SiteC Intrasite & Intersite Calling with HQ & SiteB

« Call Forward Unregistered (CFUR) (/workbook/view/ccie-collaboration/task/call-forward-unregistered-cfur-MzA2Mw%3D%3D) I CUCME Media Resources - Conference Ad-Hoc (/workbook/view/ccie-collaboration/task/cucme-media-resources-conference-ad-hoc-MzA2NQ%3D%3D) »

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Last updated: November 10, 2017

Note:

Before you begin with this task delete the Site C Phone 1 from HQ CallManager.

Tasks

- Configure R2 & R3 as Cisco Unified Border Element interconnecting HQ and Site B CUCM clusters with Site C using SIP protocol. All 4-digit inter-site calls between Site C and HQ or Site B must go through R2 using Sip. Site C dial-peers should only support iLBC. All intra-site calls should use g711ulaw codec for media and MOH should work within Site C Phones. Video should work between HQ Phone 2 and Site C Phone 2.
- To achieve this task use a separate route pattern 4XXX on HQ Publisher and Site B Publisher respectively.

Configuration: **Click to *collapse***

Site C - R3

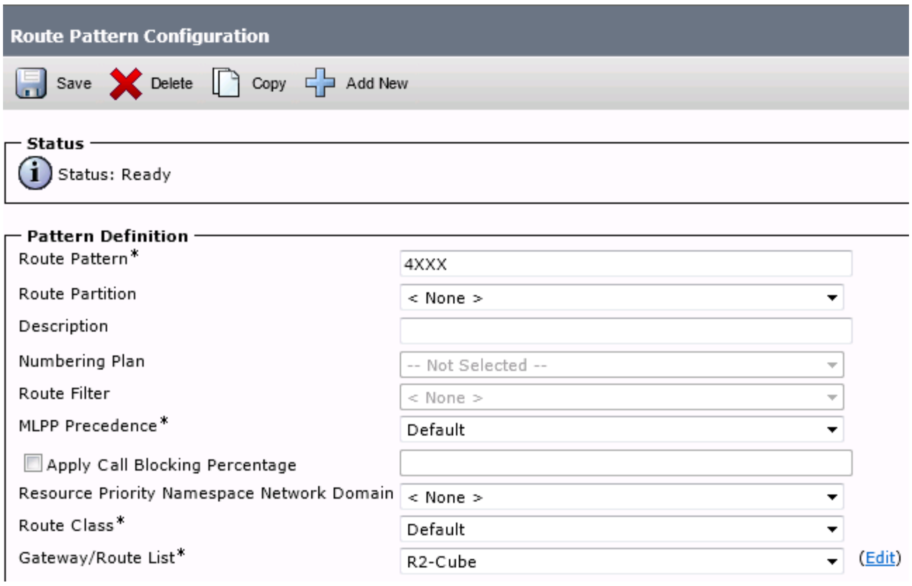
```
voice service voip
 no ip address trusted authenticate
 allow-connections sip to sip
 sip
  asymmetric payload full
 !
dial-peer voice 2301 voip
 destination-pattern ^[23]...$
 session protocol sipv2
 session target ipv4:11.102.65.254
 codec iLBC
 dtmf-relay rtp-nte sip-kpml
 no vad
 !
dial-peer voice 2300 voip
 session protocol sipv2
 incoming called-number ^4...$
 codec iLBC
 dtmf-relay rtp-nte sip-kpml
 no vad
 !
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 ilbc
 !
voice register pool 1
 voice-class codec 1
 reset
 !
voice register global
 create profile
 !
telephony-service
 moh  music\_on\_hold.au
```

Site B - R2

```
voice service voip
 no ip address trusted authenticate
 !
dial-peer voice 4000 voip
 session protocol sipv2
 incoming called-number ^4...$
 codec ilbc
 dtmf-relay rtp-nte sip-kpml
 no vad
 voice-class sip pass-thru content sdp
 !
dial-peer voice 4001 voip
 destination-pattern ^4...$
 session protocol sipv2
 session target ipv4:11.102.66.254
 codec ilbc
 dtmf-relay rtp-nte sip-kpml
 voice-class sip pass-thru content sdp
 no vad
```

HQ - Publisher

Call Routing > Route/Hunt > Route Pattern


Route Pattern Configuration

Save Delete Copy Add New

Status

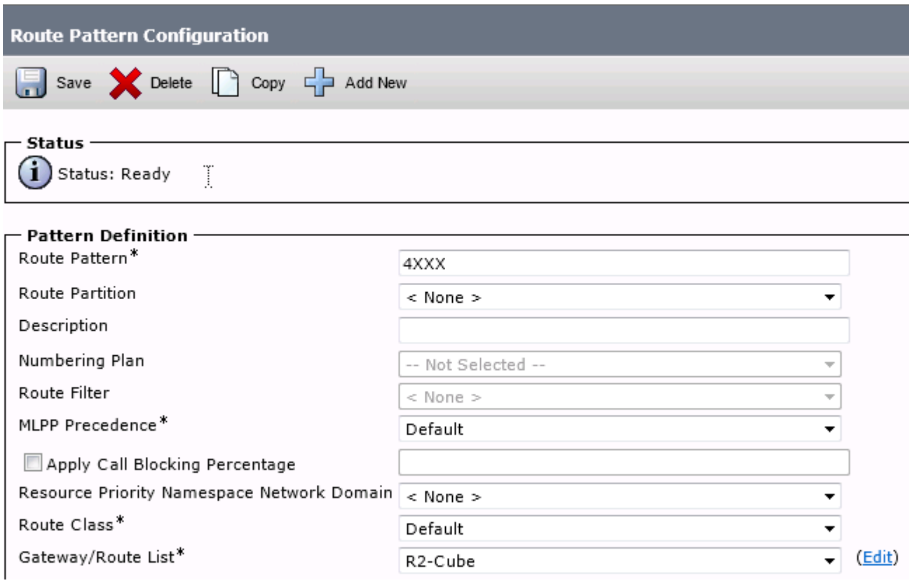
Status: Ready

Pattern Definition

Route Pattern*	4XXX
Route Partition	< None >
Description	
Numbering Plan	-- Not Selected --
Route Filter	< None >
MLPP Precedence*	Default
Apply Call Blocking Percentage	
Resource Priority Namespace Network Domain	< None >
Route Class*	Default
Gateway/Route List*	R2-Cube (Edit)

Site B - Publisher

Call Routing > Route/Hunt > Route Pattern



Verification

Dial between all sites using the 4 digit extensions only

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