

HQ SIP Trunk to PSTN

CONTENTS >

« SRST for Site B (</workbook/view/ccie-collaboration/task/srst-for-site-b-MzA0Ng%3D%3D>) | URI IntraCluster & InterCluster Dialing (</workbook/view/ccie-collaboration/task/uri-intracluster-intercluster-dialing-MzA0OA%3D%3D>) »

Last updated: November 13, 2017

Note:

[Video capabilities are only on HQ Phone 2](#)

Tasks

- The VoIP Service provider has enabled a SIP trunk at 11.11.11.4 which is the Backbone.
- Create a route pattern for 88885555, make sure you originate your SIP request from R1's loopback interface with IP address 11.11.11.1
- Make sure 85022XXX is sent to the remote destination as the calling number where 2XXX is the 4 digit internal extensions number.
- The VOIP Service provider has been configured to use only iLBC audio codec. Use iLBC as the preferred audio codec for HQ Phones.
- Ensure R1 Gateway does a pass-through for any codec coming from HQ CUCM.

Configuration: [Click to collapse](#)

Device > Phone (Navigate to the SIP Video Phone)

Cisco Camera*	Enabled	<input checked="" type="checkbox"/>
Video Capabilities*	Enabled	<input checked="" type="checkbox"/>

System > Region > Audio Codec Preference List

Audio Codec Preference List Configuration

Save Delete Copy Add New

Status

Status: Ready

CONTENTS

Audio Codec Preference List Information

Name* Audio-Pref-R1-SIP

Description* Low Loss Codec List

Codecs in List*

- ILBC 16k
- G.711 A-Law 56k
- G.711 U-Law 56k
- G.711 A-Law 64k
- G.711 U-Law 64k
- G.729a 8k
- G.729 8k
- G.729ab 8k
- G.729b 8k
- AMR-WB (7k-24k)
- AMR (5k-13k)
- MP4A-LATM 128k
- AAC-LD (MP4A Generic)
- MP4A-LATM 64k
- MP4A-LATM 56k
- L16 256k
- MP4A-LATM 48k
- G.722 64k
- ISAC 32k
- MP4A-LATM 32k
- G.722.1 32k
- G.722 56k
- G.722.1 24k
- G.722 48k
- MP4A-LATM 24k
- G.728 16k
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- GSM Half Rate 6k
- G.723.1 7k

System > Region > Region Configuration

Region Configuration

Save Delete Reset Apply Config Add New

Region Information
Name* Region-R1-SIP

Region	Audio Codec Preference List	Maximum Audio BR Rate	Maximum Session BR Rate For Video Calls
Region-Inttra	Audio-Pref-R1-SIP	16 kbps (LBC, G.729)	384
Region-R1-SIP	Audio-Pref-R1-SIP	16 kbps (LBC, G.729)	384

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

Regions	Audio Codec Preference List	Maximum Audio BR Rate	Maximum Session BR Rate For Video Calls
Default	Keep Current Setting	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting
Region-Inttra			<input type="radio"/> Use System Default
Region-Inttra			<input type="radio"/> None
Region-Inttra			<input type="radio"/> kbps

Save Delete Reset Apply Config Add New

Device > Device Pool

Device Pool Configuration

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Status

Status: Ready

Device Pool Information

Device Pool: DP-R1-SIP (1 members**)

Device Pool Settings

Device Pool Name *	DP-R1-SIP
Cisco Unified Communications Manager Group *	Default
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Local Route Group	< None >
Intercompany Media Services Enrolled Group	< None >

Roaming Sensitive Settings

Date/Time Group *	CMLocal
Region *	Region-R1-SIP
Media Resource Group List	< None >
Location	< None >
Network Locale	< None >
SRST Reference *	Disable
Connection Monitor Duration ***	
Single Button Barge *	Default

Device > Trunk (Select SIP Trunk)

Trunk Configuration

Save
 Delete
 Reset
 Add New

Status

Status: Ready

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name *	R1-SIP-Trunk
Description	
Device Pool *	DP-R1-SIP
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	0
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	

Run On All Active Unified CM Nodes

SIP Information

Destination

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1 *	11.11.11.1		5060

MTP Preferred Originating Codec *	711ulaw
BLF Presence Group *	Standard Presence group
SIP Trunk Security Profile *	Non Secure SIP Trunk Profile
Rerouting Calling Search Space	< None >
Out-Of-Dialog Refer Calling Search Space	< None >
SUBSCRIBE Calling Search Space	< None >
SIP Profile *	Standard SIP Profile
DTMF Signaling Method *	No Preference

Route Pattern Configuration

Save

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain

Route Class*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

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*

R1 - Gateway

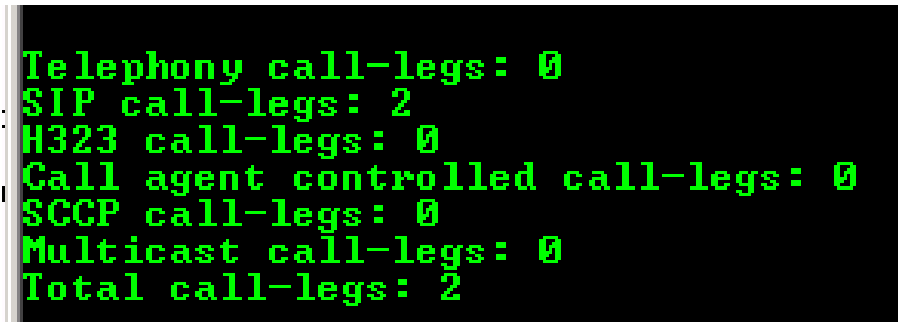
```
voice service voip
  no ip address trusted authenticate
  allow-connections sip to sip
  sip
    bind control source-interface loopback 0
    bind media source-interface loopback 0
  !
dial-peer voice 88885555 voip
  destination-pattern ^88885555$
  session protocol sipv2
  session target ipv4:11.11.11.4
  voice-class sip pass-thru content sdp
  dtmf-relay rtp-nte sip-kpml
!
dial-peer voice 88885556 voip
  session protocol sipv2
  incoming called-number 88885555
  voice-class sip pass-thru content sdp
  dtmf-relay rtp-nte sip-kpml
```

Verification

From HQ Phone 2 place a call to PSTN Phone

Dial = **88885555**

Show call active video brief.



```
Telephony call-legs: 0
SIP call-legs: 2
H323 call-legs: 0
Call agent controlled call-legs: 0
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 2
```

This section breaks the previous section, SRST & MGCP Fallback for HQ Site (<http://labs.ine.com/workbook/view/ccie-collaboration/task/srst-mgcp-fallback-for-hq-site-MzA0NQ%3D%3D>). SRST Mode SIP Phones will not register as you changed the bind control and bind media to loopback 0.

To make SRST and this section work:

1. Go to CUCM and change STST Reference to 11.11.11.1
2. On R1 Gateway Change call-manager-fallback to have source address as 11.11.11.1
« SRST for Site B ([/workbook/view/ccie-collaboration/task/srst-for-site-b-MzA0Ng%3D%3D](http://labs.ine.com/workbook/view/ccie-collaboration/task/srst-for-site-b-MzA0Ng%3D%3D)) | URI IntraCluster & InterCluster Dialing ([/workbook/view/ccie-collaboration/task/uri-intracluster-intercluster-dialing-MzA0OA%3D%3D](http://labs.ine.com/workbook/view/ccie-collaboration/task/uri-intracluster-intercluster-dialing-MzA0OA%3D%3D))

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