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

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(lr)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:

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)
```
<<CUBE Configuration excerpt BEGIN>>

voice-card 0
ds pfarm
dsp services ds pfarm
!
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-keepalive down-interval 20
dtmf-relay rtp-nre
!
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-keepalive 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 trgp 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.
### SIP Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>PCIE SIP Profile</td>
<td>PCIE SIP Profile</td>
</tr>
</tbody>
</table>

### Early Offer for G-Clear Cells
- **Enabled**

### User-Agent and Server header information
- **Send Unified CM Version Information as User-Agent**
- **Version in User-Agent and Server Header**
- **Dial String Interpretation**
- **Confidential Access Level Headers**
- **Redirect by Application**
- **Disabled Early Media on 180**
- **Outgoing T.38 INVITE include audio mime**
- **Use Fully Qualified Domain Name in SIP Requests**
- **Assured Services SIP conformance**

### SDP Information

#### SDP Session-level Bandwidth Modifier for Early Offer and Re-invites
- **TIAS and AS**

#### SDP Transparency Profile
- **Pass all unknown SDP attributes**

### Parameters used in Phone

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer: Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer: Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer: Register Expires (seconds)</td>
<td>3000</td>
</tr>
<tr>
<td>Timer: T1 (msec) <strong>t</strong></td>
<td>800</td>
</tr>
<tr>
<td>Timer: T2 (msec) <strong>t</strong></td>
<td>3000</td>
</tr>
<tr>
<td>Retry INVITE <strong>t</strong></td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE <strong>t</strong></td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port <strong>t</strong></td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port <strong>t</strong></td>
<td>5766</td>
</tr>
<tr>
<td>Call Pickup URL</td>
<td><code>~cisco-service-en-pickup</code></td>
</tr>
<tr>
<td>Call Pickup Group Other URL <strong>t</strong></td>
<td><code>~cisco-service-en-opc</code></td>
</tr>
<tr>
<td>Call Pickup Group URL <strong>t</strong></td>
<td><code>~cisco-service-en-gp</code></td>
</tr>
<tr>
<td>Meet Me Service URL <strong>t</strong></td>
<td><code>~cisco-service-en-meetme</code></td>
</tr>
<tr>
<td>User Info <strong>t</strong></td>
<td>None</td>
</tr>
<tr>
<td>DTMF Dig Level <strong>t</strong></td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back <strong>t</strong></td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block <strong>t</strong></td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking <strong>t</strong></td>
<td>Off</td>
</tr>
</tbody>
</table>

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### SDP Information
- Send send-receive SDP in mid-call INVITE
- Allow Presentation Sharing using BFCP
- Allow IX Application Media
- Allow multiple codecs in answer SDP

### SIP Profile Configuration
- **Status:** Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

#### SIP Profile Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>E911 FAX SIP Profile</strong></td>
<td>[List of values]</td>
</tr>
</tbody>
</table>

- Early Offer for O/C Clear Calls: [Options]
- User-Agent and Server header information: [Options]
- Version in User Agent and Server Header: [Options]
- Dial String Interpretation: [Options]
- Confidential Access Level Headers: [Options]
- Redirect by Application: [Options]
- Disable Early Media on 100: [Options]
- Outgoing T.38 INVITE includes audio rdma: [Options]
- Use Fully Qualified Domain Name in SIP Requests: [Options]
- Assured Services SIP conformance: [Options]

#### SDP Information

<p>| SDP Session-level Bandwidth Modifier for Early Offer and Re-Invites | [Options] |
| SDP Transparency Profile | [Options] |
| Accept Audio Codec Preferences in Received Offer | [Options] |
| Require SDP Inactive Exchange for Mid-Call Media Change | [Options] |</p>
<table>
<thead>
<tr>
<th>Parameters used in Phone</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Invite Expires (seconds)</td>
<td>180</td>
</tr>
<tr>
<td>Timer Register Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Timer Register Expires (seconds)</td>
<td>5600</td>
</tr>
<tr>
<td>Timer T1 (msecs)</td>
<td>800</td>
</tr>
<tr>
<td>Timer T2 (msecs)</td>
<td>4000</td>
</tr>
<tr>
<td>Retry INVITE</td>
<td>6</td>
</tr>
<tr>
<td>Retry Non-INVITE</td>
<td>10</td>
</tr>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32765</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7950</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Parameters used in Phone</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>16000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrdial</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Normalization Script</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Normalization Script</th>
<th>Enable Trace</th>
</tr>
</thead>
<tbody>
<tr>
<td>1 Parameter Name</td>
<td>Parameter Value</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Incoming Requests FROM URI Settings</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller ID DN</td>
<td></td>
</tr>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Trunk Specific Configuration</th>
<th>Value</th>
</tr>
</thead>
</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.
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 Table

<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>DELL FAX SIP REGION</td>
<td>Use System Default (Factory Default low loss)</td>
<td>64 kbps (0.722, 0.711)</td>
<td>394 kbps</td>
<td>2147483647 kbps</td>
</tr>
<tr>
<td>BELL SIP REGION</td>
<td>Use System Default (Factory Default low loss)</td>
<td>64 kbps (0.722, 0.711)</td>
<td>394 kbps</td>
<td>2147483647 kbps</td>
</tr>
</tbody>
</table>

**NOTE:** Regions not displayed: 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>DELL FAX SIP REGION</td>
<td>Keep Current Setting</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>BELL SIP REGION</td>
<td>Keep Current Setting</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**NOTE:** Regions not displayed: Use System Default

---

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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.
### Device Mobility Related Information

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Mobility 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>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Called Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Party Transformation CSS</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Geolocation Configuration

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Geolocation</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Geolocation Filter</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### 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>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</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>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Phone Settings

#### Caller ID For Calls From This Phone

| Calling Party Transformation CSS | < None > |

#### Connected Party Settings

| Connected Party Transformation CSS | < None > |

#### Redirecting Party Settings

| Redirecting Party Transformation CSS | < None > |
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>9.0</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:** *67.5#
- **Route Partition:** PHONE-PT
- **Description:**
- **Numbering Plan:** NAP
- **Route Filter:** None
- **MLPP Precedence:** Default
- **Apply Call Blocking Percentage:**
- **Resource Priority Namespace Network Domain:**
- **Route Class:** Default
- **Gateway/Route List:** SIP AL
- **Route Option:** Block this pattern No Error

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

### Status

2 records found

### Route List (1 - 2 of 2)

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Enabled</th>
<th>Status</th>
<th>IPV4 Address</th>
</tr>
</thead>
<tbody>
<tr>
<td>FAX RL</td>
<td>true</td>
<td>Registered with 10.220.22.161</td>
<td>10.220.22.161</td>
<td></td>
</tr>
<tr>
<td>SIP RL</td>
<td>true</td>
<td>Registered with 10.220.22.161</td>
<td>10.220.22.161</td>
<td></td>
</tr>
</tbody>
</table>
### 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**: 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**: FAX_RL
- **Description**: 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 Configuration

Route Group Information
Route Group Name: SIP_RG
Distribution Algorithm: Circular

Find Devices to Add to Route Group
Device Name contains:
Available Devices**:
BELLEFAYSIPTRUNK
BELLEFAYSIPTRUNK

Port(s):
ALL

Add to Route Group

Current Route Group Members
Selected Devices (ordered by priority):
BELLEFAYSIPTRUNK (All Ports)

Removed Devices***

Route Group Members
BELLEFAYSIPTRUNK

Route Group Information
Route Group Name: FAX_RG
Distribution Algorithm: Circular

Find Devices to Add to Route Group
Device Name contains:
Available Devices**:
BELLEFAYSIPTRUNK
BELLEFAYSIPTRUNK

Port(s):
None Available

Add to Route Group

Current Route Group Members
Selected Devices (ordered by priority):
BELLEFAYSIPTRUNK (All Ports)

Removed Devices***

Route Group Members
BELLEFAYSIPTRUNK
### Calling Search Space Configuration

#### Find and List Calling Search Spaces

- **Status:**
  - 3 records found

#### Calling Search Space (1 - 3 of 3)

<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>

- **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

- **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

- **Status:**
  - Status: Ready
Translation Pattern Configuration

- **Status**: Ready

- **Pattern Definition**
  - **Translation Pattern**: 4163321XX
  - **Partition**: None
  - **Description**: 
  - **Numbering Plan**: 
  - **Route Filter**: 
  - **NLP Priority**: Default
  - **Resource Priority Namespace Network Domain**: None
  - **Route Class**: Default
  - **Calling Search Space**
    - Use Originator’s Calling Search Space: None
    - External Call Control Profile: None
    - Route Option: Route this pattern
  - **Provide Outside Dial Tone**: 
  - **Urgent Priority**: 
  - **Do Not Wait For Interdigit Timeout On Subsequent Hops**: 
  - **Route Next Hop By Calling Party Number**: 
  - **Calling Party Transformations**
    - Use Calling Party’s External Phone Number Mask:

---

**Translation Pattern Configuration**

- **Status**: Ready
Calling Party Transformation Pattern Configuration

<table>
<thead>
<tr>
<th>Calling Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use Calling Party's External Phone Number Mask</td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
</tr>
<tr>
<td>Calling Line ID Presentation*</td>
</tr>
<tr>
<td>Calling Name Presentation*</td>
</tr>
<tr>
<td>Calling Party Number Type*</td>
</tr>
<tr>
<td>Calling Party Numbering Plan*</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation*</td>
</tr>
<tr>
<td>Connected Name Presentation*</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
</tr>
<tr>
<td>Called Party Number Type*</td>
</tr>
<tr>
<td>Called Party Numbering Plan*</td>
</tr>
</tbody>
</table>

* Indicates required item.
### Conference Bridge Configuration

<table>
<thead>
<tr>
<th>Conference Bridge Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference Bridge : CFB_2 (CFB_2.conf)</td>
</tr>
<tr>
<td>Registration : Registered with Cisco Unified Communications Manager 10.222.28.161</td>
</tr>
<tr>
<td>IPv4 Address : 10.222.28.161</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Software Conference Bridge Info</th>
</tr>
</thead>
<tbody>
<tr>
<td>Conference Bridge Type : Cisco Conference Bridge Software</td>
</tr>
<tr>
<td>Host Server : 10.222.28.161</td>
</tr>
<tr>
<td>Device is not trusted</td>
</tr>
<tr>
<td>Conference Bridge Name : CFB_2</td>
</tr>
<tr>
<td>Description : CFB_2.conf</td>
</tr>
<tr>
<td>Device Pool : Default</td>
</tr>
<tr>
<td>Common Device Configuration : X Name</td>
</tr>
<tr>
<td>Location : Hub_Name</td>
</tr>
<tr>
<td>Use Trusted Relay Point : Default</td>
</tr>
</tbody>
</table>

### Media Resource Group Configuration

#### Find and List Media Resource Groups

<table>
<thead>
<tr>
<th>Add New</th>
<th>Select All</th>
<th>Clear All</th>
<th>Delete Selected</th>
</tr>
</thead>
</table>

#### Status

2 records found

<table>
<thead>
<tr>
<th>Media Resource Group (1-2 of 2)</th>
<th>Rows per Page</th>
<th>50</th>
</tr>
</thead>
</table>

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Multi-cast</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>BELL SIP MRG</td>
<td>BELL SIP MRG</td>
<td>False</td>
<td>b</td>
</tr>
<tr>
<td>Dummy MRG</td>
<td>False</td>
<td>b</td>
<td></td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected
Media Resource Group List Configuration

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.
Phone Configuration

Phone Configuration

Phone Type:
- Product Type: Cisco 7962
- Device Protocol: SCCP

Real-time Device Status:
- Registration: Registered with Cisco Unified Communications Manager 16.20.22.161
- IPv4 Address: 10.222.0.101
- Active Load ID: SOCN4:0-0-1104-150
- Download Status: None

Device Information:
- Device is Active
- Device is Trusted
- MAC Address: 444D00C00004
- Description: 
- Device Pool: DELL SIP DP
- Common Device Configuration:
- Phone Button Template: Universal Device Template Button Layout
- Softkey Template: Standard User
- Common Phone Profile: Standard Common Phone Profile
- Calling Search Space: Phone-CBG
- AAR Ceiling Search Space:
- Media Resource Group List:
- User Hold MOC Audio Source:
- Network Hold MOC Audio Source:
- Location:
- AAR Group:

Association:
- Modify Button Items
- Call Park
- Conference List
- Do Not Disturb
- End Call
- Forward All
- Group Call Pickup
- Hold
- Hunt Group Logout
- Intercom (1 - Add a new Intercom)
- Muisic Call Identification
- Meet Me Conference
- Mobility

AAI Group
- User Local
- Network Location
- Built-In Bridge
- Privacy
- Device Mobility Mode
- Owner
- Owner User ID
- Phone Personalization
- Services Provisioning
- Phone Name
- Single Button Range
- Join Across Lines
- Use Traversal Relay Point
- BLF Audible Alert Setting (Phone Digi)
- BLF Audible Alert Setting (Phone Busy)
- Always Use Primary Line
- Always Use Primary Line for Voice Messages
- Geolocation
- [Check box] Retry Video Call as Audio
- [Check box] Ignore Presentation Indicators (internal calls only)
- [Check box] Allow Control of Device from CTI
- [Check box] Logged Into Hunt Group

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## Number Presentation Transformation

**Caller ID For Calls From This Phone**
- Calling Party Transformation CSS: [Select CSS]
- Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone)

**Remote Number**
- Calling Party Transformation CSS: [Select CSS]
- Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)

## Protocol Specific Information

- Packet Capture Mode: [Select Mode]
- Packet Capture Duration: [Select Duration]
- SLF Presence Group: [Select Group]
- Device Security Profile: [Select Profile]
- SUBSCRIBE Calling Search Space: [Select Search Space]
- Unattended Port: [Select Unattended]
- Require DTMF Reception: [Select Reception]
- RFC2833 Disabled: [Select RFC2833]

## Certification Authority Proxy Function (CAPF) Information

- Certificate Operation: [Select Operation]
- Authentication Mode: [Select Mode]
- Authentication String: [Enter String]
- Key Size (Bits): [Enter Size]
- Certificate Operation Status: [Enter Status]
- Expansion Module Information:
  - Module 1: [Enter Name]
  - Module 1 Load Name: [Enter Name]
  - Module 2: [Enter Name]
  - Module 2 Load Name: [Enter Name]

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

- Information
- Directory
- Messages
- Services
- Authentication Server
- Proxy Server
- Idle URL: [Enter URL]
- Idle Timer (seconds): [Enter Time]
- Secure Authentication URL: [Enter URL]
### Extension Information
- **Enable Extension Mobility**: False
- **Log Out Profile**: Use Current Device Settings
- **Log In Time**: < None >
- **Log Out Time**: < None >

### HII and Confidential Access Level Information
- **HII Domain**: < None >
- **HII Indication**: Default
- **HII Presentation**: Default
- **Confidential Access Mode**: < None >
- **Confidential Access Level**: < None >

### Do Not Disturb
- **Do Not Disturb**: False
- **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>Parameter</th>
<th>Value</th>
<th>Override Common Settings</th>
</tr>
</thead>
<tbody>
<tr>
<td>Disable Speakerphone</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Disable Speakerphone and Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Forwarding Delay</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>PC Port</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Settings Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Global Call Access</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Global Region Access</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Global Voice VLAN Access</td>
<td>Enabled</td>
<td></td>
</tr>
<tr>
<td>Video Capabilities</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Auto Line Select</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Web Access</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Enable Power Save Plus</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Phone On Time</td>
<td>00:00</td>
<td></td>
</tr>
<tr>
<td>Phone Off Time</td>
<td>24:00</td>
<td></td>
</tr>
<tr>
<td>Phone Off Idle Timeout</td>
<td>60</td>
<td></td>
</tr>
<tr>
<td>Enable Audible Alert</td>
<td></td>
<td></td>
</tr>
<tr>
<td>EnergyWise Domain</td>
<td></td>
<td></td>
</tr>
<tr>
<td>EnergyWise Endpoint Security Secret</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Allow EnergyWise Override</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Feature</td>
<td>Setting</td>
<td></td>
</tr>
<tr>
<td>---------------------------------------------</td>
<td>----------</td>
<td></td>
</tr>
<tr>
<td>Allow Energywise Overides</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Open to PC Port</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Logging Display</td>
<td>PC Controlled</td>
<td></td>
</tr>
<tr>
<td>Load Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Recording Tone</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Local Volume</td>
<td>100</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Remote Volume</td>
<td>60</td>
<td></td>
</tr>
<tr>
<td>Recording Tone Duration</td>
<td></td>
<td></td>
</tr>
<tr>
<td>RTPC*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>&quot;more&quot; Soft Key Timer</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Auto Call Select</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Log Server</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Advertise G-722 Codec*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Wideband Headset UE Control</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Wideband Headset</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Poor Firmware Sharing*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): Switch Port*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Cisco Discovery Protocol (CDP): PC Port*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Link Layer Discovery Protocol (LLDP): PC Port*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>LLDP Asset ID</td>
<td></td>
<td></td>
</tr>
<tr>
<td>LLDP Power Priority*</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Wireless Headset Mockswitch Control*</td>
<td>Disabled</td>
<td></td>
</tr>
<tr>
<td>Display Refresh Rate*</td>
<td>Normal</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<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>DSSCN 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>Headset 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>SIM 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 SRTP*</td>
<td></td>
</tr>
</tbody>
</table>
### Directory Number Configuration

**Status:** Ready

#### Directory Number Information
- **Directory Number:** 2154
- **Route Partition:** <None>
- **Description:** Brie Phone 1
- **Alerting Name:** Brie Phone 1
- **External Call Control Profile:** <None>
- **Allow Control of Device from CTI:**
- **Associated Devices:** SEP44460008C04
  - **Edit Device**
  - **Edit Line Appearance**
- **Associate Devices:**

#### Directory Number Settings
- **Voice Mail Profile:** Default_VM
- **Calling Search Space:** Phone-CSS
- **DLP Presence Group:**
- **User Hold MOM Audio Source:** <None>
- **Network Hold MOM Audio Source:** <None>
- **Auto Answer:** Auto Answer Off
- **Reject Anonymous Calls:**

#### Enterprise Alternate Number
- **Add Enterprise Alternate Number**

#### +E.164 Alternate Number
- **Add +E.164 Alternate Number**

#### Directory URIs
<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>

#### PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing
- **Advertised Failover Number:** <None>

#### AAR Settings
- **AAR:** Voice Mail
- **AAR Destination Mask:** <None>
- **AAR Group:**

#### Call Forward and Call Pickup Settings
- **Voice Mail:**
  - **Destination:** Use System Default
  - **Calling Search Space:** Phone-CSS
VG224 Analog Gateway (FAX application)
VG24 Analog Gateway Directory Number Configuration

Directory Number Configuration

Status

Directory Number Information

Directory Number: 2161
Description: Phone 3
Alerting Name: Phone 3
ASCII Alerting Name: Phone 3
External Call Control Profile: < None >
Associated Devices: AALN0210#BELL_VG24

Directory Number Settings

Voice Mail Profile: < None >
Calling Search Space: < None >
BLF Presence Group: Standard Presence group
User Held MOC Audio Source: < None >
Network Held MOC Audio Source: < None >
Reject Anonymous Calls

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### Network Hold MOC Audio Source
- `None`
- `Reject Anonymous Calls`

### Enterprise Alternate Number
- `Add Enterprise Alternate Number`

### +E.164 Alternate Number
- `Add +E.164 Alternate Number`

### Directory URIs
<table>
<thead>
<tr>
<th>Primary</th>
<th>URI</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>

### PSTN Failover for Enterprise Alternate Number, +E.164 Alternate Number, and URI Dialing
- `Advertised Failure 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>AAR <code>or</code></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
- Retain this destination in the call forwarding history

### Call Forward and Call Pickup Settings
#### Calling Search Space Activation Policy
- Forward All
- Use System Default

#### Secondary Calling Search Space for Forward All
- Forward Busy Internal
- Forward Busy External
- Forward No Answer Internal
- Forward No Answer External
- Forward No Coverage Internal
- Forward No Coverage External
- Forward on CTI Failure
- Forward Unregistered Internal
- Forward Unregistered External

- `No Answer Ring Duration (seconds)`: `None`
- Call Pickup Group `None`

### Park Monitoring
<table>
<thead>
<tr>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Park Monitoring forward no retrieve destination external</td>
<td><code>None</code></td>
<td>A blank value means to call the parker’s line.</td>
</tr>
<tr>
<td>Park Monitoring forward no retrieve destination internal</td>
<td><code>None</code></td>
<td>A blank value means to call the parker’s line.</td>
</tr>
</tbody>
</table>

### MLPP Alternate Party And Confidential Access Level Settings
- Target (Destination)
<table>
<thead>
<tr>
<th><strong>MLPP Calling Search Space</strong></th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>MLPP No Answer Ring Duration (seconds)</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Confidential Access Mode</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Confidential Access Level</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Call Control Agent Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</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**

<table>
<thead>
<tr>
<th>Display (Caller ID)</th>
<th>VG224 PORT 0 - FAX LINE 1</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>ASCII Display (Caller ID)</strong></td>
<td>VG224 PORT 0 - FAX LINE 1</td>
</tr>
<tr>
<td><strong>External Phone Number Mask</strong></td>
<td>316535321XX</td>
</tr>
</tbody>
</table>

**Multiple Call/Call Waiting Settings on Device AALN/S2/08BELL_VG224**

- **Maximum Number of Calls**: 2
- **Busy Trigger**: Less than or equal to Max. Calls

**Forwarded Call Information Display on Device AALN/S2/08BELL_VG224**

<table>
<thead>
<tr>
<th>Caller Name</th>
</tr>
</thead>
</table>

**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-rtp
mgcp sdq 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 2/2
!
dial-peer voice 99923 pots
  service mgcpapp
   port 2/3
!
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>G729</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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<th>Americas Headquarters</th>
<th>Asia Pacific Headquarters</th>
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<td>Cisco Systems, Inc. Capital Tower 168 Robinson Road #22-01 to #29-01 Singapore 068912 <a href="http://www.cisco.com">www.cisco.com</a></td>
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<td>Tel: +65 317 7777</td>
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<tr>
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<td>1101 CH Amsterdam</td>
<td>Fax: 408 527-0883</td>
<td>Fax: +65 317 7799</td>
</tr>
<tr>
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<td>The Netherlands</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Tel: 408 526-4000</td>
<td>www-europe.cisco.com</td>
<td></td>
<td></td>
</tr>
<tr>
<td>800 553-NETS (6387)</td>
<td>Tel: 31 0 20 357 1000</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Fax: 408 526-4100</td>
<td>Fax: 31 0 20 357 1100</td>
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</tbody>
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