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## Introduction

The Cisco Unified Border Element (CUBE) provides two types of high availability (HA) options on the Cisco Aggregation Services Router (ASR1000) platform:

1. Box-to-box redundancy
2. Inbox Redundancy

The CUBE HA implementation on the ASR Platforms supports full stateful failover for active SIP-SIP calls using UDP transport. This means both media and session signaling information is preserved after switchover. For active SIP-SIP calls using TCP transport, SIP-H323, H323-H323, we support media preservation after switchover. This capability is supported as of Cisco IOS XE Release 3.2

### Box-to-Box Redundancy

Box-to-box redundancy uses the Redundancy Group (RG) Infrastructure to form an Active/Standby pair of routers. The Active/Standby pair share the same virtual IP address (VIP) and continually exchange status messages. CUBE session information is check-pointed across the Active/Standby pair of routers enabling the Standby router to take over immediately all CUBE call processing responsibilities if the Active router should go out of service for planned or unplanned reasons.

This redundancy option is supported on the ASR 1001/1002/1004 platforms.

### Inbox Redundancy

Inbox redundancy mechanism provides redundancy within the same box. Some models of the ASR offers hardware redundancy within the box and some offers software redundancy. This section discusses the various aspects for Inbox Redundancy on the Cisco ASR1000 platforms.

**Hardware redundancy** – supports stateful failover from an active Enhanced Services Processor to a standby and from an active Route Processor to a standby on the same box. Cisco ASR1006 supports this type of failover

**Software redundancy** – supports stateful failover from an active IOS process to a standby process, both running on the same Route processor. This is different than the platforms running Cisco IOS like the ISR-G2s where only 1 process can run on the operating system. Cisco ASR1001/1002/1004 supports this type of failover.

This application note will provide detailed information on how to set up CUBE on the ASR platform for the Box-to-box redundancy and for Inbox redundancy options.

## 

## Prerequisites

Please review the information in this Prerequisite section.

### Requirements

Ensure that you meet these requirements before you attempt this configuration:

* Basic knowledge of how to configure and use Cisco IOS® voice
* Basic knowledge of how to configure and use CUBE

The basic requirements for setting up CUBE ASR box-to-box redundancy include:

* Two identical ASRs equipped with Cisco release R3.2 image or later
* Both routers must be physically located on the same Ethernet LAN.
* A separate interface should be used for check-pointing control and data traffic across the 2 routers and must be connected via a switch
* The CUBE configuration of both routers is identical and must be manually copied from one router to the other. One router is designated as the Active router and the second as the Standby.

### Components Used

The information in this document is based on a minimum software release of Cisco IOS XE Release 3.2 implemented on a Cisco ASR1001, 1002 or 1004.

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

### Conventions

Refer to the [Cisco Technical Tips Conventions](http%253A%252F%252Fwww.cisco.com%252Fen%252FUS%252Ftech%252Ftk801%252Ftk36%252Ftechnologies_tech_note09186a0080121ac5.shtml) for more information on document conventions.

## Background Information

Box-to-box redundancy requires two identical ASR platforms on the same LAN.

**Redundancy Group (RG) Infra** component will provide the box-to-box communication infrastructure support between the two ASRs and will negotiate the final stable redundancy state. The RG Infra component provides:

* An HSRP-like protocol that negotiates the final redundancy state for each router (via the control interface)
* A transport mechanism for checkpointing the signaling and media state for each call from the ACTIVE to the STANDBY router (via the data interface)
* Configuration/management of the Virtual IP (VIP) interface for the traffic interfaces (multiple traffic interfaces can be configured using the same RG)

This RG component will have to be specifically configured to support voice B2B HA. Please note that only one RG component can be configured on each router for voice B2B HA.

**Virtual IP address management (VIP)** for both signaling and media - B2B HA relies on VIP to achieve redundancy. The VIP and associated physical interfaces on both ASRs in the ASR B2B pair must reside on the same LAN subnet. Configuration of the VIP and binding of the VIP interface to a particular Symphony voice application (SIP, H.323, SWMTP) is mandatory for voice B2B HA support. External devices, such as CUCM, gateway or proxy, will use VIP as the destination IP address for the calls traversing through CUBE(Ent) router.

The signaling and RTP streams of established calls are checkpointed between the Active and Standby routers. In the case of a heartbeat failure when the Active router goes down, the Standby router takes over, and continues to forward the RTP stream that was previously routed by the first router.

Calls in a transient state (i.e. calls that are not established yet, or are in the process of being modified with a transfer or hold function) at the time of failover are disconnected. Also, any calls using DSP services such as transcoding are not preserved.

## Steps to Configure

In this section, you are presented with the information to configure the features described in this document.

CUBE B2B configuration on ASR platforms, follows a specific order of steps, outlined below:

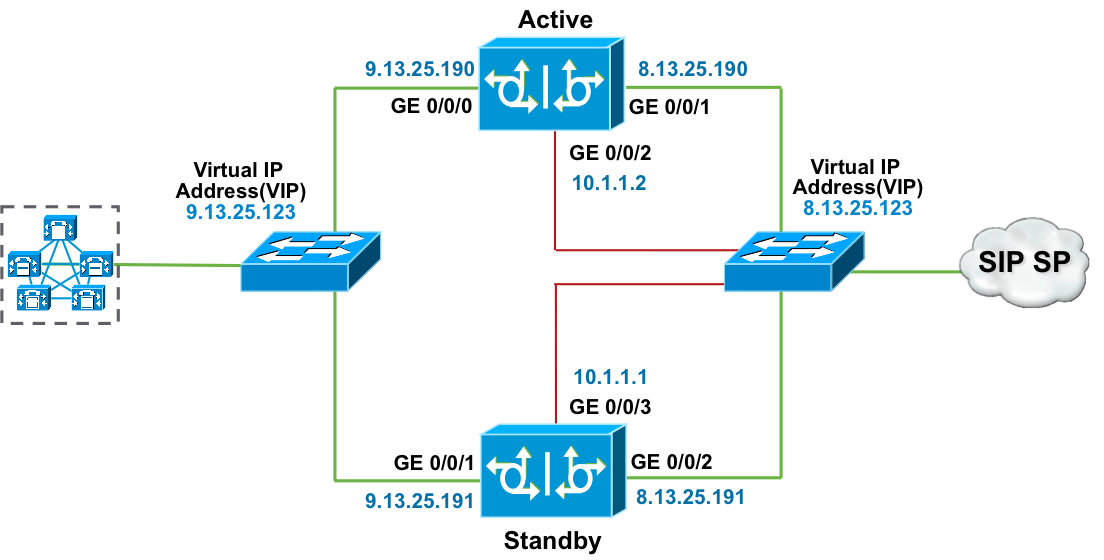
* Step 1: Disable inbox and software redundancy
* Step 2: Configure Redundancy Group (RG)
* Step 3: Configure interfaces
* Step 4: Configure SIP binding (only for SIP calls)
* Step 5: Configure H.323 binding (only for H.323 calls)
* Step 6: Enable B2B Redundancy
* Step 7: Configure Media Inactivity timer
* Step 8: Reload the Routers

Both the ASRs within a B2B HA pair must be manually configured, as B2B infrastructure does not provide configuration-sync to maintain the configuration between the two ASRs used for voice B2B HA.

**Note:**Use the [Command Lookup Tool](http%253A%252F%252Fwww.cisco.com%252Fcgi-bin%252FSupport%252FCmdlookup%252Fhome.pl) ( [registered](http://tools.cisco.com/RPF/register/register.do) customers only) to obtain more information on the commands used in this section.

### Network Diagram

Figure 1 shows the topology of an Active/Standby pair of ASR routers used in a SIP trunk deployment between a Cisco Unified Communications Manager (CUCM) and a service provider (SP) SIP trunk for PSTN access.

****

Note: The Gig0/2 interface used for checkpointing traffic, should be connected via a switch and not directly connected

### Step 1: Disable inbox and software redundancy

1. Change the redundancy mode to “none”

redundancy

mode none

2. Save the running configuration to a text file in bootflash:

Router# copy running-configuration bootflash:<filename>

3. Force the router to go into rommon upon next reload:

Router(config)Config-register 0x0

Router(config)write erase

4. Reload the router

5. At rommon prompt, unset the IOSXE\_Dual\_IOS variable to disable the software redundancy

Rommon1> IOSXE\_DUAL\_IOS=0

Rommon2> sync

6. Boot the ASR image from the bootflash or harddisk: or from the network

7. When the router is up, re-apply the old configuration by copying the configuration file to the running-configuration

Router# copy bootflash:<filename> running-configuration

8. Change the config register back to a non-zero value

Router(config)Config-register 0x2102

### Step 2: Configure Redundancy Group (RG)

Configure an RG group for use with VoIP HA under the “application redundancy” submode

redundancy

mode none

application redundancy

group 1

name voice-b2bha

priority 100 failover threshold 75

timers delay 30 reload 60

control GigabitEthernet0/0/2 protocol 1

data GigabitEthernet0/0/2

protocol 1

timers hellotime 3 holdtime 10

An explanation of the fields used in this configuration is as follows:

* **data GigabitEthernet0/0/2** – Configures the interface used for checkpointing of data traffic
* **control GigabitEthernet0/0/2 protocol 1** – Configures the interface used to exchange keepalive and hello messages between the ASRs pair
* **name voice-b2bha** config is optional
* **timers delay 30 reload 60** – Configures the two timers for delay and reload:
  + Delay timer which is the amount of time to delay RG group’s initialization and role negotiation after the interface comes up – Default 30 seconds. Range is 0-10000 seconds
  + Reload - This is the amount of time to delay RG group initialization and role-negotiation after a reload – Default 60 seconds. Range is 0-10000 seconds
* **timers hellotime 3 holdtime 10** – Configures the two timers for hellotime and holdtime:
  + Hellotime - Interval between successive hello messages – Default 3 seconds. Range is 250 milliseconds-254 seconds
  + Holdtime – The interval between the receipt of a Hello message and the presumption that the sending router has failed. This duration has to be greater than the hello-time – Default 10 seconds. Range is 750 milliseconds-255 seconds

It is recommended to have the holdtime timer configured to be at least 3 times the value of the hellotime timer

### Step 3: Configure interface tracking:

Track CLI is used in RG to track the voice traffic interface state so that the Active router will initiate switchover after the traffic interface is down

Configure the below at global level to track the status of the interface.

track 1 interface GigabitEthernet0/0/0 line-protocol

track 2 interface GigabitEthernet0/0/1 line-protocol

application redundancy

group 1

track 1 shutdown

   track 2 shutdown

### Step 4: Configure the interfaces

Under each physical interface to be used, configure the following CLIs

interface GigabitEthernet0/0/0

ip address 9.13.25.190 255.255.0.0

negotiation auto

**bfd interval 50 min\_rx 50 multiplier 3**

**redundancy rii 1**

**redundancy group 1 ip 9.13.25.123 exclusive**

interface GigabitEthernet0/0/1

ip address 8.13.25.190 255.255.255.0

media-type rj45

negotiation auto

**bfd interval 50 min\_rx 50 multiplier 3**

**redundancy rii 2**

**redundancy group 1 ip 8.13.25.123 exclusive**

interface GigabitEthernet0/0/2

ip address 10.1.1.2 255.255.255.0

media-type rj45

negotiation auto

An explanation of the fields used in this configuration is as follows:

* Configure “redundancy rii” (Redundant Interface Identifier) which configuration is mandatory & used for generating a VMAC)
* The same rii ID value must be used on the interface of each router that has the same VIP
* Configure the RG group employed, as well as the VIP assigned to this physical interface

Note: It is mandatory to use separate interface for redundancy. Ie. Interface used for traffic cannot be used for HA keep-alives and checkpointing. In this example, Gigabit interface 0/0/2 is used for checkpointing.

### Step 4: Configure SIP Binding

Configure CUBE to bind SIP messages to the interface that is configured with a Virtual IP address (VIP) for the RG group employed.

dial-peer voice 1 voip

session protocol sipv2

incoming called-number 2000

**voice-class sip bind control source-interface GigabitEthernet0/0/0**

**voice-class sip bind media source-interface GigabitEthernet0/0/0**

codec g711ulaw

!

dial-peer voice 2 voip

destination-pattern 2000

session protocol sipv2

session target ipv4:9.41.34.11

**voice-class sip bind control source-interface GigabitEthernet0/0/1**

**voice-class sip bind media source-interface GigabitEthernet0/0/1**

codec g711ulaw

### Step 5: Configure H323 binding (only if H323 calls are involved)

Under the interface used by H.323, configure voip-bind with its source address equal to this interface’s VIP for the RG group employed

voice service voip

h323

**call preserve limit-media-detection**

**no h225 timeout keepalive**

interface GigabitEthernet0/0/0

ip address 9.13.25.190 255.255.0.0

media-type rj45

negotiation auto

bfd interval 50 min\_rx 50 multiplier 3

redundancy rii 1

redundancy group 1 ip 9.13.25.123 exclusive

**h323-gateway voip interface**

**h323-gateway voip bind srcaddr 9.13.25.123**

interface GigabitEthernet0/0/1

ip address 8.13.25.190 255.255.255.0

media-type rj45

negotiation auto

bfd interval 50 min\_rx 50 multiplier 3

redundancy rii 2

redundancy group 1 ip 8.13.25.123 exclusive

**h323-gateway voip interface**

**h323-gateway voip bind srcaddr 8.13.25.123**

### Step 6: Enable B2B Redundancy

Configure this RG group under the “voice service voip” . This is to enable voice B2B HA

voice service voip

**redundancy-group 1**

* Adding/removing this command requires a reload for the updated configuration to take effect

### Step 7: Media Inactivity Timer

The Media Inactivity Timer enables the Active/Standby router pair to monitor and disconnect calls if no Real-Time Protocol (RTP) packets are received within a configurable time period.

In case of SIP calls, the switched over calls will be cleared with signaling (as signaling information is preserved for switched calls)

For calls which are TCP-based, H.323, or Software MTP based, will be released by the Media Inactivity timer. This is used to guard against any hung sessions that may have resulted from the failover in the event that a normal call disconnect does not clear the call.

The same duration for the Media Inactivity Timer should be configured on both routers. The default value is 30 seconds for SIP and H323 calls. For SW MTP calls the default value is 1200 seconds. This timer is configured as follows:

ip rtcp report interval 9000

gateway

  media-inactivity-criteria all

  timer receive-rtp 1200

  timer receive-rtcp 5

SIP/H323 call legs will be cleared once RTCP timer expires and SWMTP legs will be cleared after RTP timer expired

In the above example, the RTCP timer value will be 9000x5=45000millisecs=45 secs and RTP timer value will be 1200  secs

### Step 8: Reload the Router

Once all the above configs are completed, save and reload the router

### Step 9: Configure the peer ASR router:

Follow the above steps to configure the Standby ASR router. Make sure the correct IP addresses are used.

### Step 10: Point Attached Devices to the CUBE Virtual IP (VIP) Address

The IP-PBX, SIP proxy or service provider must route the calls to CUBE’s virtual IP address .

SIP/H323 messages to the CUBE’s physical IP addresses are not handled with this HA configuration.

For H323 calls, you should disable the keepalive messages in CUCM configuration.

* Go to System Menu and Choose “Service Parameters”.  At the bottom of the Service Parameters, enable Advanced.
* Set the “Allow TCP KeepAlives for H323” to False.
* After this setting is saved, restart the Call Manager Services.

### Configuration of Software MTP on the CUBE ASR (Optional)

Below is a sample configuration of Software MTP on the CUBE ASR:

Note: ASR platform does not support Hardware MTP

sccp local GigabitEthernet0/0/0

sccp ccm <CUCM\_IP\_Address> identifier 1 version 6.0

sccp

!

sccp ccm group 1

bind interface GigabitEthernet0/0/0

associate ccm 1 priority 1

associate profile 6 register RR4-MTP

!

dspfarm profile 6 mtp

codec g711ulaw

maximum sessions software 100

associate application SCCP

## Removing B2B HA Configurations

To remove a previously entered B2B HA configuration from a CUBE router, follow the steps below in the specific order.

Step 1: Remove the application level HA Redundancy configuration:

Router1(config)# voice service voip

Router(config-voice service voip)# no redundancy-group 1

Step 2: Remove the redundancy application group:

Router1(config)# redundancy

Router1(config-red)# redundancy application

Router1(config-red-app)#group 1

Router1(config-red-app-grp)#shutdown

Router1(config-red-app-grp)#exit

Router1(config-red-app)#no group 1

Router1(config-red-app)#exit

Router1(config-red)#no redundancy application

Step 3: Remove the configurations from each of the interfaces

Router1(config)#interface GigabitEthernet0/0/0

Router1(config-int)# no redundancy group 1 ip 9.13.25.123 exclusive

Router1(config-int)#no redundancy rii 1

Step 4:Save configuration changes to memory and reload

Router(config)#write

Router#reload

## Full Sample Configurations for CUBE Box to Box Redundancy

**Below sample configuration assumes interfaces Gig0/0/0 is used for incoming and Gig0/0/1 is used for outgoing calls and Gig0/0/2 is used for redundancy**

**ACTIVE Router CONFIGS**

########################################################################

Router1#sh run

Building configuration...

Current configuration : 3082 bytes

!

! Last configuration change at 21:33:13 UTC Sun Sep 19 2010

!

version 15.1

service timestamps debug datetime msec

service timestamps log datetime msec

!

hostname b2bred2

!

boot-start-marker

boot system flash bootflash:asr1000rp2-adventerprisek9.BLD\_MCP\_DEV\_LATEST\_201008

24\_091509.bin

boot-end-marker

!

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family

!

address-family ipv6

exit-address-family

!

logging buffered 777777777

no logging console

enable secret 5 $1$kan3$QsGBuVkgGDZgRlg4lSrsW1

!

no aaa new-model

!

!

!

ip source-route

!

!

!

!

!

!

!

!

multilink bundle-name authenticated

!

!

!

voice service voip

media bulk-stats

allow-connections h323 to h323

allow-connections h323 to sip

allow-connections sip to h323

allow-connections sip to sip

redundancy-group 1

h323

emptycapability

call preserve limit-media-detection

no h225 timeout keepalive

h245 passthru tcsnonstd-passthru

sip

early-offer forced

midcall-signaling passthru

!

!

voice iec syslog

!

!

track 1 interface GigabitEthernet0/0/0 line-protocol

track 2 interface GigabitEthernet0/0/1 line-protocol

!

!

!

!

redundancy

mode none

application redundancy

group 1

name voice-b2bha

priority 100 failover threshold 75

timers delay 30 reload 60

control GigabitEthernet0/0/2 protocol 1

data GigabitEthernet0/0/2

track 1 shutdown

track 2 shutdown

protocol 1

timers hellotime 3 holdtime 10

!

!

!

ip ftp username bhks

ip ftp password bhks

!

!

!

!

!

!

!

interface GigabitEthernet0/0/0

ip address 9.13.25.190 255.255.255.0

media-type rj45

negotiation auto

no mop enabled

bfd interval 50 min\_rx 50 multiplier 3

redundancy rii 1

redundancy group 1 ip 9.13.25.123 exclusive

h323-gateway voip interface

h323-gateway voip bind srcaddr 9.13.25.123

!

interface GigabitEthernet0/0/1

ip address 8.13.25.190 255.255.255.0

media-type rj45

negotiation auto

bfd interval 50 min\_rx 50 multiplier 3

redundancy rii 2

redundancy group 1 ip 8.13.25.123 exclusive

h323-gateway voip interface

h323-gateway voip bind srcaddr 8.13.25.123

interface GigabitEthernet0/0/2

ip address 10.1.1.2 255.255.255.0

media-type rj45

negotiation auto

!

interface GigabitEthernet0

vrf forwarding Mgmt-intf

no ip address

negotiation auto

!

!

no ip http server

no ip http secure-server

ip rtcp report interval 9000

ip route 0.0.0.0 0.0.0.0 9.44.0.1

!

logging esm config

dialer-list 1 protocol ip permit

dialer-list 1 protocol ipx permit

!

!

!

control-plane

!

!

!

dial-peer voice 10 voip

destination-pattern 140854.....

session protocol sipv2

session target ipv4:8.13.25.102

voice-class sip bind control source-interface GigabitEthernet0/0/1

voice-class sip bind media source-interface GigabitEthernet0/0/1

codec g711ulaw

no vad

!

dial-peer voice 20 voip

session protocol sipv2

session target ipv4:9.13.25.101

incoming called-number 140854.....

voice-class sip bind control source-interface GigabitEthernet0/0/0

voice-class sip bind media source-interface GigabitEthernet0/0/0

codec g711ulaw

no vad

!

!

gateway

media-inactivity-criteria all

timer receive-rtcp 5

timer receive-rtp 1200

!

!

line con 0

exec-timeout 0 0

stopbits 1

line vty 0 4

no login

!

exception data-corruption buffer truncate

end

**STANDBY ROUTER CONFIGS**

Router2#sh run

Building configuration...

Current configuration : 2606 bytes

!

! Last configuration change at 21:34:07 UTC Sun Sep 19 2010

!

version 15.1

service timestamps debug datetime msec

service timestamps log datetime msec

!

hostname b2bred1

!

boot-start-marker

boot system flash bootflash:asr1000rp2-adventerprisek9.BLD\_MCP\_DEV\_LATEST\_201008

24\_091509.bin

boot-end-marker

!

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family

!

address-family ipv6

exit-address-family

!

logging buffered 777777777

no logging console

!

no aaa new-model

!

!

!

ip source-route

!

!

!

!

!

!

!

!

multilink bundle-name authenticated

!

!

!

voice service voip

media bulk-stats

allow-connections h323 to h323

allow-connections h323 to sip

allow-connections sip to h323

allow-connections sip to sip

redundancy-group 1

h323

emptycapability

call preserve limit-media-detection

no h225 timeout keepalive

h245 passthru tcsnonstd-passthru

sip

early-offer forced

midcall-signaling passthru

!

!

voice iec syslog

!

!

!

track 1 interface GigabitEthernet0/0/0 line-protocol

track 2 interface GigabitEthernet0/0/1 line-protocol

!

!

!

redundancy

mode none

application redundancy

group 1

name voice-b2bha

priority 100 failover threshold 75

timers delay 30 reload 60

control GigabitEthernet0/0/2 protocol 1

data GigabitEthernet0/0/2

track 1 shutdown

track 2 shutdown

protocol 1

timers hellotime 3 holdtime 10

!

!

!

ip ftp username bhks

ip ftp password bhks

!

!

!

!

!

!

!

interface GigabitEthernet0/0/0

ip address 9.13.25.191 255.255.255.0

media-type rj45

negotiation auto

bfd interval 50 min\_rx 50 multiplier 3

redundancy rii 1

redundancy group 1 ip 9.13.25.123 exclusive

h323-gateway voip interface

h323-gateway voip bind srcaddr 9.13.25.123

!

interface GigabitEthernet0/0/1

ip address 8.13.25.191 255.255.255.0

media-type rj45

negotiation auto

bfd interval 50 min\_rx 50 multiplier 3

redundancy rii 2

redundancy group 1 ip 8.13.25.123 exclusive

h323-gateway voip interface

h323-gateway voip bind srcaddr 8.13.25.123

interface GigabitEthernet0/0/2

ip address 10.1.1.1 255.255.255.0

media-type rj45

negotiation auto

!

interface GigabitEthernet0

vrf forwarding Mgmt-intf

no ip address

shutdown

negotiation auto

!

!

no ip http server

no ip http secure-server

ip rtcp report interval 9000

ip route 0.0.0.0 0.0.0.0 9.44.0.1

!

logging esm config

!

!

!

control-plane

!

!

!

dial-peer voice 10 voip

destination-pattern 140854.....

session protocol sipv2

session target ipv4:8.13.25.102

voice-class sip bind control source-interface GigabitEthernet0/0/1

voice-class sip bind media source-interface GigabitEthernet0/0/1

codec g711ulaw

no vad

!

dial-peer voice 20 voip

session protocol sipv2

session target ipv4:9.13.25.101

incoming called-number 140854.....

voice-class sip bind control source-interface GigabitEthernet0/0/0

voice-class sip bind media source-interface GigabitEthernet0/0/0

codec g711ulaw

no vad

!

!

gateway

media-inactivity-criteria all

timer receive-rtcp 5

timer receive-rtp 1200

!

!

line con 0

exec-timeout 0 0

stopbits 1

line vty 0 4

no login

!

exception data-corruption buffer truncate

end

## Feature Use Notes

* It is recommended to use the same hardware for both boxes in the active/standby pair to ensure compatibility before & after failover.
* It is mandatory to use separate interface for redundancy. Ie. Interface used for traffic cannot be used for HA keep-alives and checkpointing.
* After failover, CUBE will continue to send and process received Options ping message
* Only media preservation is supported for H323, TCP based and Software MTP based calls
* Transcoded calls are not preserved.
* Call Admission Control will continue to work after failover. After stateful switchover, no calls will be allowed if CAC limit is reached before the switchover took place
* CDRs are sent to the Radius Server even after a switchover occurs. Thus, close CDRs are sent by the newly ACTIVE router (which is the STANDBY router prior to the switchover) when the call disconnects.
* Only RFC2833 to RFC2833 and voice-inband to voice-inband DTMF works after switchover

## Verify

Use the CLI below to verify the Box-to-box configuration is correct and working.

### Verify Redundancy State on the Active Router

Verify the redundancy state with the “show redundancy application group all” command. This command shows the redundancy inter-device information such as the redundancy inter-device states.

**Router1#show redundancy application group all**

Faults states Group 1 info:

Runtime priority: [100]

RG Faults RG State: Up.

Total # of switchovers due to faults: 0

Total # of down/up state changes due to faults: 2

Group ID:1

Group Name:voice-b2bha

Administrative State: No Shutdown

Aggregate operational state : Up

My Role: ACTIVE

Peer Role: STANDBY

Peer Presence: Yes

Peer Comm: Yes

Peer Progression Started: Yes

RF Domain: btob-one

RF state: ACTIVE

Peer RF state: STANDBY HOT

RG Protocol RG 1

------------------

Role: Active

Negotiation: Enabled

Priority: 100

Protocol state: Active

Ctrl Intf(s) state: Up

Active Peer: Local

Standby Peer: address 10.1.1.1, priority 100, intf Gi0/0/2

Log counters:

role change to active: 1

role change to standby: 0

disable events: rg down state 1, rg shut 0

ctrl intf events: up 1, down 2, admin\_down 1

reload events: local request 0, peer request 0

RG Media Context for RG 1

--------------------------

Ctx State: Active

Protocol ID: 1

Media type: Default

Control Interface: GigabitEthernet0/0/2

Current Hello timer: 3000

Configured Hello timer: 3000, Hold timer: 10000

Peer Hello timer: 3000, Peer Hold timer: 10000

Stats:

Pkts 27719, Bytes 1718578, HA Seq 0, Seq Number 27719, Pkt Loss

0

Authentication not configured

Authentication Failure: 0

Reload Peer: TX 0, RX 0

Resign: TX 0, RX 0

Standby Peer: Present. Hold Timer: 10000

Pkts 27700, Bytes 941800, HA Seq 0, Seq Number 27708, Pkt Loss 0

### Verify Redundancy State on the Standby Router

**Router2#Show redundancy application group all**

Faults states Group 1 info:

Runtime priority: [100]

RG Faults RG State: Up.

Total # of switchovers due to faults: 0

Total # of down/up state changes due to faults: 2

Group ID:1

Group Name:voice-b2bha

Administrative State: No Shutdown

Aggregate operational state : Up

My Role: STANDBY

Peer Role: ACTIVE

Peer Presence: Yes

Peer Comm: Yes

Peer Progression Started: Yes

RF Domain: btob-one

RF state: STANDBY HOT

Peer RF state: ACTIVE

RG Protocol RG 1

------------------

Role: Standby

Negotiation: Enabled

Priority: 100

Protocol state: Standby-hot

Ctrl Intf(s) state: Up

Active Peer: address 10.1.1.2, priority 100, intf Gi0/0/2

Standby Peer: Local

Log counters:

role change to active: 0

role change to standby: 1

disable events: rg down state 1, rg shut 0

ctrl intf events: up 1, down 2, admin\_down 1

reload events: local request 0, peer request 0

RG Media Context for RG 1

--------------------------

Ctx State: Standby

Protocol ID: 1

Media type: Default

Control Interface: GigabitEthernet0/0/2

Current Hello timer: 3000

Configured Hello timer: 3000, Hold timer: 10000

Peer Hello timer: 3000, Peer Hold timer: 10000

Stats:

Pkts 27832, Bytes 1725584, HA Seq 0, Seq Number 27832, Pkt Loss

0

Authentication not configured

Authentication Failure: 0

Reload Peer: TX 0, RX 0

Resign: TX 0, RX 0

Active Peer: Present. Hold Timer: 10000

Pkts 27830, Bytes 946220, HA Seq 0, Seq Number 27843, Pkt Loss 0

### Verify Call State after a Switchover

The “show voice high-availability summary” command is used to verify the following:

* The checkpointing of calls on the Standby router after a switchover
* The media-inactivity count on the Active when the calls are over
* To check for native and nonnative (i.e. preserved) calls when both types of calls are present
* To identify the presence of leaked RTP, HA, SPI sessions

On the Active Router:

**Router1#show voice high-availability summary**

========= HA Message Sizes =======

SCCPAPP Data Size:412

SIPSPI Data Size:4260

H323SPI Data Size:2164

RTSPI Data Size:861

CCAPI Data Size:188

VOIPRTP Data Size:158

HA Data Size:68

Total Data Size:4842

======== Voice HA DB INFO ========

Number of calls in HA DB: 0

Number of calls in HA sync pending DB: 0

Number of current SWMTP calls with HA: 0

-----------------------------

First a few entries in HA DB:

-----------------------------

---------------------------------------

First a few entries in Sync Pending DB:

---------------------------------------

----------------------------

======== Voice HA Process INFO ========

Active process current tick: 92663

Active process number of tick events pending: 0

Active process number of tick events processed: 0

======== Voice HA RF INFO ========

FUNCTIONING RF DOMAIN: 0x2

-----

RF Domain: 0x0

Voice HA Client Name: VOIP RF CLIENT

Voice HA RF Client ID: 1345

Voice HA RF Client SEQ: 128

My current RF state ACTIVE (13)

Peer current RF state DISABLED (1)

Current VOIP HA state [LOCAL / PEER] :

[ACTIVE (13) / UNKNOWN (0)]

-----

RF Domain: 0x2 [RG: 1]

Voice HA Client Name: VOIP RG CLIENT

Voice HA RF Client ID: 4054

Voice HA RF Client SEQ: 418

My current RF state ACTIVE (13)

Peer current RF state STANDBY HOT (8)

Current VOIP HA state [LOCAL / PEER] :

[ACTIVE (13) / STANDBY HOT (8)]

-----

Voice HA Active and Standby are in sync.

System has experienced switchover.

======== Voice HA CF INFO ========

Voice HA CF for RG(1):

local ip = 9.13.25.190; remote ip = 9.13.25.191

local port = 4026; remote port = 4025

CF setup done: TRUE

Role is Active. Client side stats:

Received checkpointing requests: 0

Wrote to sockets: 0

Checkpoint buffer in use: 0

Pending transmit events: 0

======== Voice HA COUNTERS ========

Total number of checkpoint requests sent (Active): 0

Total APP DATA sent on Active: 0

Total CREATE sent on Active: 0

Total MODIFY sent on Active: 0

Total DELETE sent on Active: 0

Total number of checkpoint requested received (Standby): 0

Total APP DATA received on Standby: 0

Total CREATE received on Standby: 0

Total MODIFY received on Standby: 0

Total DELETE received on Standby: 0

Media Inactivity event count: 0

Max Media Up time since Call Create: 0 msecs

Queue Failed for MEDIA EVENT - move entry 2 sync pending db: 0

Queue Failed for CREATE - move entry to sync pending db: 0

Queue Failed for MODIFY - move entry to sync pending db: 0

Queue Failed for DELETE - move entry to sync pending db: 0

No Entry Found when processing Tick Queue Event: 0

Entry Deleted - never checkpointed :0

Added Element to Multi Delete List: 0

Standby received Delete as part of Multi-Delete Message: 0

Active Sent Multi Delete Message to Standby: 0

Standby Callback Invoked by CF: 0

Standby Callback Invoked by CF - Negotiation Message: 0

Standby Callback Invoked by CF - No Msg Header: 0

Standby Callback Invoked by CF - ISSU Xform Fail: 0

Standby Callback Invoked by CF - malloc VOIP Buffer fail: 0

Standby Callback Invoked by CF - enqueue to voip ha fail: 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

Checkpoint overflow: 0

HA DB elememnt pool overrun count: 0

HA DB aux element pool overrun count: 0

HA DB insertion failure count: 0

HA DB deletion failure count: 0

Tick event pool overrun count: 0

Tick event queue overrun count: 0

Checkpoint send failure count - ISSU Transform Failure: 0

Checkpoint send failure count - CF failed: 0

Checkpoint get buffer failure count: 0

Checkpoint Received IPC Flow ON from CF: 0

Checkpoint Received IPC Flow OFF from CF: 0

On the Standby Router:

**Router2#sh voice high-availability summary**

========= HA Message Sizes =======

SCCPAPP Data Size:412

SIPSPI Data Size:4260

H323SPI Data Size:2164

RTSPI Data Size:861

CCAPI Data Size:188

VOIPRTP Data Size:158

HA Data Size:68

Total Data Size:4842

======== Voice HA DB INFO ========

Number of calls in HA DB: 0

Number of calls in HA sync pending DB: 0

Number of current SWMTP calls with HA: 0

-----------------------------

First a few entries in HA DB:

-----------------------------

---------------------------------------

First a few entries in Sync Pending DB:

---------------------------------------

----------------------------

======== Voice HA Process INFO ========

Active process current tick: 46846

Active process number of tick events pending: 0

Active process number of tick events processed: 0

======== Voice HA RF INFO ========

FUNCTIONING RF DOMAIN: 0x2

-----

RF Domain: 0x0

Voice HA Client Name: VOIP RF CLIENT

Voice HA RF Client ID: 1345

Voice HA RF Client SEQ: 128

My current RF state ACTIVE (13)

Peer current RF state DISABLED (1)

Current VOIP HA state [LOCAL / PEER] :

[ACTIVE (13) / UNKNOWN (0)]

-----

RF Domain: 0x2 [RG: 1]

Voice HA Client Name: VOIP RG CLIENT

Voice HA RF Client ID: 4054

Voice HA RF Client SEQ: 418

My current RF state STANDBY HOT (8)

Peer current RF state ACTIVE (13)

Current VOIP HA state [LOCAL / PEER] :

[STANDBY HOT (8) / ACTIVE (13)]

-----

Voice HA Standby is not available.

System has not experienced switchover.

======== Voice HA CF INFO ========

Voice HA CF for RG(1):

local ip = 9.13.25.191; remote ip = 9.13.25.190

local port = 4025; remote port = 4026

CF setup done: TRUE

Role is Standby. Server side stats:

Received raw message: 0

Received checkpointing requests: 0

Invalid header counter: 0

======== Voice HA COUNTERS ========

Total number of checkpoint requests sent (Active): 0

Total APP DATA sent on Active: 0

Total CREATE sent on Active: 0

Total MODIFY sent on Active: 0

Total DELETE sent on Active: 0

Total number of checkpoint requested received (Standby): 0

Total APP DATA received on Standby: 0

Total CREATE received on Standby: 0

Total MODIFY received on Standby: 0

Total DELETE received on Standby: 0

Media Inactivity event count: 0

Max Media Up time since Call Create: 0 msecs

Queue Failed for MEDIA EVENT - move entry 2 sync pending db: 0

Queue Failed for CREATE - move entry to sync pending db: 0

Queue Failed for MODIFY - move entry to sync pending db: 0

Queue Failed for DELETE - move entry to sync pending db: 0

No Entry Found when processing Tick Queue Event: 0

Entry Deleted - never checkpointed :0

Added Element to Multi Delete List: 0

Standby received Delete as part of Multi-Delete Message: 0

Active Sent Multi Delete Message to Standby: 0

Standby Callback Invoked by CF: 0

Standby Callback Invoked by CF - Negotiation Message: 0

Standby Callback Invoked by CF - No Msg Header: 0

Standby Callback Invoked by CF - ISSU Xform Fail: 0

Standby Callback Invoked by CF - malloc VOIP Buffer fail: 0

Standby Callback Invoked by CF - enqueue to voip ha fail: 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0 0

Checkpoint overflow: 0

HA DB elememnt pool overrun count: 0

HA DB aux element pool overrun count: 0

HA DB insertion failure count: 0

HA DB deletion failure count: 0

Tick event pool overrun count: 0

Tick event queue overrun count: 0

Checkpoint send failure count - ISSU Transform Failure: 0

Checkpoint send failure count - CF failed: 0

Checkpoint get buffer failure count: 0

Checkpoint Received IPC Flow ON from CF: 0

Checkpoint Received IPC Flow OFF from CF: 0

### Verify SIP IP Address Bindings

The “show sip-ua status” command displays SIP binding status.

Router1#show sip-ua status

SIP User Agent Status

SIP User Agent for UDP : ENABLED

SIP User Agent for TCP : ENABLED

SIP User Agent for TLS over TCP : ENABLED

SIP User Agent bind status(signaling): DISABLED

SIP User Agent bind status(media): DISABLED

Snapshot of SIP listen sockets : 2

Local Address Listen Port Secure Listen Port

============================= =========== ==================

10.10.25.14 5060 5061

10.10.24.14 5060 5061

SIP early-media for 180 responses with SDP: ENABLED

SIP max-forwards : 70

### Verify Current CPU Use

The “show process cpu history” command is used to verify the CPU utilization percentage at regular intervals.

Check CPU utilization before performing a switchover and proceed with a forced failover only when the CPU utilization is less than 70%. “show process cpu sorted” can also be issued repeatedly to get an idea of the CPU utilization for a particular process.

### Forcing a Manual Failover for Testing

Box-to-box redundancy on the ASR platform supports full stateful switchover of calls. This means the media (RTP) and signaling information of the calls is preserved.

Switchovers occurring in real environments where there is a constant mixture of calls in transient (call setup or being modified) and established state, there will always be a certain number of calls dropped during a failover.

Follow the procedure below to force a manual switchover to check that the configuration and operation is correct.

To ensure smooth forced switchover, do the following:

* Monitor the CPU utilization % on the Active/Standby pair. The Active will be having higher CPU utilization as it is actively handling the calls, while the Standby will show little CPU utilization.
* Ensure a manual switchover is performed when the CPU utilization of the Active router is no more than 70%.
* Use the “show voip rtp connection” commands to make sure existing calls have been synced across the Active/Standby router pair.

A switchover involves the formerly Active router reloading, while the formerly Standby router takes over and becomes the new Active router, processing new calls and maintaining the media streams and signaling information for calls until they are complete. The new Active router will continue to act as the Active router until another switchover occurs. There is no pre-emption mechanisms on the B2B Redundancy

Manual (forced) switchovers can be achieved in any one of the following ways:

* Initiate it by the CLI “redundancy application reload group <RG ID> self” on the Active router
* Reload of the Active router
* Power cycle the Active router
* Pull out any RG configured interface of the Active router
* Shutdown any RG configured interface of the Active router

## Troubleshoot

This section provides information you can use to troubleshoot your configuration.

**Note:**Refer to [Important Information on Debug Commands](http%253A%252F%252Fwww.cisco.com%252Fen%252FUS%252Ftech%252Ftk801%252Ftk379%252Ftechnologies_tech_note09186a008017874c.shtml) before you use **debug** commands.

The following is a summary list of the show commands useful during troubleshooting of B2B HA:

1. **show redundancy application group all**
2. **show redundancy application transport clients**
3. **show redundancy client domain all | inc VOIP RG**
4. **show voice high-availability summary**
5. **show voip fpi stats**

The following is a summary list of the debugs useful during troubleshooting of B2B HA:

1. **Debug voip rtp session**
2. **Debug voice high-availability all**
3. **Debug voip fpi all**
4. **debug redundancy application group <config | faults | media | protocol | rii | transport | vp>**

Note: On every switchover after reload the debugs need to be enabled on the new STBY.

**Note:** Do not turn on a large number of debugs on a system carrying a high volume of active call traffic.

### 

### Troubleshooting tips

1. Check for proper HA states on both the active & standby in the output of the show commands, like “show redundancy application group”.
2. Perform incoming & outgoing ping tests with the VIPs employed.
3. In the presence of active calls, look for any use of any physical interface’s IP address in the output of “show voip rtp connections” on both the active & standby. VIP should be used in the show outputs and in the debugs as well
4. In the output of “show voip rtp connection | inc Found” and “show call active voice compact | inc Total” on both the active & standby, check for any large number of mismatched calls.
5. To debug problems, enable the corresponding debug options:
   * + VoIP RTP
     + VoIP FPI
     + VoIP HA
     + SPIs (SIP, H.323, SCCPAPP, etc)

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