

H.323 Gateway Dial–Peer Configuration for Cisco CallManager Server Redundancy

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Introduction

Architecture for Voice, Video, and Integrated Data (AVVID) designs generally recommend that you have a primary and secondary Cisco CallManager server for redundancy. If the primary Cisco CallManager server fails, the secondary Cisco CallManager server takes over and the IP phones re–home to this server.

This scenario works for IP–phone–to–IP–phone communications. However, if an H.323 gateway directs calls to the failed (out–of–service) primary Cisco CallManager server, it needs a way to redirect the call to the secondary Cisco CallManager server.

This procedure works on Cisco AS5300, 36xx, 26xx, MC3810, and 1750 routers equipped with voice cards and configured for Voice over IP (VoIP). This can be done with the help of two VoIP dial peers on the H.323 gateway, that configures one for each Cisco CallManager server.

Prerequisites

Requirements

There are no specific requirements for this document.

Components Used

The information in this document is based on these software and hardware versions:

- Cisco IOS® Software Release 12.1.2T, with IP Plus feature set
- Cisco 2610 gateway
- Cisco CallManager 3.x and 4.0

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 for more information on document conventions.

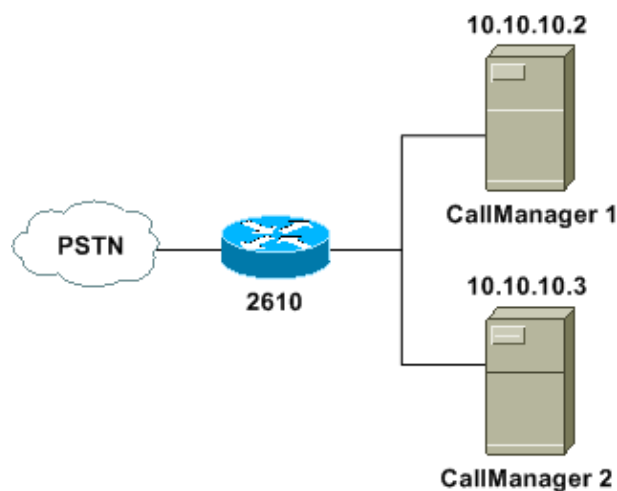
Configure

This section presents you with the information you can use to configure the features this document describes.

Note: Refer to the Command Lookup Tool [↗](#) (registered customers only) in order to find additional information on the commands used in this document.

Network Diagram

This document uses this network setup.



Configurations

This document uses this configuration.

- Cisco 2610

Cisco 2610
Current configuration: version 12.1 voice class h323 1 h225 timeout tcp establish 3 <i>!--- Set the timeout to three seconds.</i> dial-peer voice 100 voip preference 1 <i>!--- Make this the first choice dial peer.</i> ip precedence 5 destination-pattern 1...

```

voice-class h323 1

!--- Apply the voice class to the dial peer.

session target ipv4:10.10.10.2

!--- This is the address of the primary Cisco CallManager.

dtmf-relay h245-alpha

dial-peer voice 101 voip
  preference 2

!--- This is the second choice.

ip precedence 5
destination-pattern 1...
session target ipv4:10.10.10.3

!--- This is the address of the secondary Cisco CallManager.

dtmf-relay h245-alpha

end

```

In some situations, **publisher** and **subscriber** are used as the **secondary** and **primary** servers respectively. In this case the lower preference should be given to the subscriber because it is the Cisco CallManager server designated for call handling, while the publisher is designated to handle both the SQL database and the LDAP directory. If there is any problem with the subscriber, the call should be routed to the publisher, which can also route calls.

Note: When you set the preference order, the lower the preference number, the higher the priority. **The highest priority is given to the dial peer with preference order 0 and it is the default value.** You can have a preference value between **0 to 10**.

Further configuration procedures are:

1. Set up the dial peer that points to the primary Cisco CallManager server with a preference of **1**.

Note: In this case, the **preference** command takes effect because the destination patterns under the two dial peers are the same (otherwise "longest match" overrides it).

Primary Cisco CallManager Server
<pre> dial-peer voice 100 voip preference 1 !--- Make this the first choice dial peer. ip precedence 5 destination-pattern 1... voice-class h323 1 !--- Apply the voice class to the dial peer. session target ipv4:10.10.10.2 !--- This is the address of the primary Cisco CallManager. dtmf-relay h245-alpha </pre>

2. Set up the dial peer that points to the secondary Cisco CallManager server with a preference of 2.

```
Secondary Cisco CallManager Server
dial-peer voice 101 voip
  preference 2

!--- This is the second choice.

  ip precedence 5
  destination-pattern 1...
  session target ipv4:10.10.10.3

!--- This is the address of the secondary Cisco CallManager.

  dtmf-relay h245-alpha
```

In order to use the secondary Cisco CallManager server as **preference 1**, ensure the **Run H.225D On Every Node** flag is checked in the Cisco CallManager configuration for this gateway. If this is unchecked, it causes ten to fifteen seconds of delay in the call setup. If this condition exists, you must check this value and reset the related gateway.

3. Decrease the H.225 timer on the H.323 gateway to three seconds.

This step is necessary because by default, the H.225 timer that controls redirection to a less preferred dial peer on a "no-response" failure is longer than the ten second timer of the Q.931 Call Proceeding timer. When a call comes in to an H.323 gateway through an ISDN trunk and gets forwarded to an inoperative Cisco CallManager, the router waits for forty seconds before it attempts to use a dial peer with a lower preference, or before it clears the call. By the time this occurs, the ISDN Q.931 that signals on the H.323 gateway has already sent an ISDN Q.931 CALL DISCONNECT to the ISDN switch. The router provides an ISDN clearing code of 0x8066 "recovery on timer expiry."

It is not possible for the H.323 gateway to reset the Call Proceeding timer because it attempts to use a different dial-peer. Therefore, the H.323 gateway must switch peers and complete the call with the help of the secondary Cisco CallManager server within the ten seconds allowed by the Q.931 (Incoming Call Proceeding) timer.

When the H.225 timer is set to three seconds, the router attempts a connection to the primary Cisco CallManager server. If it does not receive a response in three seconds, it falls back to the secondary Cisco CallManager server.

Issue the **voice class h323 1** command in order to set the H.225 timer to three seconds.

```
Set the H.225 Timer to Three Seconds
voice class h323 1
  h225 timeout tcp establish 3

!--- Set the timeout to three seconds.
```

4. Issue the **voice class h323 1** command in order to apply the voice class to the dial peers.

```
Apply the Voice Class
dial-peer voice 100 voip
  voice-class h323 1

!--- Apply the voice class to the dial peer.
```

Verify

Issue the **show dial-peer voice summary** command in order to confirm the dial peers are configured and are in the up state, as this output shows.

Use the show dial-peer voice summary Command									
2600GW# show dial-peer voice summary									
dial-peer hunt 0									
							PASS		
TAG	TYPE	ADMIN	OPER	PREFIX	DEST-PATTERN	PREF	THRU	SESS-TARGET	PORT
100	voip	up	up		1...	1	syst	ipv4:10.10.10.2	
101	voip	up	up		1...	2	syst	ipv4:10.10.10.3	
200	pots	up	up		0				1/0/0
201	pots	up	up		0				1/0/1
2600GW#									

- This **show** command indicates that VoIP dial peers 100 and 101 are in an Administrative and Operational state of up.
- Dial peer 100 has a preference of 1 and points to Cisco CallManager 10.10.10.2.
- Dial peer 101 has a preference of 2 and points to Cisco CallManager 10.10.10.3.

Note: If the primary Cisco CallManager server does not accept the call within the three second timeout period, the initial call setup rolls to the secondary Cisco CallManager. The ISDN Q.931 Call Progress timer does not have a chance to expire with this shorter rollover period.

Certain **show** commands are supported by the Output Interpreter Tool [🔗](#) (registered customers only), which allows you to view an analysis of **show** command output.

Troubleshoot

There is currently no specific troubleshooting information available for this configuration.

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Voice & Video: IP Phone Services for End Users
Voice & Video: Unified Communications
Voice & Video: IP Phone Services for Developers
Voice & Video: General

Related Information

- **Voice Technology Support**
 - **Voice and Unified Communications Product Support**
 - **Recommended Reading: Troubleshooting Cisco IP Telephony**
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-

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