CHAPTER 8

# Call Admission Control

Call admission control enables you to control the audio quality and video quality of calls over a wide-area (IP WAN) link by limiting the number of calls that are allowed on that link at the same time. For example, you can use call admission control to regulate the voice quality on a 56-kbps frame relay line that connects your main campus and a remote site.

Audio and video quality can begin to degrade when too many active calls exist on a link and the amount of bandwidth is oversubscribed. Call admission control regulates audio and video quality by limiting the number of calls that can be active on a particular link at the same time. Call admission control does not guarantee a particular level of audio or video quality on the link, but it does allow you to regulate the amount of bandwidth that active calls on the link consume.

This section describes two types of call admission control that you can use with Cisco CallManager:

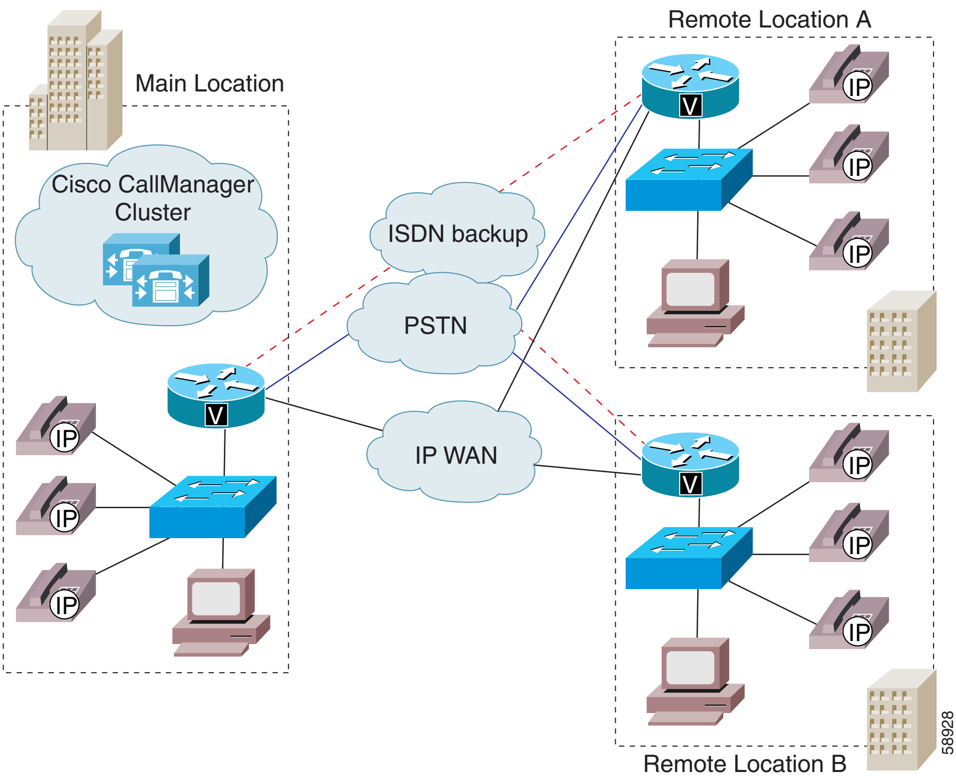
- Locations, page 8-2, for systems with centralized call processing
- Gatekeepers and Trunks, page 8-8, for systems with distributed call processing

You can choose either of these two methods of call admission control, but you cannot combine them in the same Cisco CallManager system. If your system does not contain IP WAN links with limited available bandwidth, you do not have to use call admission control.

# Locations

The locations feature, available in Cisco CallManager, provides call admission control for centralized call-processing systems. A centralized system uses a single Cisco CallManager cluster to control all the locations. Figure 8-1 illustrates call admission control that is using locations. For more information, refer to the "Location Configuration" section in the *Cisco CallManager Administration Guide* and to the *Cisco IP Telephony Solution Reference Network Design Guide*.

***Figure 8-1 Call Admission Control Using Locations in a Centralized System***




In a centralized call-processing system, as illustrated in Figure 8-1, the Cisco CallManager cluster resides at the main location, along with other devices such as phones and gateways. The remote locations (for example, branch offices of your company) house additional phones and other devices, but they do not contain any call-processing capability. The remote locations connect to the main location and to each other by means of IP WAN links (and possibly PSTN and ISDN links as backups).

Calls between devices at the same location do not need call admission control because those devices reside on the same LAN, which has unlimited available bandwidth. However, calls between devices at different locations must travel over an IP WAN link, which has limited available bandwidth.

The locations feature in Cisco CallManager lets you specify the maximum amount of audio bandwidth (for audio calls) and video bandwidth (for video calls) that is available for calls to and from each location, which thereby limits the number of active calls and limits oversubscription of the bandwidth on the IP WAN links.

**Note** Each audio call has two streams, one in each direction. Video calls have four or six streams (that is, two or three streams in each direction).

For example, assume that you have configured the following locations in Cisco CallManager Administration:

| Location | Bandwidth (kbps) |
|---|---|
| San Francisco (main location) | Unlimited |
| Austin (remote location) | 160 |
| Dallas (remote location) | 200 |

Cisco CallManager continues to admit new calls to a link as long as sufficient bandwidth is still available. Thus, if the link to the Austin location in our example has 160 kbps of available bandwidth, that link can support one G.711 call at 80 kbps (in each direction), three G.723 or G.729 calls at 24 kbps each (in each direction), or two GSM calls at 29 kbps each (in each direction). If any additional calls try to exceed the bandwidth limit, the system rejects them, the calling party receives reorder tone, and a text message displays on the phone.

When you configure a location in Cisco CallManager Administration, you assign it a name and maximum audio bandwidth. If you set the audio or video bandwidth to *Unlimited*, you allocate unlimited available bandwidth and allow an unlimited number of active calls on the IP WAN link for that location. In configuring a location, you also assign a video bandwidth for the location. If you set the video bandwidth setting to *None*, no video calls can connect between this location and other locations, but they can take place within this location.

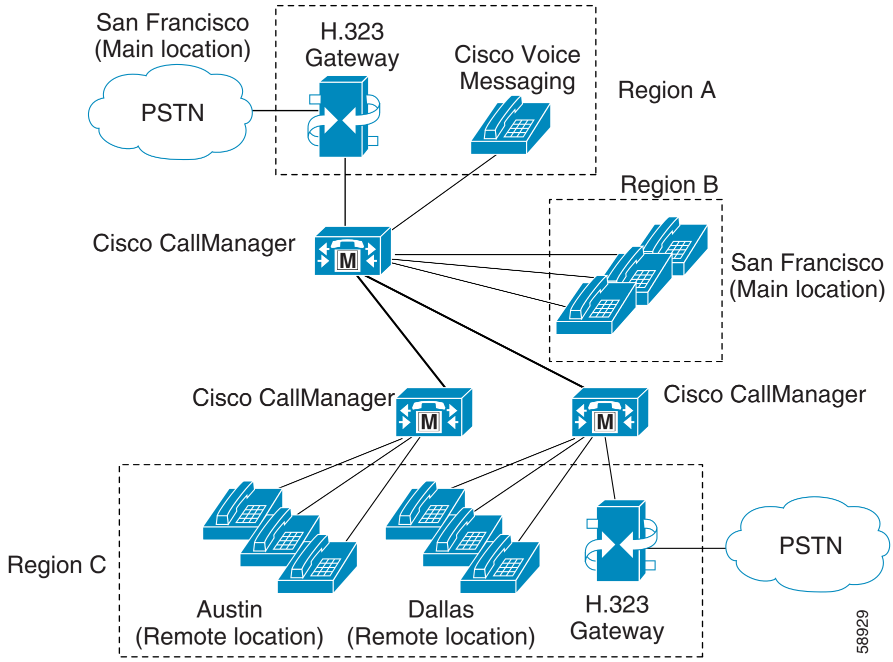
When you configure a phone or other device in Cisco CallManager Administration, you can assign it to a location. If you set the location to *None*, you assign that device to an unnamed location with unlimited available bandwidth and allow an unlimited number of active calls to and from that device.

Location reservations move to reflect the type of call. When a call changes from video to audio-only, the location reservation moves from the video location to an audio location. Calls that change from audio-only to video cause the opposite change of location reservation.

# Locations and Regions

Locations work in conjunction with regions to define the characteristics of a network link. Regions define the type of compression (G.711, G.723, or G.729) that is used on the link, and locations define the amount of available bandwidth for the link. You assign each device in the system to both a region (by means of a device pool) and a location. As illustrated in Figure 8-2, the regions and locations can overlap and intersect in various ways, depending on how you define them. For more information, see the "Regions" section on page 5-5.

*Figure 8-2 Interaction Between Locations and Regions*




# Bandwidth Calculations

In performing location bandwidth calculations for purposes of call admission control, Cisco CallManager assumes that each call stream consumes the following amount of bandwidth:

- G.711 call uses 80 kbps
- G.722 call uses 80 kbps
- G.723 call uses 24 kbps
- G.728 call uses 16 kbps
- G.729 call uses 24 kbps
- GSM call uses 29 kbps
- Wideband call uses 272 kbps

**Note** Each audio call comprises two call streams. Actual bandwidth consumption per call varies, depending on factors such as data packet size. Cisco CallManager uses these fixed values to simplify the bandwidth calculations for purposes of the locations feature only.

Each video call can comprise four or six call streams. For a video call, total bandwidth is the sum of the call audio bandwidth plus video bandwidth but does not include the call overhead.

The audio bandwidth value specified for a location includes overhead, whereas the video bandwidth value specified for a location does not include overhead. For a location, the bandwidth that is available for video calls represents the sum of the audio bandwidth and the video bandwidth. Refer to the "Understanding Video Telephony" chapter for more details.

Cisco CallManager allows calls to complete over a link until sufficient bandwidth does not exist for a new call. At that point, any additional calls fail, and the calling party receives reorder tone.

When a link to a location experiences blockage, it might result from bandwidth leakage that has reduced the usable bandwidth for the location. You can resynchronize the bandwidth allotment to the maximum setting for the location without restarting the Cisco CallManager server. For instructions, refer to "Resynchronizing a Location Bandwidth" in the *Cisco CallManager Administration Guide.*

**Note** If you resynchronize the bandwidth for a location when calls are using the link, the bandwidth might be oversubscribed until all calls that are using the link disconnect. An oversubscribed link can cause audio and video quality to degrade. For this reason, resynchronize the location bandwidth during hours when the link has low traffic.

Media Termination Point (MTP) and transcoder represent exceptions to the bandwidth rules that are outlined in the preceding paragraph. Calls made through an MTP can complete even if they exceed the available bandwidth limit. Calls made through an MTP, however, cannot provide video.

**Caution** In the United States and Canada, routing an emergency 911 call to a link that has no more available bandwidth can block the 911 call. For each location on your network, always route 911 calls to the local public switched telephone network (PSTN) through a local VoIP gateway.

# Locations Configuration Checklist

Table 8-1 lists the general steps for configuring call admission control on the basis of locations.

***Table 8-1 Locations Configuration Checklist***

| Configuration Steps | | Procedures and Related Topics |
|---|---|---|
| **Step 1** | Configure a region for each type of codec that is used in your system. | Locations and Regions, page 8-4<br><br>Region Configuration, *Cisco CallManager Administration Guide*. |
| **Step 2** | Configure a separate location for each IP WAN link to which you want to apply call admission control. Allocate the maximum available bandwidth for calls across the link to that location.<br><br>**Note** If you set the bandwidth to *Unlimited*, you allocate unlimited available bandwidth and allow an unlimited number of active calls on the IP WAN link for that location. | Location Configuration, *Cisco CallManager Administration Guide* |

*Table 8-1 Locations Configuration Checklist (continued)*

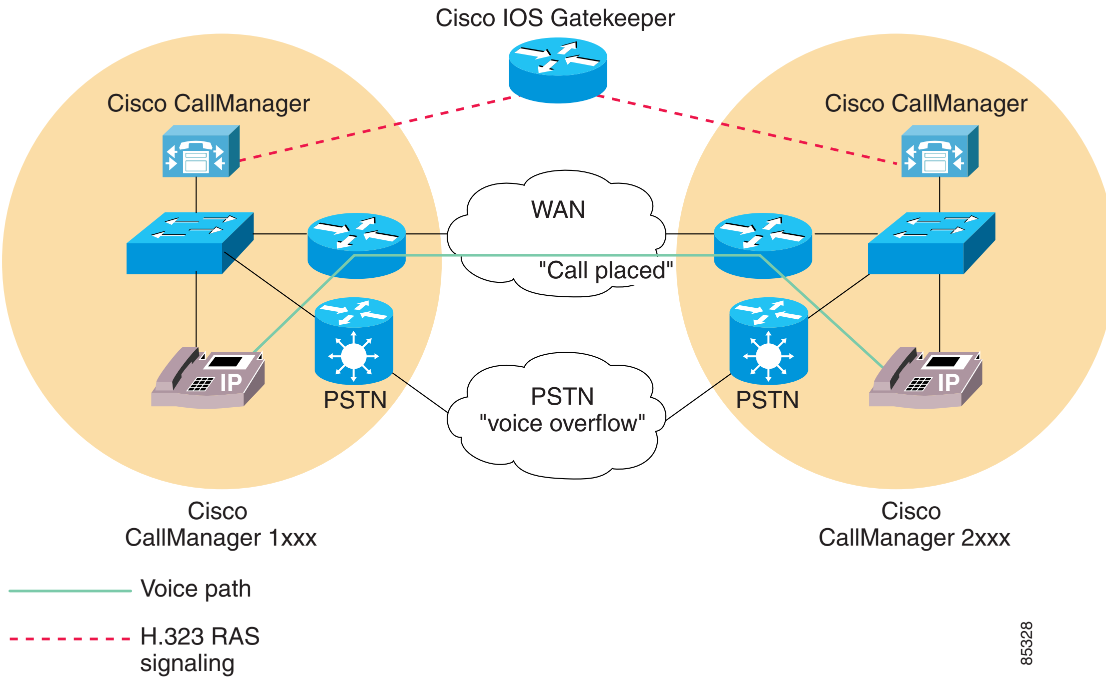
| Configuration Steps | | Procedures and Related Topics |
|---|---|---|
| **Step 3** | Configure the device pools for your system and choose the appropriate region for each. | Device Pool Configuration, *Cisco CallManager Administration Guide* |
| **Step 4** | Configure the phones and other devices and assign each of them to the appropriate device pool and location.<br>**Note** If you set the location to *None*, you assign that device to an unnamed location with unlimited available bandwidth and allow an unlimited number of active calls to and from that device. | Cisco IP Phones, page 40-1<br>Cisco IP Phone Configuration, *Cisco CallManager Administration Guide* |

# Gatekeepers and Trunks

A gatekeeper device, the Cisco Multimedia Conference Manager (MCM), provides call admission control for distributed call-processing systems. In a distributed system, each site contains its own call-processing capability. For example, Figure 8-3 shows two sites, each with its own Cisco CallManager, that are connected by an IP WAN link. A gatekeeper provides call admission control over the IP WAN link in this example.

In addition to call admission control, gatekeepers can also perform E.164 address resolution to route calls between sites. For example, in Figure 8-3, the extension range for one Cisco CallManager specifies 1XXX and 2XXX for the other. Both register with the gatekeeper for call admission control. Each Cisco CallManager incorporates an appropriate entry in its respective dial plan route pattern configuration that points the other Cisco CallManager extension number range to the gatekeeper. In practice, when user 1001 dials user 2002, Cisco CallManager 1XXX sends 2002 to the gatekeeper for address resolution. If the call satisfies the call admission control criteria, the gatekeeper returns the IP address of Cisco CallManager 2XXX to Cisco CallManager 1XXX. Using the IP address of Cisco CallManager 2XXX, Cisco CallManager 1XXX can then complete the call to directory number 2002.

*Figure 8-3 Call Admission Control Using a Gatekeeper in a Distributed System*




If the IP WAN is not available in this scenario, the call cannot go through as dialed. To simplify the dial plan and also provide fallback to the PSTN, use 10-digit dialing (or adhere to the national dial plan). For example, under the North American Numbering Plan (NANP), a route pattern of XXXXXXXXXX would direct calls to the gatekeeper for address resolution. If the gatekeeper does not allow the call to go over the WAN, Cisco CallManager can add the prefix 91 to the dialed digits to reroute the call through the PSTN.

Refer to the *Cisco IP Telephony Solution Reference Network Design Guide* for more detailed information about gatekeeper configuration, dial plan considerations when using a gatekeeper, and gatekeeper interaction with Cisco CallManager.

Trunks replace all previously configured intercluster trunk devices. An H.225 trunk device represents a logical route to the wholesale network. Previously configured anonymous devices with H.225 protocol migrate to H.225 trunks with gatekeeper control. Previously configured anonymous devices with intercluster protocol migrate to intercluster trunks with gatekeeper control. Previously configured intercluster gateways migrate to intercluster trunks without gatekeeper control.

Use an intercluster trunk to connect two Cisco CallManagers in remote clusters. For information about configuring gatekeeper-controlled intercluster trunks for routing intercluster calls across a remote WAN link, refer to the "Trunk Configuration" section of the *Cisco CallManager Administration Guide* and to the *Cisco IP Telephony Solution Reference Network Design Guide*.

You can configure H.323 gateways either with gatekeeper control or locally as gateways. If configuring with gatekeeper control, use an H.225 trunk.

# Components of Gatekeeper Call Admission Control

Gatekeeper call admission control provides great flexibility:

- Gatekeepers reduce configuration overhead by eliminating the need to configure a separate H.323 device for each remote Cisco CallManager that is connected to the IP WAN.
- A gatekeeper can determine the IP addresses of devices that are registered with it, or you can enter the IP addresses explicitly.
- The gatekeeper offers a choice of protocols for communicating with Cisco CallManagers or H.225 gateways.
- The gatekeeper can perform basic call routing in addition to call admission control.
- You can connect up to 100 Cisco CallManager clusters to a single gatekeeper.

The following sections describe the components of gatekeeper call admission control:

- Gatekeeper and Trunk Configuration on the Router, page 8-10
- Gatekeeper and Trunk Configuration in Cisco CallManager, page 8-12

## Gatekeeper and Trunk Configuration on the Router

Recommended platforms for the gatekeeper include Cisco 2600, 3600, 3700, or 7200 routers with Cisco IOS Release 12.1(3)T or higher. When configuring the gatekeeper function on one of these routers, you define a set of zones for call admission control. The unique name for each zone includes the IP address of each Cisco CallManager that registers with that zone, the zone prefix (directory number range), and the bandwidth that is allocated for that zone.

Cisco CallManager registers with a gatekeeper by using its IP address. You can specify the IP address in one of the following ways:

- Use the **gw-type-prefix** command on the gatekeeper to specify each Cisco CallManager IP address explicitly.
- In the Technology Prefix field under **Device** > **Trunk** in Cisco CallManager Administration, enter **1#*** and enter the command **gw-type-prefix 1#* default-technology** on the gatekeeper. When a Cisco CallManager registers with the gatekeeper, it sends its IP address and the specified technology prefix to the gatekeeper. The gatekeeper then registers this Cisco CallManager as a valid gatekeeper-controlled VoIP device.

You can associate the Cisco CallManager IP address with a particular zone in one of the following ways:

- Use the **zone local** command on the gatekeeper to define local zones. Enter the zone name in the Zone field.
- In the Zone field under **Device** > **Trunk** in Cisco CallManager Administration, enter the zone name. When a Cisco CallManager registers with the gatekeeper, it sends its IP address and the specified zone name to the gatekeeper. The gatekeeper then registers each Cisco CallManager and associates it with the appropriate zone.

To specify the directory number range for a particular Cisco CallManager, you use the **zone prefix** command to configure the range on the gatekeeper. For example, the following command specifies that the DN for zone LHR ranges from 3000 to 3999.

```
zone prefix LHR 3...
```

The maximum number of active calls that are allowed per zone depends on the codec that is used for each call and the bandwidth that is allocated for the zone. With Cisco CallManager, G.711 calls request 128 kbps, and G.723 and G.729 calls request 20 kbps. Use regions in Cisco CallManager to specify the codec type and use the **zone bw** command on the gatekeeper to specify the available bandwidth. For example, the following command allocates 512 kbps to the LHR zone.

```
zone bw LHR 512
```

With an allocation of 512 kbps, the LHR zone in this example could support up to four G.711 calls at the same time.

For more information on programming the gatekeeper, refer to the Cisco Multimedia Conference Manager documentation.

## Gatekeeper and Trunk Configuration in Cisco CallManager

You can configure gatekeepers and trunks in Cisco CallManager administration to function in either of the following ways:

### Non-Gatekeeper-Controlled Trunks

In this case, you explicitly configure a separate intercluster trunk for each remote device cluster that the local Cisco CallManager can call over the IP WAN. You also configure the necessary route patterns and route groups to route calls to and from the various intercluster trunks. The intercluster trunks statically specify the IP addresses of the remote devices. To choose this method, use **Device** > **Trunk** and select Inter-Cluster Trunk (Non-Gatekeeper Controlled) in Cisco CallManager Administration.

**Note** For a local non-gatekeeper-controlled intercluster trunk, you must specify the IP addresses of all remote Cisco CallManager nodes that belong to the device pool of the remote non-gatekeeper-controlled intercluster trunk.

### Gatekeeper-Controlled Trunks

In this case, a single intercluster trunk suffices for communicating with all remote clusters. Similarly, you need a single H.225 trunk is needed to communicate with any H.323 gatekeeper-controlled endpoints. You also configure route patterns or route groups to route the calls to and from the gatekeeper. In this configuration, the gatekeeper dynamically determines the appropriate IP address for the destination of each call to a remote device, and the local Cisco CallManager uses that IP address to complete the call.

This configuration works well in large as well as smaller systems. For large systems where many clusters exist, this configuration helps to avoid configuring individual intercluster trunks between each cluster. To choose this method, use **Device** > **Trunk** and select Inter-Cluster Trunk (Gatekeeper Controlled) in Cisco CallManager Administration.

If you configure gatekeeper-controlled trunks, Cisco CallManager automatically creates a virtual trunk device. The IP address of this device changes dynamically to reflect the IP address of the remote device as determined by the gatekeeper. Use trunks when configuring the route patterns or route groups that route calls to and from a gatekeeper.

# Gatekeeper and Trunk Configuration Checklist

Table 8-2 lists the general steps for configuring call admission control that is based on gatekeepers and trunks.

***Table 8-2 Gatekeeper and Trunk Configuration Checklist***

| Configuration Steps | | Procedures and related topics |
|---|---|---|
| **Step 1** | On the gatekeeper device, configure the appropriate zones and bandwidth allocations for the various Cisco CallManagers that will route calls to it. | Refer to your Cisco Multimedia Conference Manager documentation. |
| **Step 2** | Configure gatekeeper settings in Cisco CallManager Administration.<br>Repeat this step for each Cisco CallManager that will register with the gatekeeper. Make sure Host Name or IP Address is set the same way on each Cisco CallManager. | Gatekeeper Configuration, *Cisco CallManager Administration Guide* |
| **Step 3** | Configure the appropriate intercluster trunks or H.225 trunks to specify gatekeeper information (if gatekeeper-controlled). | Trunk Configuration, *Cisco CallManager Administration Guide* |
| **Step 4** | Configure a route pattern to route calls to each gatekeeper-controlled trunk. | Understanding Route Plans, page 15-1<br>Route Pattern Configuration, *Cisco CallManager Administration Guide* |

# Where to Find More Information

### Related Topics

- Location Configuration, *Cisco CallManager Administration Guide*
- Region Configuration, *Cisco CallManager Administration Guide*
- Route Pattern Configuration, *Cisco CallManager Administration Guide*
- Gatekeeper Configuration, *Cisco CallManager Administration Guide*
- Gateway Configuration, *Cisco CallManager Administration Guide*
- Cisco IP Phones, page 40-1
- Cisco IP Phone Configuration, *Cisco CallManager Administration Guide*
- Understanding Video Telephony, page 41-1
- Trunk Configuration, *Cisco CallManager Administration Guide*

### Additional Cisco Documentation

- *Cisco IP Telephony Solution Reference Network Design Guide*
- Cisco Multimedia Conference Manager (Command Reference) IOS documentation