partition</th>
<th>Advertise Globally via ILS</th>
<th>Remove</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- Add URL

**PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialog**

- Advertised Fellower Number: <None>

**AAR Settings**

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

- Select AAR
- Retain this destination in the call forwarding history

**Call Forward and Call Pickup Settings**

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

- Forward Search Space Activation Policy: Use System Default
- Forward All: <None>

**Secondary Calling Search Space for Forward All**

- Forward Busy Internal: <None>
- Forward Busy External: <None>
- Forward No Answer Internal: <None>
- Forward No Answer External: <None>
- Forward No Coverage Internal: <None>
- Forward No Coverage External: <None>
- Forward on CTI Failure: <None>
- Forward Unregistered Internal: <None>
- Forward Unregistered External: <None>

- No Answer Ring Duration (seconds): <None>

**Park Monitoring**

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

- Park Monitoring Forward No Retrieve Destination Internal: <None>
- Park Monitoring Reversion Timer: A blank value will use value set in Park Monitoring Reversion Timer service parameter

**MLPP Alternate Party and Confidential Access Level Settings**

<table>
<thead>
<tr>
<th>Target (Destination)</th>
<th></th>
</tr>
</thead>
</table>
### Line Settings for All Devices
- **Hold Reversion Ring Duration (seconds)**: Setting the Hold Reversion Ring Duration to zero will disable the feature.
- **Hold Reversion Notification Interval (seconds)**: Setting the Hold Reversion Notification Interval to zero will disable the feature.
- **Party Entrance Tone**: Default

### Line 1 on Device AALN/S2/08BELL_VG224
- **Display (Caller ID)**: VG224 PORT 0 - FAX LINE 1
- **ASCII Display (Caller ID)**: VG224 PORT 0 - FAX LINE 1
- **External Phone Number Mask**: 41535321XX

### Multiple Call/Call Waiting Settings on Device AALN/S2/08BELL_VG224
- Note: The range for selecting the Max Number of Calls is 1-2.
- **Maximum Number of Calls**: 2
- **Busy Trigger**: (Less than or equal to Max. Calls)

### Forwarded Call Information Display on Device AALN/S2/08BELL_VG224
- **Call Name**: Checked
- **Caller Number**: Checked
- **Redirected Number**: Checked
- **Dialled Number**: Checked

### Users Associated with Line
- **Associate End Users**
IOS Configuration of VG224 Analog Gateway

Bell_VG224#show run
Building configuration...

Current configuration : 3403 bytes

! version 12.4
no service pad
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Bell_VG224
!
boot-start-marker
boot-end-marker
!
no aaa new-model
!
resource policy
!
ip subnet-zero
no ip domain lookup
no ip dhcp use vrf connected
!
voice-card 0
!
voice service voip
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
supplementary-service h450.12
!
interface FastEthernet0/0
 ip address 10.220.144.14 255.255.255.240
duplex auto
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
duplex auto
 speed auto
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.220.144.1
!
ip http server
!
control-plane
!
voice-port 2/0
timeouts initial 60
timeouts interdigit 60
timeouts ringing infinity
!
voice-port 2/1
timeouts initial 60
timeouts interdigit 60
timeouts ringing infinity

voice-port 2/2
output attenuation -6

voice-port 2/3

voice-port 2/4

voice-port 2/5

voice-port 2/6

voice-port 2/7

voice-port 2/8

voice-port 2/9

voice-port 2/10

voice-port 2/11

voice-port 2/12

voice-port 2/13

voice-port 2/14

voice-port 2/15

voice-port 2/16

voice-port 2/17

voice-port 2/18

voice-port 2/19

voice-port 2/20

voice-port 2/21

voice-port 2/22

voice-port 2/23

ccm-manager mgcp
ccm-manager music-on-hold
ccm-manager config
ccm-manager sccp local FastEthernet0/0

mgcp
mgcp call-agent 10.220.22.161 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp rtp unreachable timeout 1000 action notify
mgcp modem passthrough voip mode nse
mgcp ip qos dscp cs3 signaling
mgcp package-capability rtp-package
no mgcp package-capability res-package
mgcp package-capability sst-package
no mgcp package-capability fxr-package
no mgcp timer receive-rtcp
mgcp sdp simple
mgcp fax t38 inhibit
mgcp rtp payload-type g726r16 static
mgcp bind control source-interface FastEthernet0/0
mgcp bind media source-interface FastEthernet0/0
!
mgcp profile default
!
dial-peer voice 99920 pots
  service mgcpapp
  port 2/0
!
dial-peer voice 99921 pots
  service mgcpapp
  port 2/1
!
dial-peer voice 99922 pots
  service mgcpapp
  port 2/22
!
dial-peer voice 99923 pots
  service mgcpapp
  port 2/23
!
dial-peer voice 99924 pots
  service mgcpapp
  port 2/4
!
dial-peer voice 99925 pots
  service mgcpapp
  port 2/5
!
dial-peer voice 99926 pots
  service mgcpapp
  port 2/6
!
dial-peer voice 99927 pots
  service mgcpapp
  port 2/7
!
dial-peer voice 99928 pots
  service mgcpapp
  port 2/8
!
dial-peer voice 99929 pots
  service mgcpapp
  port 2/9
!
dial-peer voice 999210 pots
  service mgcpapp
  port 2/10
! dial-peer voice 999211 pots
  service mgcpapp
  port 2/11
!
dial-peer voice 999212 pots
  service mgcpapp
  port 2/12
!
dial-peer voice 999213 pots
  service mgcpapp
  port 2/13
!
dial-peer voice 999214 pots
  service mgcpapp
  port 2/14
!
dial-peer voice 999215 pots
  service mgcpapp
  port 2/15
!
dial-peer voice 999216 pots
  service mgcpapp
  port 2/16
!
dial-peer voice 999217 pots
  service mgcpapp
  port 2/17
!
dial-peer voice 999218 pots
  service mgcpapp
  port 2/18
!
dial-peer voice 999219 pots
  service mgcpapp
  port 2/19
!
dial-peer voice 999220 pots
  service mgcpapp
  port 2/20
!
dial-peer voice 999221 pots
  service mgcpapp
  port 2/21
!
 line con 0
 line aux 0
 line vty 0 4
 login
!
end
<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control 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>DNS</td>
<td>Domain Name Resolution</td>
</tr>
<tr>
<td>G.711</td>
<td>Voice Codec (Uncompressed)</td>
</tr>
<tr>
<td>G.729</td>
<td>Voice Codec (Compressed)</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>PAI</td>
<td>P-Asserted Identity</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>URI</td>
<td>Uniform Resource Indicator</td>
</tr>
<tr>
<td>RTP</td>
<td>Real Time Protocol</td>
</tr>
</tbody>
</table>
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