

Codec Negotiation

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« CUCM SIP Trunks – Intersite SIP Trunk using CUBE: HQ & Site B (/workbook/view/ccie-collaboration/task/cucm-sip-trunks-intersite-sip-trunk-using-cube-hq-site-b-MzAzMQ%3D%3D) | DialPlan for HQ Site (/workbook/view/ccie-collaboration/task/dialplan-for-hq-site-MzAzMw%3D%3D) »

Last updated: November 13, 2017

Tasks

- Make sure within intrasite the codec used between the phones is g729 as a preference at both Sites – HQ and Site B.
- Also ensure between Intersite (HQ & Site B) the codec used is iLBC as a preference. Make sure R2 for intersite calls does not modify codec.
- After successful configuration MOH should work intra and inter site.

Configuration: [Click to collapse](#)

HQ – Publisher


System > Region Information > Audio Codec Preference List

Name	Description	Copy
Audio-HQ-INTRA-PHONES	Lossy Codec List	
Audio-SIP-Trunk-LIST	Lossy Codec List	
Factory-Default-List	Lossy Codec List	
Factory-Default-Low-Loss	Low Loss Codec List	

Audio Codec Preference List Configuration

 Save  Delete  Copy  Add New

Status

 Status: Ready

Audio Codec Preference List Information

Name*

Description*

Codecs in List*

- G.729 8k
- G.729ab 8k
- G.729b 8k
- ILBC 16k
- 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.711 U-Law 64k
- G.711 A-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 56k
- G.728 16k
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- G.729a 8k
- GSM Half Rate 6k
- G.723.1 7k

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Audio Codec Preference List Configuration

Save Delete Copy Add New

Status

Status: Ready

Audio Codec Preference List Information

Name* Audio-SIP-Trunk-LIST

Description* Low Loss Codec List

Codecs in List*

- ILBC 16k
- 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.711 U-Law 64k
- G.711 A-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 56k
- G.728 16k
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- G.729b 8k
- G.729ab 8k
- G.729 8k
- G.729a 8k
- GSM Half Rate 6k
- G.723.1 7k

SEC

Save Delete Copy Add New

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System > Region Information > Region

Find and List Regions

Status: 1 records found

Regions (1 - 3 of 3)

Name*
Default
RegionIntra-Phones
RegionTrunk

Region Configuration

Region Information: Name: RegionIntra-Phones

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
RegionIntra-Phones	Audio-HQ-INTRA-PHONES	8 kbps (G.729)	384
RegionTrunk	Audio-SIP-Trunk-LIST	16 kbps (ILBC, G.728)	384

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

Region Configuration

Region Information: Name: RegionTrunk

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
RegionIntra-Phones	Audio-SIP-Trunk-LIST	16 kbps (ILBC, G.728)	384
RegionTrunk	Audio-SIP-Trunk-LIST	16 kbps (ILBC, G.728)	384

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

System > Device Pool (Change the Default Device Pool) DP-HQ-PHONES

Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Status
 Status: Ready

Device Pool Information
 Device Pool: DP-HQ-PHONES (2 members)**

Device Pool Settings

Device Pool Name*

Cisco Unified Communications Manager Group*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Intercompany Media Services Enrolled Group

Roaming Sensitive Settings

Date/Time Group*

Region*

Create New Device Pool **DP-R2-CUBE-TRUNK**

Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Status
 Update successful
 Click on the Reset button to have the changes take effect.

Device Pool Information
 Device Pool: DP-R2-CUBE-TRUNK (0 members)**

Device Pool Settings

Device Pool Name*

Cisco Unified Communications Manager Group*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Local Route Group

Intercompany Media Services Enrolled Group

Roaming Sensitive Settings

Date/Time Group*

Region*

Device > Phone
 Assign Phones the Device Pool

Find and List Phones Related Links: [Actively Logged In Device Report](#) Go

Add New Select All Clear All Delete Selected Reset Selected Apply Config to Selected

Status
 1 records found

Phone (1 - 2 of 2) Rows per Page: 50

Find Phone where Device Name begins with Find Clear Filter Select item or enter search text

<input type="checkbox"/>	Device Name(Line)*	Description	Device Pool	Device Protocol	Status	IP Address	Copy	Super Copy
<input type="checkbox"/>	SEP808087223648	HQ Phone 1	DP-HQ-PHONES	SCCP	Registered with 11.100.64.12	11.100.64.43		
<input type="checkbox"/>	SEP808087223649	HQ Phone 2	DP-HQ-PHONES	SIP	Registered with 11.100.64.12	11.100.64.43		

Device > Trunk
 Assign the Device Pool to the Trunk

Trunks (1 - 1 of 1)

Find Trunks where Device Name begins with Find Clear Filter Select item or enter search text

<input type="checkbox"/>	Name ^	Description	Calling Search Space	Device Pool
<input type="checkbox"/>	R2-Cube			DP-R2-CUBE-TRUNK

Service Parameter Configuration

Save Set to Default Advanced

Server* 11.100.64.11 (Active)

Service* Cisco IP Voice Media Streaming App (Active)

All parameters apply only to the current server except parameters that are in the cluster-wide group(s).

Cisco IP Voice Media Streaming App (Active) Parameters on server 11.100.64.11 (Active)

Parameter Name	Parameter Value
Annunciator (ANN) Parameters	
Call Count *	48
Run Flag *	True
Conference Bridge (CFB) Parameters	
Call Count *	48
Run Flag *	True
Media Termination Point (MTP) Parameters	
Call Count *	48
Run Flag *	True
Clusterwide Parameters (Parameters that apply to all servers)	
Supported MOH Codes *	711 mulaw 711 alaw 729 Annex A wideband

Site B – Publisher

System > Region Information > Audio Codec Preference List

Audio Codec Preference Lists

Add New Select All Clear All Delete Selected

Status: 1 records found

Audio Codec Preference Lists (1 - 4 of 4)

Find Audio Codec Preference Lists where Name begins with Find Clear Filter

Name	Description	Copy
Audio-Codec-Phones-List	Low Loss Codec List	0
Audio-Codec-SIP-Trunk-List	Low Loss Codec List	0
Factory-Default-List	Legacy Codec List	0
Factory-Default-List	Low Loss Codec List	0

Add New Select All Clear All Delete Selected

Audio Codec Preference List Configuration

Save Delete Copy Add New

Status: Ready

Audio Codec Preference List Information

Name* Audio-Codec-Phones-List

Description* Low Loss Codec List

Codecs in List*

- G.729 8k
- G.729b 8k
- G.729ab 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.711 U-Law 64k
- G.711 A-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 56k
- ILBC 16k
- G.728 16k
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- G.729a 8k
- GSM Half Rate 6k
- G.723.1 7k

Save Delete Copy Add New

CONTENTS

Audio Codec Preference List Configuration

Save
 Delete
 Copy
 Add New

Status

Status: Ready

Audio Codec Preference List Information

Name*

Description*

Codecs in List*

- ILBC 16k
- G.729 8k
- G.729b 8k
- G.729ab 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.711 U-Law 64k
- G.711 A-Law 64k
- G.711 U-Law 56k
- G.711 A-Law 56k
- G.728 16k
- GSM Enhanced Full Rate 13k
- GSM Full Rate 13k
- G.729a 8k
- GSM Half Rate 6k
- G.723.1 7k

System > Region Information > Region

Find and List Regions

Add New
 Select All
 Clear All
 Delete Selected

Status

3 records found

Regions (1 - 3 of 3)

Find Regions where Name

<input type="checkbox"/>	
<input type="checkbox"/>	Default
<input type="checkbox"/>	Region4Phones-Intrastate
<input type="checkbox"/>	Region4R2CubeTrunk

Region Configuration

Save
 Delete
 Reset
 Apply Config
 Add New

Region Information
Name*

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Region4Phones-Intrastate	Audio-Codec-Phones-List	8 kbps (G.729)	384
Region4R2CubeTrunk	Audio-Codec-SIP-Trunk-List	16 kbps (ILBC, G.729)	384

NOTE: Regions not displayed: 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
Default	<input type="text" value="Keep Current Setting"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting
Region4Phones-Intrastate	<input type="text" value="Use System Default"/>	<input type="text" value="Use System Default"/>	<input type="radio"/> Use System Default
Region4R2CubeTrunk	<input type="text" value="None"/>	<input type="text" value="None"/>	<input type="radio"/> None
	<input type="text" value="kbps"/>	<input type="text" value="kbps"/>	<input type="radio"/> kbps

Region Configuration Related Links: [Back To Find/Up](#)

Save Delete Reset Apply Config Add New

Region Information
Name:

Region	Audio Codec Preference List	Maximum Audio Bit Rate	Maximum Session Bit Rate for Video Calls
Region4Phones-Intrasite	Audio-Codecs-SIP-Trunk-List	16 kbps (ILBC, G.723)	384
Region4R2CubeTrunk	Audio-Codecs-SIP-Trunk-List	16 kbps (ILBC, G.723)	384

NOTE: Regions not displayed 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
Default	Keep Current Setting	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=" kbps"/>
Region4Phones-Intrasite			
Region4R2CubeTrunk			

Save Delete Reset Apply Config Add New

System > Device Pool (Change the Default Device Pool)

Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Device Pool Information
Device Pool: DP-SiteB-Phones (1 members**)

Device Pool Settings

Device Pool Name*

Cisco Unified Communications Manager Group*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Intercompany Media Services Enrolled Group

Roaming Sensitive Settings

Date/Time Group*

Region*

Device Pool Configuration

Save Delete Copy Reset Apply Config Add New

Status
Status: Ready

Device Pool Information
Device Pool: DP-R2-Sip-Trunk (1 members**)

Device Pool Settings

Device Pool Name*

Cisco Unified Communications Manager Group*

Calling Search Space for Auto-registration

Adjunct CSS

Reverted Call Focus Priority

Intercompany Media Services Enrolled Group

Roaming Sensitive Settings

Date/Time Group*

Region*

Device > Phone

Assign Phones the Device Pool

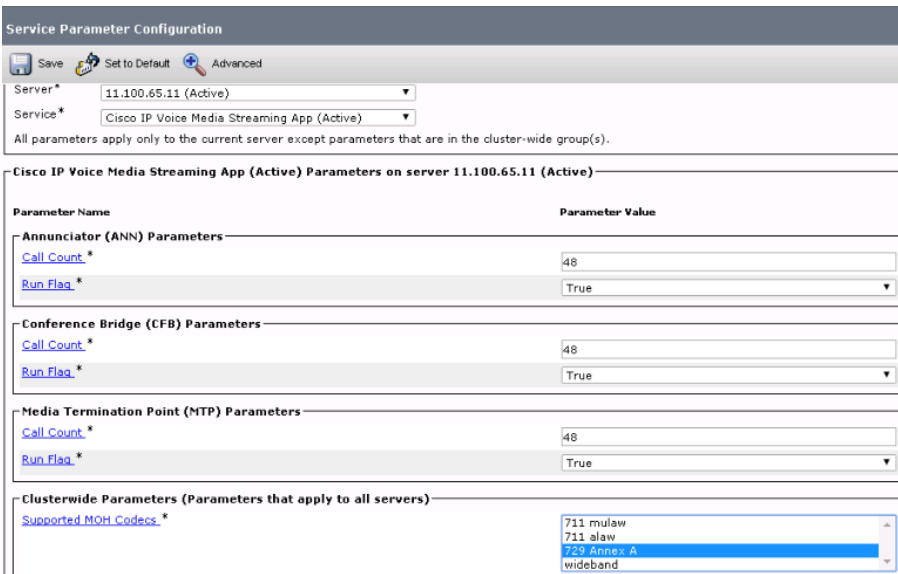
<input type="checkbox"/>	Device Name(Line) ^	Description	Device Pool
<input type="checkbox"/>	SEP64AE0C5F9D4C 7962	Site B Phone 1	DP-SiteB-Phones

Device > Trunk

Assign the Trunk the Device Pool

<input type="checkbox"/>	Name ^	Description	Calling Search Space	Device Pool
<input type="checkbox"/>	R2-Cube			DP-R2-Sip-Trunk

System > Service Parameters



Site B – R2

```

dial-peer voice 2300 voip
  voice-class sip pass-thru content sdp
!
dial-peer voice 2301 voip
  voice-class sip pass-thru content sdp
!
dial-peer voice 3200 voip
  voice-class sip pass-thru content sdp
!
dial-peer voice 3201 voip
  voice-class sip pass-thru content sdp
!
dial-peer voice 3202 voip
  voice-class sip pass-thru content sdp

```

Verification

Call from HQ Phone 1 to HQ Phone 2 and check the codec

Press Hold on either HQ Phone 1 or Phone 2 and check if MOH is being played

Call between HQ Phone 1/HQ Phone 2 to Site B Phone 1 and check the codec

Place a call from HQ Phone 1 to Site B Phone, and vice-versa. Issue the following commands,

Show voip rtp connections on R2 CUBE (Before hold)

```

R2#show voip rtp connections
VoIP RTP Port Usage Information:
Max Ports Available: 8091, Ports Reserved: 101, Ports in Use: 2
Port range not configured, Min: 16384, Max: 32767

Media-Address Range          Ports Available  Ports Reserved  Ports In-use
Default Address-Range        8091            101             2

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP  RmtRTP  LocalIP  RemoteIP
1    23        24        16480   31790   11.102.65.254  11.102.64.49
2    24        23        16476   22304   11.102.65.254  11.102.65.50
Found 2 active RTP connections

```

show voip rtp connections on R2 CUBE (After hold from HQ)

```

R2#show voip rtp connections
VoIP RTP Port Usage Information:
Max Ports Available: 8091, Ports Reserved: 101, Ports in Use: 2
Port range not configured, Min: 16384, Max: 32767

Media-Address Range          Ports Available  Ports Reserved  Ports In-use
Default Address-Range        8091             101              2

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP  RmtRTP  LocalIP                               RemoteIP
1    23        24        16484   4000   11.102.65.254                         11.100.64.12
2    24        23        16476   23488  11.102.65.254                         11.102.65.50
Found 2 active RTP connections

```

› show voip rtp connections on R2 CUBE (After hold from SB)

CONTENTS

```

R2#show voip rtp connections
VoIP RTP Port Usage Information:
Max Ports Available: 8091, Ports Reserved: 101, Ports in Use: 2
Port range not configured, Min: 16384, Max: 32767

Media-Address Range          Ports Available  Ports Reserved  Ports In-use
Default Address-Range        8091             101              2

VoIP RTP active connections :
No. CallId  dstCallId  LocalRTP  RmtRTP  LocalIP                               RemoteIP
1    23        24        16486   17508  11.102.65.254                         11.102.64.49
2    24        23        16490   4000   11.102.65.254                         11.100.65.11
Found 2 active RTP connections

```

show call active voice brief on R2 CUBE (To check Dial-peer matching and voip protocols)

« CUCM SIP Trunks – Intersite SIP Trunk using CUBE: HQ & Site B (/workbook/view/ccie-collaboration/task/cucm-sip-trunks-intersite-sip-trunk-using-cube-hq-site-b-MzAzMQ%3D%3D) | DialPlan for HQ Site (/workbook/view/ccie-collaboration/task/dialplan-for-hq-site-MzAzMw%3D%3D) »