

Automated Alternate Routing (AAR)

« Enhanced Location Call Admission Control using RSVP (/workbook/view/ccie-collaboration/task/enhanced-location-call-admission-control-using-rsvp-MzA1NA%3D%3D) | Call Control Discovery over Service Advertisement Framework (CCD/SAF) (/workbook/view/ccie-collaboration/task/call-control-discovery-over-service-advertisement-framework-ccd-saf-MzA1Ng%3D%3D) »

Last updated: November 10, 2017

CONTENTS >

Tasks

- When HQ & Site C Phones call each other and there is no bandwidth available make sure they can still reach other using alternate routing and by dialing 4 digits.
- To test AAR increase the busy trigger on line of Site C Phone 1 to 3.

Configuration: [Click to collapse](#)

Increase the number of Busy Trigger to 3 for Site C Phone 1

HQ – Publisher

Call Routing > AAR Group

The screenshot shows the 'Automated Alternate Routing Group Configuration' page. At the top, there is a title bar with the text 'Automated Alternate Routing Group Configuration'. Below the title bar, there are three buttons: 'Save' (with a floppy disk icon), 'Delete' (with a red 'X' icon), and 'Add New' (with a blue plus icon). The main content area is divided into three sections. The first section is titled 'Status' and contains an information icon (i) and the text 'Status: Ready'. The second section is titled 'Automated Alternate Routing Group Information' and contains a text input field labeled 'Name*' with the value 'AAR-Group-HQ'. The third section is titled 'Prefix Digits within AAR-Group-HQ' and contains a text input field labeled 'Dial Prefix' with the value 'AAR-Group-HQ'. At the bottom of the form, there are three buttons: 'Save', 'Delete', and 'Add New'.

System > Device Pool

Device Pool Configuration

Save
 Delete
 Copy
 Reset
 Apply Config
 Add New

Status

Status: Ready

Device Pool Information

Device Pool: DP-Phones (5 members**)

Device Pool Settings

Device Pool Name*

Device Mobility Related Information****

Device Mobility Calling Search Space
 AAR Calling Search Space
 AAR Group

Device > Phone > Line

Directory Number Configuration

Save
 Delete
 Reset
 Apply Config
 Add New

Status

Status: Ready

Directory Number Information

Directory Number*
 Route Partition
 Description
 Alerting Name
 ASCII Alerting Name
 Allow Control of Device from CTI

AAR Settings	Voice Mail	AAR Destination Mask	AAR Group
AAR <input type="checkbox"/>	<input type="text" value=""/>	<input type="text" value="+85270044001"/>	<input type="text" value="AAR-Group-HQ"/>
<input type="checkbox"/> Retain this destination in the call forwarding history			

Device > 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*
 Description
 Device Pool*

Inbound Calls

Significant Digits*

SIP Information

Destination

Destination Address is an SRV

1*	Destination Address	Destination Address IPv6	Destination Port
	<input type="text" value="11.102.66.254"/>	<input type="text" value=""/>	<input type="text" value="5060"/>

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*

Service Parameter Configuration

Save Set to Default Advanced

Status

Update successful

Select Server and Service

Server* 11.100.64.11 (Active)

Service* Cisco CallManager (Active)

Service Parameter Configuration Related Links: Param

Save Set to Default Advanced

Default Interregion Max Video Call Bit Rate (Includes Audio)* 384 384

On R3 Gateway

```
dial-peer voice 111006412 voip
 destination-pattern ^70044...$
 session protocol sipv2
 session target ipv4:11.100.64.12
 dtmf-relay rtp-nte sip-kpml
!
dial-peer voice 1 pots
 incoming called-number .
 direct-inward-dial
```

Verification

Make a call from HQ Phone 1 to Site C Phone 1. Pickup the call.

Make another call from HQ Phone 2 to Site C Phone 1. Pickup the call.

Place a call from any HQ Phone to Site C Phone 1 and you should see NETWORK CONGESTION REROUTING.

On R1 & R3

```
Debug isdn q931
```

« Enhanced Location Call Admission Control using RSVP (/workbook/view/ccie-collaboration/task/enhanced-location-call-admission-control-using-rsvp-MzA1NA%3D%3D) | Call Control Discovery over Service Advertisement Framework (CCD/SAF) (/workbook/view/ccie-collaboration/task/call-control-discovery-over-service-advertisement-framework-cd-saf-MzA1Ng%3D%3D) »