



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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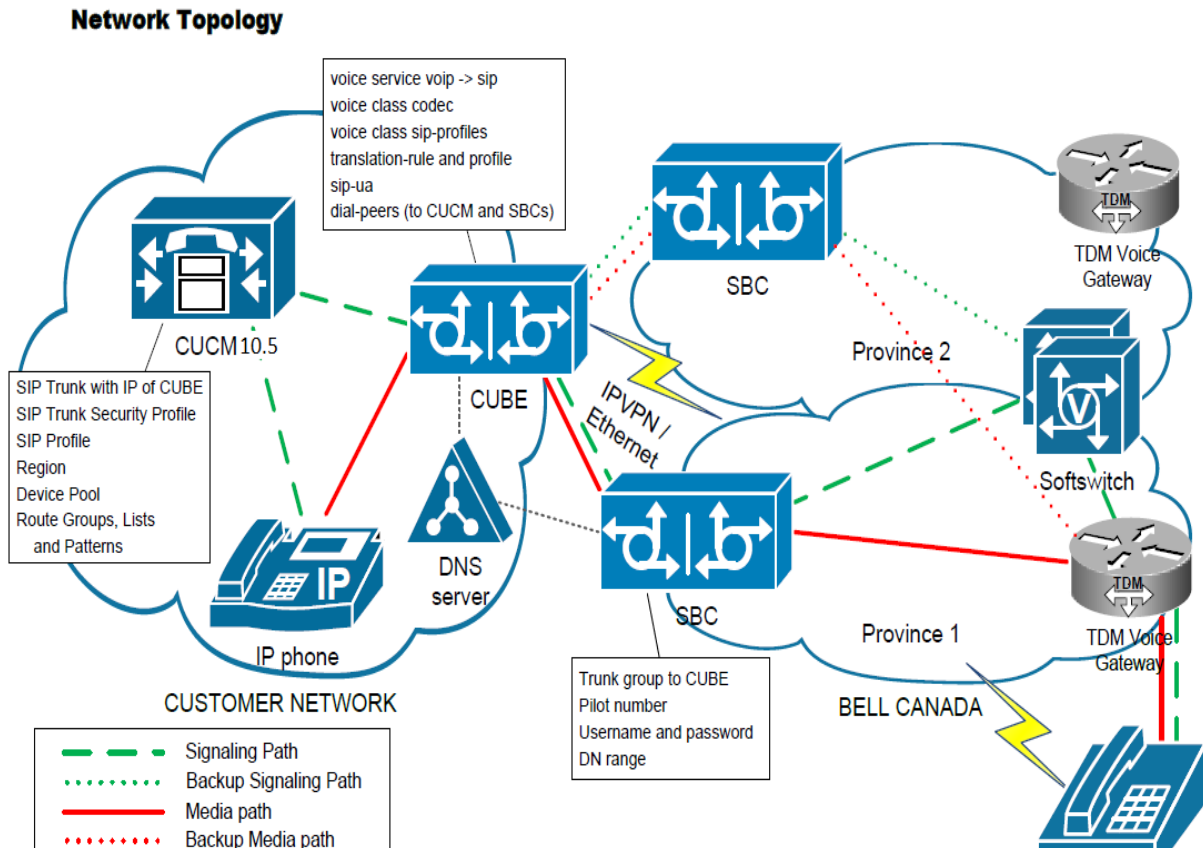
Introduction

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 CISC02921/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
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-keepalive down-interval 20
dtmf-relay rtp-nte
!
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 trgp 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

Name*	BELL SIP Profile
Description	BELL SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> 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
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	

Parameters used in Phone

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Start Media Port*	16384
Stop Media Port*	32766
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None
DTMF DB Level*	Nominal
Call Hold Ring Back*	Off
Anonymous Call Block*	Off
Caller ID Blocking*	Off



Caller ID Blocking*	Off								
Do Not Disturb Control*	User								
Telnet Level for 7940 and 7960*	Disabled								
Resource Priority Namespace	< None >								
Timer Keep Alive Expires (seconds)*	120								
Timer Subscribe Expires (seconds)*	120								
Timer Subscribe Delta (seconds)*	5								
Maximum Redirections*	70								
Off Hook To First Digit Timer (milliseconds)*	15000								
Call Forward URI*	x-cisco-serviceuri-cfwdall								
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial								
<input checked="" type="checkbox"/> Conference Join Enabled									
<input type="checkbox"/> RFC 2543 Hold									
<input checked="" type="checkbox"/> Semi Attended Transfer									
<input type="checkbox"/> Enable VAD									
<input type="checkbox"/> Stutter Message Waiting									
<input type="checkbox"/> MLPP User Authorization									
Normalization Script									
Normalization Script < None >									
<input type="checkbox"/> Enable Trace									
<table><thead><tr><th></th><th>Parameter Name</th><th>Parameter Value</th><th></th></tr></thead><tbody><tr><td>1</td><td></td><td></td><td>+ -</td></tr></tbody></table>			Parameter Name	Parameter Value		1			+ -
	Parameter Name	Parameter Value							
1			+ -						
Incoming Requests FROM URI Settings									
Caller ID DN									
Caller Name									
Trunk Specific Configuration									
Reroute Incoming Request to new Trunk based on* Never									
RSVP Over SIP* Local RSVP									
Resource Priority Namespace List < None >									
<input checked="" type="checkbox"/> Fall back to local RSVP									
SIP Rel1XX Options* Disabled									
Video Call Traffic Class* Mixed									
Calling Line Identification Presentation* Default									
Session Refresh Method* Invite									
Early Offer support for voice and video calls* Mandatory (insert MTP if needed)									
<input type="checkbox"/> Enable ANAT									
<input type="checkbox"/> Deliver Conference Bridge Identifier									
<input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information									
<input type="checkbox"/> Reject Anonymous Incoming Calls									
<input type="checkbox"/> Reject Anonymous Outgoing Calls									
<input type="checkbox"/> Send ILS Learned Destination Route String									
SIP OPTIONS Ping									
<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"									
Ping Interval for In-service and Partially In-service Trunks (seconds)* 60									
Ping Interval for Out-of-service Trunks (seconds)* 120									
Ping Retry Timer (milliseconds)* 500									
Ping Retry Count* 6									



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

[Save](#) [Delete](#) [Copy](#) [Reset](#) [Apply Config](#) [Add New](#)

*- indicates required item.

SIP Profile Configuration

Related Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Copy](#) [Reset](#) [Apply Config](#) [Add New](#)

Status

Status: Ready

All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information

Name*	BELL FAX SIP Profile
Description	BELL FAX SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agen
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> 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
Accept Audio Codec Preferences in Received Offer*	Default
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	



☐ Allow RR/RS bandwidth modifier (RFC 3556)

Parameters used in Phone

Timer Invite Expires (seconds)*	<input type="text" value="180"/>
Timer Register Delta (seconds)*	<input type="text" value="5"/>
Timer Register Expires (seconds)*	<input type="text" value="3600"/>
Timer T1 (msec)*	<input type="text" value="500"/>
Timer T2 (msec)*	<input type="text" value="4000"/>
Retry INVITE*	<input type="text" value="5"/>
Retry Non-INVITE*	<input type="text" value="10"/>
Start Media Port*	<input type="text" value="16384"/>
Stop Media Port*	<input type="text" value="32766"/>
Call Pickup URI*	<input type="text" value="x-cisco-serviceuri-pickup"/>
Call Pickup Group Other URI*	<input type="text" value="x-cisco-serviceuri-opickup"/>
Call Pickup Group URI*	<input type="text" value="x-cisco-serviceuri-gpickup"/>
Meet Me Service URI*	<input type="text" value="x-cisco-serviceuri-meetme"/>
User Info*	<input type="text" value="None"/>
DTMF DB Level*	<input type="text" value="Nominal"/>
Call Hold Ring Back*	<input type="text" value="Off"/>
Anonymous Call Block*	<input type="text" value="Off"/>
Caller ID Blocking*	<input type="text" value="Off"/>
Do Not Disturb Control*	<input type="text" value="User"/>
Telnet Level for 7940 and 7960*	<input type="text" value="Disabled"/>
Resource Priority Namespace	<input type="text" value=" < None >"/>

Timer Keep Alive Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Expires (seconds)*	<input type="text" value="120"/>
Timer Subscribe Delta (seconds)*	<input type="text" value="5"/>
Maximum Redirections*	<input type="text" value="70"/>
Off Hook To First Digit Timer (milliseconds)*	<input type="text" value="15000"/>
Call Forward URI*	<input type="text" value="x-cisco-serviceuri-cfwdall"/>
Speed Dial (Abbreviated Dial) URI*	<input type="text" value="x-cisco-serviceuri-abbrdial"/>
<input checked="" type="checkbox"/> Conference Join Enabled	
<input type="checkbox"/> RFC 2543 Hold	
<input checked="" type="checkbox"/> Semi Attended Transfer	
<input type="checkbox"/> Enable VAD	
<input type="checkbox"/> Stutter Message Waiting	
<input type="checkbox"/> MLPP User Authorization	

Normalization Script

Normalization Script

☐ Enable Trace

	Parameter Name	Parameter Value		
1			<input type="button" value="+"/>	<input type="button" value="-"/>

Incoming Requests FROM URI Settings

Caller ID DN	<input type="text"/>
Caller Name	<input type="text"/>

Trunk Specific Configuration



Reroute Incoming Request to new Trunk based on*	Never
RSVP Over SIP*	Local RSVP
Resource Priority Namespace List	< None >
<input checked="" type="checkbox"/> Fall back to local RSVP	
SIP Rel1XX Options*	Disabled
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Mandatory (insert MTP if needed)
<input type="checkbox"/> Enable ANAT	
<input type="checkbox"/> Deliver Conference Bridge Identifier	
<input type="checkbox"/> Allow Passthrough of Configured Line Device Caller Information	
<input type="checkbox"/> Reject Anonymous Incoming Calls	
<input type="checkbox"/> Reject Anonymous Outgoing Calls	
<input type="checkbox"/> Send ILS Learned Destination Route String	
SIP OPTIONS Ping	
<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	5
SDP Information	
<input type="checkbox"/> Send send-receive SDP in mid-call INVITE	

SDP Information	
<input type="checkbox"/> Send send-receive SDP in mid-call INVITE	
<input type="checkbox"/> Allow Presentation Sharing using BFCP	
<input type="checkbox"/> Allow iX Application Media	
<input type="checkbox"/> Allow multiple codecs in answer SDP	

Save Delete Copy Reset Apply Config Add New

*- indicates required item.

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		Related Links: Back To Find/List Go
<div>Save Delete Reset Add New</div>		
Status Status: Ready		
SIP Trunk Status Service Status: Full Service Duration: Time In Full Service: 8 days 21 hours 52 minutes		
Device Information		
Product:	SIP Trunk	
Device Protocol:	SIP	
Trunk Service Type	None(Default)	
Device Name*	<input type="text" value="BELLSIPTRUNK"/>	
Description	<input type="text" value="BELL SIP TRUNK"/>	
Device Pool*	<input type="text" value="BELL SIP DP"/>	
Common Device Configuration	< None >	
Call Classification*	Use System Default	
Media Resource Group List	< None >	
Location*	Hub_None	
AAR Group	< None >	
Tunneled Protocol*	None	
QSIG Variant*	No Changes	
ASN.1 ROSE OID Encoding*	No Changes	
Packet Capture Mode*	None	
Packet Capture Duration	<input type="text" value="0"/>	
<input checked="" type="checkbox"/> Media Termination Point Required		
<input type="checkbox"/> Retry Video Call as Audio		
<input checked="" type="checkbox"/> Path Replacement Support		
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name		
<input checked="" type="checkbox"/> Transmit UTF-8 Names in QSIG APDU		
<input type="checkbox"/> Unattended Port		
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.		
Consider Traffic on This Trunk Secure*	<input type="text" value="When using both sRTP and TLS"/>	
Route Class Signaling Enabled*	<input type="text" value="Default"/>	
Use Trusted Relay Point*	<input type="text" value="Default"/>	
<input checked="" type="checkbox"/> PSTN Access		
<input type="checkbox"/> Run On All Active Unified CM Nodes		
Intercompany Media Engine (IME)		
E.164 Transformation Profile	<input type="text" value="< None >"/>	
MLPP and Confidential Access Level Information		
MLPP Domain	<input type="text" value="< None >"/>	
Confidential Access Mode	<input type="text" value="< None >"/>	
Confidential Access Level	<input type="text" value="< None >"/>	
Call Routing Information		
<input checked="" type="checkbox"/> Remote-Party-Id		
<input checked="" type="checkbox"/> Asserted-Identity		
Asserted-Type*	<input type="text" value="PAI"/>	
SIP Privacy*	<input type="text" value="Default"/>	



Inbound Calls

Significant Digits*
All

Connected Line ID Presentation*
Default

Connected Name Presentation*
Default

Calling Search Space
< None >

AAR Calling Search Space
< None >

Prefix DN

☒ 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.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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.

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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



SIP Information

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Time
1 *	10.220.33.2		5060	up		

MTP Preferred Originating Codec*
BLF Presence Group*
SIP Trunk Security Profile*
Rerouting Calling Search Space
Out-Of-Dialog Refer Calling Search Space
SUBSCRIBE Calling Search Space
SIP Profile*
DTMF Signaling Method*

G729/G729a

Standard Presence group

BELL SIP Trunk Profile

< None >

< None >

< None >

BELL SIP Profile

RFC 2833

View Details

Normalization Script

Normalization Script
Enable Trace

	Parameter Name	Parameter Value		
1			+	-

Recording Information

☒ None

☐ This trunk connects to a recording-enabled gateway

☐ This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation
Geolocation Filter
☐ Send Geolocation Information

< None >

< None >

Save

Delete

Reset

Add New

i

*- indicates required item.

i

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



Trunk Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Reset Add New

Status

Status: Ready

SIP Trunk Status

Service Status: Full Service

Duration: Time In Full Service: 9 days 23 hours 30 minutes

Device Information

Product:

SIP Trunk

Device Protocol:

SIP

Trunk Service Type:

None(Default)

Device Name*:

BELLFAXSIPTRUNK

Description:

BELL FAX SIP TRUNK

Device Pool*:

BELL FAX SIP DP

Common Device Configuration:

< None >

Call Classification*:

Use System Default

Media Resource Group List:

BELL SIP MRGL

Location*:

Hub_None

AAR Group:

< None >

Tunneled Protocol*:

None

QSIG Variant*:

No Changes

ASN.1 ROSE OID Encoding*:

No Changes

Packet Capture Mode*:

None

Packet Capture Duration:

0

☐ Media Termination Point Required

☒ Retry Video Call as Audio

☐ Path Replacement Support

☐ Transmit UTF-8 for Calling Party Name

☐ Transmit UTF-8 Names in QSIG APDU

☐ Unattended Port

☐ SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information. Consider Traffic on This Trunk Secure*

When using both sRTP and TLS

Route Class Signaling Enabled*

Default

Use Trusted Relay Point*

Default

☒ PSTN Access

☐ Run On All Active Unified CM Nodes

Intercompany Media Engine (IME)

E.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

☒ Asserted-Identity

Asserted-Type*

Default

SIP Privacy*

Default



Inbound Calls
Significant Digits*
Connected Line ID Presentation*
Connected Name Presentation*
Calling Search Space
AAR Calling Search Space
Prefix DN
☐ 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.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="< None >"/>	<input checked="" type="checkbox"/>

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.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value="< None >"/>	<input checked="" type="checkbox"/>

Connected Party Settings
Connected Party Transformation CSS
☒ Use Device Pool Connected Party Transformation CSS

Outbound Calls
Called Party Transformation CSS
☒ Use Device Pool Called Party Transformation CSS
Calling Party Transformation CSS
☒ Use Device Pool Calling Party Transformation CSS
Calling Party Selection*
Calling Line ID Presentation*
Calling Name Presentation*
Calling and Connected Party Info Format*
☐ Redirecting Diversion Header Delivery - Outbound
Redirecting Party Transformation 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

Destination

☐ Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Time
1 *	10.220.33.17		5060	up		

MTP Preferred Originating Codec*
711ulaw

BLF Presence Group*
Standard Presence group

SIP Trunk Security Profile*
BELL SIP Trunk Profile

Rerouting Calling Search Space
< None >

Out-Of-Dialog Refer Calling Search Space
< None >

SUBSCRIBE Calling Search Space
< None >

SIP Profile*
BELL FAX SIP Profile [View Details](#)

DTMF Signaling Method*
No Preference

Normalization Script

Normalization Script
< None >

☐ Enable Trace

	Parameter Name	Parameter Value
1		

Recording Information☒ None
☐ This trunk connects to a recording-enabled gateway
☐ This trunk connects to other clusters with recording-enabled gateways**Geolocation Configuration**Geolocation
< None >Geolocation Filter
< None >☐ Send Geolocation Information

SaveDeleteResetAdd New

i

 *- indicates required item.

i

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

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.



SIP Trunk Security Profile Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

SIP Trunk Security Profile Information

Name*	BELL SIP Trunk Profile
Description	BELL Non Secure SIP Trunk Profile authenticated by nul
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	UDP
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	600
X.509 Subject Name	
Incoming Port*	5060
<input type="checkbox"/> Enable Application level authorization	
<input checked="" type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	Use Default Filter

Save Delete Copy Reset Apply Config Add New

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 ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)

Region Information
Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
BELL FAX SIP REGION	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
BELL SIP REGION	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<div>BELL FAX SIP REGION</div> <div>BELL SIP REGION</div>	<div>Keep Current Setting</div>	<div><input checked="" type="radio"/> Keep Current Setting</div> <div><input type="radio"/> <input type="text" value=""/> kbps</div>	<div><input checked="" type="radio"/> Keep Current Setting</div> <div><input type="radio"/> Use System Default</div> <div><input type="radio"/> None</div> <div><input type="radio"/> <input type="text" value=""/> kbps</div>	<div><input checked="" type="radio"/> Keep Current Setting</div> <div><input type="radio"/> Use System Default</div> <div><input type="radio"/> None</div> <div><input type="radio"/> <input type="text" value=""/> kbps</div>

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)

Region ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)

Region Information
Name*

Region Relationships

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
BELL FAX SIP REGION	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
BELL SIP REGION	Use System Default (Factory Default low loss)	64 kbps (G.722, G.711)	384 kbps	2147483647 kbps
NOTE: Regions not displayed	Use System Default	Use System Default	Use System Default	Use System Default

Modify Relationship to other Regions

Regions	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls	Maximum Session Bit Rate for Immersive Video Calls
<div>BELL FAX SIP REGION</div> <div>BELL SIP REGION</div>	<div>Keep Current Setting</div>	<div><input checked="" type="radio"/> Keep Current Setting</div> <div><input type="radio"/> <input type="text" value=""/> kbps</div>	<div><input checked="" type="radio"/> Keep Current Setting</div> <div><input type="radio"/> Use System Default</div> <div><input type="radio"/> None</div> <div><input type="radio"/> <input type="text" value=""/> kbps</div>	<div><input checked="" type="radio"/> Keep Current Setting</div> <div><input type="radio"/> Use System Default</div> <div><input type="radio"/> None</div> <div><input type="radio"/> <input type="text" value=""/> kbps</div>

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)



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 Pool ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Copy](#) [Reset](#) [Apply Config](#) [Add New](#)

Status
 Status: Ready

Device Pool Information
Device Pool: BELL SIP DP (8 members**)

Device Pool Settings
Device Pool Name* BELL SIP DP
Cisco Unified Communications Manager Group* Default
Calling Search Space for Auto-registration < None >
Adjunct CSS < None >
Reverted Call Focus Priority Default
Intercompany Media Services Enrolled Group < None >

Local Route Group Settings
Standard Local Route Group < None >

Roaming Sensitive Settings
Date/Time Group* CMLocal
Region* BELL SIP REGION
Media Resource Group List BELL SIP MRGL
Location Hub_None
Network Locale < None >
SRST Reference* Use Default Gateway
Connection Monitor Duration***



Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >
Wireless LAN Profile Group	< None > View Details

Device Mobility Related Information****

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

Geolocation Configuration

Geolocation	< None >
Geolocation Filter	< None >

Call Routing Information

Incoming Calling Party Settings

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

Clear Prefix Settings		Default Prefix Settings	
Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >
Unknown Number	Default		< None >
Subscriber Number	Default		< None >

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.

Clear Prefix Settings		Default Prefix Settings	
Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

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 >
--------------------------------------	----------

[Save](#) [Delete](#) [Copy](#) [Reset](#) [Apply Config](#) [Add New](#)



Device Pool Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Reset Apply Config Add New

Status

Status: Ready

Device Pool Information

Device Pool: BELL FAX SIP DP (2 members**)

Device Pool Settings

Device Pool Name*

BELL FAX SIP DP

Cisco Unified Communications Manager Group*

Default

Calling Search Space for Auto-registration

< None >

Adjunct CSS

< None >

Reverted Call Focus Priority

Default

Intercompany Media Services Enrolled Group

< None >

Local Route Group Settings

Standard Local Route Group

< None >

Roaming Sensitive Settings

Date/Time Group*

CMLocal

Region*

BELL FAX SIP REGION

Media Resource Group List

BELL SIP MRGL

Location

Hub_None

Network Locale

< None >

SRST Reference*

Use Default Gateway

Connection Monitor Duration***

Single Button Barge*

Default

Join Across Lines*

Default

Physical Location

< None >

Device Mobility Group

< None >

Wireless LAN Profile Group

< None >

[View Details](#)

Device Mobility Related Information****

Device Mobility Calling Search Space

< None >

AAR Calling Search Space

< None >

AAR Group

< None >

Calling Party Transformation CSS

< None >

Called Party Transformation CSS

< None >

Geolocation Configuration

Geolocation

< None >

Geolocation Filter

< None >

Call Routing Information

Incoming Calling Party Settings

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

Clear Prefix Settings

Default Prefix Settings

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default		< None >
International Number	Default		< None >



Unknown Number	Default		< None >
Subscriber Number	Default		< None >

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.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	Default	0	< None >
International Number	Default	0	< None >
Unknown Number	Default	0	< None >
Subscriber Number	Default	0	< None >

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

[+](#) Add New [Select All](#) [Clear All](#) [Delete Selected](#)

Status

i 3 records found

Route Patterns (1 - 3 of 3)

Rows per Page 50

Find Route Patterns where Pattern begins with [Find](#) [Clear Filter](#) [+](#) [-](#)

<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device	Copy
<input type="checkbox"/>	*67.9@	Block Calling Name and Number	PHONE-PT		SIP_RL	Copy
<input type="checkbox"/>	9.@		FAX-PT		FAX_RL	Copy
<input type="checkbox"/>	9.@		PHONE-PT		SIP_RL	Copy

[Add New](#) [Select All](#) [Clear All](#) [Delete Selected](#)

Route Pattern Configuration

Related Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Copy](#) [Add New](#)

Status

i Status: Ready

Pattern Definition

Route Pattern*	<input type="text" value="9.@"/>
Route Partition	<input type="text" value="PHONE-PT"/>
Description	<input type="text"/>
Numbering Plan*	<input type="text" value="NANP"/>
Route Filter	<input type="text" value=" < None >"/>
MLPP Precedence*	<input type="text" value="Default"/>
<input type="checkbox"/> Apply Call Blocking Percentage	<input type="text"/>
Resource Priority Namespace Network Domain	<input type="text" value=" < None >"/>
Route Class*	<input type="text" value="Default"/>
Gateway/Route List*	<input type="text" value="SIP_RL"/> (Edit)
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="No Error"/>
Call Classification*	<input type="text" value="OffNet"/>
External Call Control Profile	<input type="text" value=" < None >"/>
<input type="checkbox"/> Allow Device Override <input checked="" type="checkbox"/> Provide Outside Dial Tone <input type="checkbox"/> Allow Overlap Sending <input type="checkbox"/> Urgent Priority	
<input type="checkbox"/> Require Forced Authorization Code	
Authorization Level*	<input type="text" value="0"/>
<input type="checkbox"/> Require Client Matter Code	

Calling Party Transformations



☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value=" < Not Exist >"/>	<input type="text"/>

Route List Configuration

Status
 2 records found

Route List (1 - 2 of 2) Rows per Page 50

Find Route List where begins with

<input type="checkbox"/>	Name ^	Description	Enabled	Status	IPv4 Address
<input type="checkbox"/>	FAX_RL		true	Registered with 10.220.22.161	10.220.22.161
<input type="checkbox"/>	SIP_RL		true	Registered with 10.220.22.161	10.220.22.161



Route List Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save

Delete

Copy

Reset

Apply Config

Add New

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_RG

Add Route Group

Removed Groups***

Route List Details

SIP_RG

Route List Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save

Delete

Copy

Reset

Apply Config

Add New

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

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_RG

Add Route Group

Removed Groups***

Route List Details

FAX_RG

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Route Group Configuration

Route Group Configuration

Related Links: [Back To Find/List](#) [Go](#)

Save Delete Add New

Route Group Information

Route Group Name*

Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

Available Devices**

BELLFAXSIPTRUNK

BELLSIPTRUNK

Port(s)

Add to Route Group

Current Route Group Members

Selected Devices (ordered by priority)*

BELLSIPTRUNK (All Ports)

Reverse Order of Selected Devices

Removed Devices***

Route Group Members

BELLSIPTRUNK

Route Group Information

Route Group Name*

Distribution Algorithm*

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains

Available Devices**

BELLFAXSIPTRUNK

BELLSIPTRUNK

Port(s)

Add to Route Group

Current Route Group Members

Selected Devices (ordered by priority)*

BELLFAXSIPTRUNK (All Ports)

Reverse Order of Selected Devices

Removed Devices***

Route Group Members

BELLFAXSIPTRUNK



Calling Search Space Configuration

Find and List Calling Search Spaces

Add New Select All Clear All Delete Selected

Status
 3 records found

Calling Search Space (1 - 3 of 3) Rows per Page 50

Find Calling Search Space where CSS Name begins with

<input type="checkbox"/>	CSS Name ^	Description	Copy
<input type="checkbox"/>	Diversion-CSS		
<input type="checkbox"/>	Fax-CSS		
<input type="checkbox"/>	Phone-CSS		

Calling Search Space Configuration Related Links:

Save Delete Copy Add New

Status
 Status: Ready

Calling Search Space Information
Name*
Description

Route Partitions for this Calling Search Space
Available Partitions**

Directory URI
FAX-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*
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

Calling Search Space Information
Name*
Description

Route Partitions for this Calling Search Space
Available Partitions**

Directory URI
Diversion-PT
FAX-PT
Global Learned E164 Numbers
Global Learned E164 Patterns

Selected Partitions

PHONE-PT

Save Delete Copy Add New

Translation Pattern Configuration

Translation Pattern Configuration Related Links: [Back To Find/List](#) [Go](#)

Save Delete Copy Add New

Status
 Status: Ready

Pattern Definition
Translation Pattern
Partition
Description
Numbering Plan
Route Filter
MLPP Precedence*
Resource Priority Namespace Network Domain
Route Class*
Calling Search Space
☐ Use Originator's Calling Search Space
External Call Control Profile
Route Option
☒ Route this pattern
☐ Block 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



Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

*- indicates required item.

Calling Party Transformation Pattern Configuration

Calling Party Transformation Pattern Configuration Related Links: [Back To Find/List](#)

Status

Status: Ready

Pattern Definition

Pattern*

Partition

Description

Numbering Plan

Route Filter

☒ Urgent Priority

☐ MLPP Preemption Disabled

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Discard Digits

Calling Party Transformation Mask

Prefix Digits

Calling Line ID Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

*- indicates required item.



Transcoder Configuration

Transcoder ConfigurationRelated Links: [Back To Find/List](#)

Save Delete Copy Reset Apply Config Add New

Transcoder Information
Transcoder: Transcoder (CUBE Transcode)
Registration: Registered with Cisco Unified Communications Manager 10.220.22.161
IPv4 Address: 10.220.33.2

IOS Transcoder Info
Transcoder Type* Cisco IOS Enhanced Media Termination Point
Description CUBE Transcode
Device Name* Transcoder
Device Pool* BELL SIP DP [View Details](#)
Common Device Configuration < None > [View Details](#)
Special Load Information
☐ Trusted Relay Point Leave blank to use default

Save Delete Copy Reset Apply Config Add New

*- indicates required item.

Media Termination Point

Media Termination Point ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

Save Reset Apply Config

Status
 Status: Ready

Media Termination Point Information
Registration: Registered with Cisco Unified Communications Manager 10.220.22.161
IPv4 Address: 10.220.22.161
Media Termination Point Type* Cisco Media Termination Point Software
Host Server* 10.220.22.161
Media Termination Point Name* MTP_2
Description MTP_cucm105
Device Pool* Default
☐ Trusted Relay Point

Save Reset Apply Config



Conference Bridge Configuration

Conference Bridge Information

Conference Bridge : CFB_2 (CFB_cucm105)
Registration: Registered with Cisco Unified Communications Manager 10.220.22.161
IPv4 Address: 10.220.22.161

Software Conference Bridge Info

Conference Bridge Type* Cisco Conference Bridge Software
Host Server 10.220.22.161

⚠ Device is not trusted

Conference Bridge Name* CFB_2

Description CFB_cucm105

Device Pool* Default

Common Device Configuration < None >

Location* Hub_None

Use Trusted Relay Point* Default

Save Reset Apply Config

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)

Rows per Page 50

Find Media Resource Group where Name begins with Find Clear Filter

<input type="checkbox"/>	Name ^	Description	Multi-cast	Copy
<input type="checkbox"/>	BELL SIP MRG	BELL SIP MRG	false	
<input type="checkbox"/>	Dummy MRG		false	

Add New Select All Clear All Delete Selected



Media Resource Group ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Copy](#) [Add New](#)

Status
 Status: Ready

Media Resource Group Status
Media Resource Group: BELL SIP MRG (used by 10 devices)

Media Resource Group Information
Name*
Description

Devices for this Group
Available Media Resources**

ANN_2
CFB_2
bellmtp
mtpsip

▼ ▲

Selected Media Resources*

Conference (CFB)
MOH_2 (MOH)
MTP_2 (MTP)
Transcoder (XCODE)

☐ Use Multi-cast for MOH Audio (If at least one multi-cast MOH resource is available)

[Save](#) [Delete](#) [Copy](#) [Add New](#)

Media Resource Group List Configuration

Media Resource Group List ConfigurationRelated Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Copy](#) [Add New](#)

Status
 Status: Ready

Media Resource Group List Status
Media Resource Group List: BELL SIP MRGL (used by 10 devices)

Media Resource Group List Information
Name*

Media Resource Groups for this List
Available Media Resource Groups

Dummy_MRG

▼ ▲

Selected Media Resource Groups

BELL SIP MRG

▼ ▲

[Save](#) [Delete](#) [Copy](#) [Add New](#)

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

Related Links: Back To Find/List

Go

Save

Delete

Copy

Reset

Apply Config

Add New

Association

Modify Button Items

1 775 Line [1] - 2154 (no partition)

----- Unassigned Associated Items -----

2 775 Line [2] - Add a new DN

3 Add a new SURF

4 Add a new BLF SD

5 Add a new SD

6 Call Back

7 775 Add a new BLF Directed Call Park

8 Call Park

9 Call Pickup

10 Conference List

11 Conference

12 Do Not Disturb

13 End Call

14 Forward All

15 Group Call Pickup

16 Hold

17 Hunt Group Logout

18 775 Intercom [1] - Add a new Intercom

19 Malicious Call Identification

20 Meet Me Conference

21 Mobility

Phone Type

Product Type: Cisco 7962

Device Protocol: SCCP

Real-time Device Status

Registration: Registered with Cisco Unified Communications Manager 10.220.22.161

IPv4 Address: 10.220.0.101

Active Load ID: SCCP42.9-3-1SR4-1S

Download Status: None

Device Information

☒ Device is Active

☒ Device is trusted

MAC Address* 44ADD9D58C04

Description Niraj Phone 1

Device Pool* BELL SIP DP [View Details](#)

Common Device Configuration < None > [View Details](#)

Phone Button Template* Universal Device Template Button Layout

Softkey Template Standard User

Common Phone Profile* Standard Common Phone Profile [View Details](#)

Calling Search Space Phone-CSS

AAR Calling Search Space < None >

Media Resource Group List BELL SIP MRGL

User Hold MOH Audio Source < None >

Network Hold MOH Audio Source < None >

Location* Hub_None

AAR Group < None >

21 Mobility

22 New Call

23 Other Pickup

24 Quality Reporting Tool

25 Redial

26 Remove Last Participant

27 Transfer

28 Video Mode

29 Queue Status

30 Privacy

31 None

AAR Group < None >

User Locale < None >

Network Locale < None >

Built In Bridge* Default

Privacy* Default

Device Mobility Mode* Default [View Current Device Mobility Settings](#)

Owner ☐ User ☒ Anonymous (Public/Shared Space)

Owner User ID

Phone Personalization* Default

Services Provisioning* Default

Phone Load Name

Single Button Barge Default

Join Across Lines Default

Use Trusted Relay Point* Default

BLF Audible Alert Setting (Phone Idle)* Default

BLF Audible Alert Setting (Phone Busy)* Default

Always Use Prime Line* Default

Always Use Prime Line for Voice Message* Default

Geolocation < None >

☒ Retry Video Call as Audio

☐ Ignore Presentation Indicators (internal calls only)


☒ Allow Control of Device from CTI

☒ Logged Into Hunt Group



<input type="checkbox"/> Remote Device <input type="checkbox"/> Protected Device **** <input type="checkbox"/> Hot line Device ***** <input type="checkbox"/> Require off-premise location
Number Presentation Transformation Caller ID For Calls From This Phone Calling Party Transformation CSS <input type="text" value=" < None >"/> <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Caller ID For Calls From This Phone) Remote Number Calling Party Transformation CSS <input type="text" value=" < None >"/> <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS (Device Mobility Related Information)
Protocol Specific Information Packet Capture Mode* <input type="text" value="None"/> Packet Capture Duration <input type="text" value="0"/> BLF Presence Group* <input type="text" value="Standard Presence group"/> Device Security Profile* <input type="text" value="Universal Device Template - Model-independent Se"/> SUBSCRIBE Calling Search Space <input type="text" value=" < None >"/> <input type="checkbox"/> Unattended Port <input type="checkbox"/> Require DTMF Reception <input type="checkbox"/> RFC2833 Disabled
Certification Authority Proxy Function (CAPF) Information Certification Authority Proxy Function (CAPF) Information Certificate Operation* <input type="text" value="No Pending Operation"/> Authentication Mode* <input type="text" value="By Null String"/> Authentication String <input type="text"/> <input type="button" value="Generate String"/> Key Size (Bits)* <input type="text" value="1024"/> Operation Completes By <input type="text" value="2014 12 21 12"/> (YYYY:MM:DD:HH) Certificate Operation Status: None Note: Security Profile Contains Addition CAPF Settings.
Expansion Module Information Module 1 <input type="text" value=" < None >"/> Module 1 Load Name <input type="text"/> Module 2 <input type="text" value=" < None >"/> Module 2 Load Name <input type="text"/>
External Data Locations Information (Leave blank to use default) Information <input type="text"/> Directory <input type="text"/> Messages <input type="text"/> Services <input type="text"/> Authentication Server <input type="text"/> Proxy Server <input type="text"/> Idle <input type="text" value="https://cucm105:443/cucm-uds/xps/selfProvision"/> Idle Timer (seconds) <input type="text" value="1"/> Secure Authentication URL <input type="text"/>



Extension Information																																																											
<input type="checkbox"/> Enable Extension Mobility																																																											
Log Out Profile -- Use Current Device Settings --																																																											
Log in Time < None >																																																											
Log out Time < None >																																																											
MLPP and Confidential Access Level Information																																																											
MLPP Domain	< None >																																																										
MLPP Indication*	Default																																																										
MLPP Preemption*	Default																																																										
Confidential Access Mode	< None >																																																										
Confidential Access Level	< None >																																																										
Do Not Disturb																																																											
<input type="checkbox"/> Do Not Disturb																																																											
DND Option*	Use Common Phone Profile Setting																																																										
DND Incoming Call Alert	< None >																																																										
Secure Shell Information																																																											
Secure Shell User																																																											
Secure Shell Password																																																											
Product Specific Configuration Layout																																																											
																																																											
<table border="1"><thead><tr><th></th><th>Parameter Value</th><th>Override Common Settings</th></tr></thead><tbody><tr><td><input type="checkbox"/> Disable Speakerphone</td><td></td><td></td></tr><tr><td><input type="checkbox"/> Disable Speakerphone and Headset</td><td></td><td></td></tr><tr><td>Forwarding Delay*</td><td>Disabled</td><td></td></tr><tr><td>PC Port *</td><td>Enabled</td><td></td></tr><tr><td>Settings Access*</td><td>Enabled</td><td><input type="checkbox"/></td></tr><tr><td>Gratuitous ARP*</td><td>Disabled</td><td></td></tr><tr><td>PC Voice VLAN Access*</td><td>Enabled</td><td></td></tr><tr><td>Video Capabilities*</td><td>Disabled</td><td><input type="checkbox"/></td></tr><tr><td>Auto Line Select*</td><td>Disabled</td><td></td></tr><tr><td>Web Access*</td><td>Disabled</td><td><input type="checkbox"/></td></tr><tr><td>Enable Power Save Plus</td><td>Sunday Monday Tuesday</td><td><input type="checkbox"/></td></tr><tr><td>Phone On Time</td><td>00:00</td><td><input type="checkbox"/></td></tr><tr><td>Phone Off Time</td><td>24:00</td><td><input type="checkbox"/></td></tr><tr><td>Phone Off Idle Timeout*</td><td>60</td><td><input type="checkbox"/></td></tr><tr><td><input type="checkbox"/> Enable Audible Alert</td><td></td><td><input type="checkbox"/></td></tr><tr><td>EnergyWise Domain</td><td></td><td><input type="checkbox"/></td></tr><tr><td>EnergyWise Endpoint Security Secret</td><td></td><td><input type="checkbox"/></td></tr><tr><td><input type="checkbox"/> Allow EnergyWise Overrides</td><td></td><td><input type="checkbox"/></td></tr></tbody></table>				Parameter Value	Override Common Settings	<input type="checkbox"/> Disable Speakerphone			<input type="checkbox"/> Disable Speakerphone and Headset			Forwarding Delay*	Disabled		PC Port *	Enabled		Settings Access*	Enabled	<input type="checkbox"/>	Gratuitous ARP*	Disabled		PC Voice VLAN Access*	Enabled		Video Capabilities*	Disabled	<input type="checkbox"/>	Auto Line Select*	Disabled		Web Access*	Disabled	<input type="checkbox"/>	Enable Power Save Plus	Sunday Monday Tuesday	<input type="checkbox"/>	Phone On Time	00:00	<input type="checkbox"/>	Phone Off Time	24:00	<input type="checkbox"/>	Phone Off Idle Timeout*	60	<input type="checkbox"/>	<input type="checkbox"/> Enable Audible Alert		<input type="checkbox"/>	EnergyWise Domain		<input type="checkbox"/>	EnergyWise Endpoint Security Secret		<input type="checkbox"/>	<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>
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PC Voice VLAN Access*	Enabled																																																										
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<input type="checkbox"/> Allow EnergyWise Overrides		<input type="checkbox"/>																																																									



<input type="checkbox"/> Allow energywise Overrides	
Span to PC Port*	Disabled
Logging Display*	PC Controlled
Load Server	
Recording Tone*	Disabled
Recording Tone Local Volume*	100
Recording Tone Remote Volume*	50
Recording Tone Duration	
RTCP*	Disabled
"more" Soft Key Timer	5
Auto Call Select*	Enabled
Log Server	
Advertise G.722 Codec*	Use System Default
Wideband Headset UI Control*	Enabled
Wideband Headset*	Enabled
Peer Firmware Sharing*	Enabled
Cisco Discovery Protocol (CDP): Switch Port*	Enabled
Cisco Discovery Protocol (CDP): PC Port*	Enabled
Link Layer Discovery Protocol - Media Endpoint Discover (LLDP-MED): Switch Port*	Enabled
Link Layer Discovery Protocol (LLDP): PC Port*	Enabled
LLDP Asset ID	
LLDP Power Priority*	Unknown
Wireless Headset Hookswitch Control*	Disabled
Display Refresh Rate*	Normal

IPv6 Load Server	
IPv6 Log Server	
802.1x Authentication*	User Controlled
Detect Unified CM Connection Failure*	Normal
Minimum Ring Volume*	0-Silent
Headset Sidetone Level*	Default
Headset Send Gain*	Default
HTTPS Server*	http and https Enabled
Handset/Headset Monitor*	Enabled
Headset Recording*	Disabled
Enbloc Dialing*	Enabled
Switch Port Remote Configuration*	Disabled
PC Port Remote Configuration*	Disabled
Automatic Port Synchronization*	Disabled
SSH Access*	Disabled
LOGIN Access*	Enabled
FIPS Mode*	Disabled
80-bit SRTP*	Disabled

Save Delete Copy Reset Apply Config Add New



Directory Number Configuration

Related Links: [Configure Device \(SEP44ADD9D58C04\)](#) [Go](#)

Save Delete Reset Apply Config Add New

Status

Status: Ready

Directory Number Information

Directory Number* ☐ Urgent Priority
Route Partition
Description
Alerting Name
ASCII Alerting Name
External Call Control Profile

☒ Allow Control of Device from CTI

Associated Devices

[Edit Device](#)

[Edit Line Appearance](#)

Dissociate Devices

Directory Number Settings

Voice Mail Profile (Choose <None> to use system default)
Calling Search Space
BLF Presence Group*
User Hold MOH Audio Source
Network Hold MOH Audio Source

Auto Answer*

☐ Reject Anonymous Calls

Enterprise Alternate Number

[Add Enterprise Alternate Number](#)

+E.164 Alternate Number

[Add +E.164 Alternate Number](#)

Directory URIs

Primary	URI	Partition	Advertise Globally via ILS	Remove
<input checked="" type="radio"/>	<input type="text"/>	<input type="text" value=" < None >"/>	<input checked="" type="checkbox"/>	
Add Row				

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

Advertised Failover Number

AAR Settings

	Voice Mail	AAR Destination Mask	AAR Group
AAR	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value=" < None >"/>
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history			

Call Forward and Call Pickup Settings

	Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			<input type="text" value="Use System Default"/>
Forward All	<input type="checkbox"/> or	<input type="text"/>	<input type="text" value="Phone-CSS"/>



Secondary Calling Search Space for Forward All			< None >
Forward Busy Internal	<input checked="" type="checkbox"/> or		Phone-CSS
Forward Busy External	<input checked="" type="checkbox"/> or		Phone-CSS
Forward No Answer Internal	<input checked="" type="checkbox"/> or		Phone-CSS
Forward No Answer External	<input checked="" type="checkbox"/> or		Phone-CSS
Forward No Coverage Internal	<input type="checkbox"/> or		< None >
Forward No Coverage External	<input type="checkbox"/> or		< None >
Forward on CTI Failure	<input type="checkbox"/> or		< None >
Forward Unregistered Internal	<input checked="" type="checkbox"/> or		Phone-CSS
Forward Unregistered External	<input checked="" type="checkbox"/> or		Phone-CSS
No Answer Ring Duration (seconds)			8
Call Pickup Group			< None >

Park Monitoring			
	Voice Mail	Destination	Calling Search Space
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or		< None > A blank value means to call the parker's line.
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or		< None > A blank value means to call the parker's line.
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	
Target (Destination)	
Target (Destination)	
MLPP Calling Search Space	< None >
MLPP No Answer Ring Duration (seconds)	
Confidential Access Mode	< None >
Confidential Access Level	< None >
Call Control Agent Profile	< None >

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 SEP44ADD9D58C04	
Display (Caller ID)	Niraj Phone 1 Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	Niraj Phone 1
Line Text Label	2154
External Phone Number Mask	416353XXXX
Visual Message Waiting Indicator Policy*	Use System Policy
Audible Message Waiting Indicator Policy*	Default
Ring Setting (Phone Idle)*	Ring
Ring Setting (Phone Active)	Use System Default Applies to this line when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting(Phone Idle)	Use System Default



Recording Option*	Call Recording Disabled
Recording Profile	< None >
Recording Media Source*	Gateway Preferred
Monitoring Calling Search Space	< None >
<input checked="" type="checkbox"/> Log Missed Calls	

Multiple Call/Call Waiting Settings on Device SEP44ADD9D58C04

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

Maximum Number of Calls*

Busy Trigger* (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP44ADD9D58C04

☒ Caller Name

☐ Caller Number

☐ Redirected Number

☒ Dialed Number

Users Associated with Line

	Full Name	User ID	Permission
<input type="checkbox"/>	Nirai phone 1	Niraj phone 1	

[Associate End Users](#) [Select All](#) [Clear All](#) [Delete Selected](#)

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)

VG224 Analog Gateway (FAX application)

Gateway Configuration Related Links: [Back To Find/List](#) [Go](#)

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)

Gateway Details

Product VG224

Gateway BELL_VG224

Protocol MGCP

Device is not trusted

Domain Name*

Description

Cisco Unified Communications Manager Group*

Configured Slots, VICs and Endpoints

Module in Slot 2

Subunit 0 2/ 0 2/ 1 2/ 2 2/ 3 2/ 4 2/ 5

2/ 6 2/ 7 2/ 8 2/ 9 2/10 2/11

2/12 2/13 2/14 2/15 2/16 2/17

2/18 2/19 2/20 2/21 2/22 2/23

Product Specific Configuration Layout

Modem Passthrough*

Cisco Fax Relay*

T38 Fax Relay*

RTP Package Capability*

MT Package Capability*

RES Package Capability*



Product Specific Configuration Layout

?

Modem Passthrough*	Enable
Cisco Fax Relay*	Disable
T38 Fax Relay*	Enable
RTP Package Capability*	Enable
MT Package Capability*	Enable
RES Package Capability*	Enable
PRE Package Capability*	Enable
SST Package Capability*	Enable
RTP Unreachable OnOff*	Enable
RTP Unreachable timeout (ms)*	1000
RTCP Report Interval (secs)*	0
Simple SDP*	Enable

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)

Gateway Configuration

Related Links: [Back to MGCP Configuration](#) [Go](#)

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)

Status

Status: Ready

Directory Number Information

[7795 Line \[1 - 2161 \(no partition\)\]](#)

Device Information

Product	Cisco MGCP FXS Port
Gateway	BELL_VG224
Device Protocol	Analog Access
Device is not trusted	
Registration:	Registered with Cisco Unified Communications Manager 10.220.22.161
IPv4 Address:	10.220.144.14
End-Point Name *	AALN/S2/0@BELL_VG224
Description	AALN/S2/0@BELL_VG224
Device Pool*	BELL FAX SIP DP
Common Device Configuration	< None >
Media Resource Group List	BELL SIP MRGL
Packet Capture Mode*	None
Packet Capture Duration	0
Calling Search Space	Fax-CSS
AAR Calling Search Space	< None >
Location*	Hub_None
AAR Group	< None >
Network Locale	< None >
Use Trusted Relay Point*	Default
Geolocation	< None >

☐ Transmit UTF-8 for Calling Party Name



<input type="checkbox"/> Transmit UTF-8 for Calling Party Name Calling Party Transformation CSS <input type="text" value=" < None >"/> <input checked="" type="checkbox"/> Use Device Pool Calling Party Transformation CSS <input type="checkbox"/> Hot line Device <input type="checkbox"/> Device is trusted
MLPP and Confidential Access Level Information MLPP Domain <input type="text" value=" < None >"/> MLPP Indication Not available on this device MLPP Preemption Not available on this device Confidential Access Mode <input type="text" value=" < None >"/> Confidential Access Level <input type="text" value=" < None >"/>
Port Information (POTS) Port Direction* <input type="text" value=" Bothways"/> Prefix DN <input type="text"/> Num Digits* <input type="text" value=" 0"/> Expected Digits* <input type="text" value=" 0"/> SMDI Port Number(0-4096)* <input type="text" value=" 0"/> <input type="checkbox"/> Unattended Port

Save Delete Reset Apply Config Add New

VG224 Analog Gateway Directory Number Configuration

Directory Number Configuration		Related Links: <input type="text" value="Configure Device (AALN/S2/0@BELL_VG224)"/> Go	
Save Delete Reset Apply Config Add New			
Status Status: Ready			
Directory Number Information Directory Number* <input type="text" value="2161"/> <input type="checkbox"/> Urgent Priority Route Partition <input type="text" value=" < None >"/> Description <input type="text" value="Niraj Phone 3"/> Alerting Name <input type="text" value="Niraj Phone 3"/> ASCII Alerting Name <input type="text" value="Niraj Phone 3"/> External Call Control Profile <input type="text" value=" < None >"/> Associated Devices <input type="text" value="AALN/S2/0@BELL_VG224"/> <input type="button" value="Edit Device"/> <input type="button" value="Edit Line Appearance"/> Dissociate Devices <input type="text"/>			
Directory Number Settings Voice Mail Profile <input type="text" value=" < None >"/> (Choose <None> to use system default) Calling Search Space <input type="text" value=" < None >"/> BLF Presence Group* <input type="text" value="Standard Presence group"/> User Hold MOH Audio Source <input type="text" value=" < None >"/> Network Hold MOH Audio Source <input type="text" value=" < None >"/> <input type="checkbox"/> Reject Anonymous Calls			



Network Hold MOH Audio Source < None >				
<input type="checkbox"/> Reject Anonymous Calls				
Enterprise Alternate Number				
<input type="button" value="Add Enterprise Alternate Number"/>				
+E.164 Alternate Number				
<input type="button" value="Add +E.164 Alternate Number"/>				
Directory URIs				
Primary	URI	Partition	Advertise Globally via ILS	Remove
<input type="radio"/>	<input type="text"/>	< None >	<input checked="" type="checkbox"/>	<input type="button" value="Remove"/>
<input type="button" value="Add Row"/>				
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	AAR Group	
<input type="checkbox"/>	<input type="checkbox"/> or	<input type="text"/>	< None >	
<input checked="" type="checkbox"/> Retain this destination in the call forwarding history				
Call Forward and Call Pickup Settings				
	Voice Mail	Destination	Calling Search Space	
Calling Search Space Activation Policy			Use System Default	
Forward All	<input type="checkbox"/> or	<input type="text"/>	< None >	
Secondary Calling Search Space for Forward All				
Forward Busy Internal	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward Busy External	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward No Answer Internal	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward No Answer External	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward No Coverage Internal	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward No Coverage External	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward on CTI Failure	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward Unregistered Internal	<input type="checkbox"/> or	<input type="text"/>	< None >	
Forward Unregistered External	<input type="checkbox"/> or	<input type="text"/>	< None >	
No Answer Ring Duration (seconds)	<input type="text"/>			
Call Pickup Group	< None >			
Park Monitoring				
	Voice Mail	Destination	Calling Search Space	
Park Monitoring Forward No Retrieve Destination External	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.	
Park Monitoring Forward No Retrieve Destination Internal	<input type="checkbox"/> or	<input type="text"/>	< None > A blank value means to call the parker's line.	
Park Monitoring Reversion Timer	<input type="text"/> A blank value will use value set in Park Monitoring Reversion Timer service parameter			
MLPP Alternate Party And Confidential Access Level Settings				
Target (Destination)	<input type="text"/>			



MLPP Calling Search Space		< None >
MLPP No Answer Ring Duration (seconds)		
Confidential Access Mode		< None >
Confidential Access Level		< None >
Call Control Agent Profile		< None >

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/S2/0@BELL_VG224		
Display (Caller ID)	VG224 PORT 0 - FAX LINE 1	Display text for a line appearance is intended for displaying text such as a name instead of a directory number for calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Caller ID)	VG224 PORT 0 - FAX LINE 1	
External Phone Number Mask	41635321XX	

Multiple Call/Call Waiting Settings on Device AALN/S2/0@BELL_VG224		
Note: The range to select the Max Number of calls is: 1-2		
Maximum Number of Calls*	2	
Busy Trigger*	1	(Less than or equal to Max. Calls)

Forwarded Call Information Display on Device AALN/S2/0@BELL_VG224		
<input checked="" type="checkbox"/> Caller Name		

Forwarded Call Information Display on Device AALN/S2/0@BELL_VG224		
<input checked="" type="checkbox"/> Caller Name		
<input type="checkbox"/> Caller Number		
<input type="checkbox"/> Redirected Number		
<input checked="" type="checkbox"/> Dialed Number		

Users Associated with Line		
Associate End Users		

[Save](#) [Delete](#) [Reset](#) [Apply Config](#) [Add New](#)



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
```



Acronyms

Acronym	Definitions
SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
SCCP	Skinny Client Control Protocol
CUCM	Cisco Unified Communications Manager
CUBE	Cisco Unified Border Element
DNS	Domain Name Resolution
G.711	Voice Codec (Uncompressed)
G729	Voice Codec (Compressed)
MGCP	Media Gateway Control Protocol
PAI	P-Asserted Identity
SCCP	Skinny Client Control Protocol
URI	Uniform Resource Indicator
RTP	Real Time Protocol



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