H.323 Gateway Configuration Between CallManager Express and CallManager

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Contents

Introduction

This document provides a sample configuration of the deployment of Cisco Unified CallManager Express (Cisco Unified CME) for branch offices in conjunction with a Cisco Unified CallManager deployed at a central office site. In this situation, the central Cisco Unified CallManager site can communicate with the remote CME with a H.323 gateway. In H.323 networks, Cisco Unified CME provides supplementary service interworking (H.450) with Voice over IP (VoIP) hairpin call routing when needed for intersite call transfer and forwarding.

Note: Direct MGCP integration between Cisco Unified CME IP phones and Cisco Unified CallManager is not supported.

Prerequisites

Requirements

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

- Knowledge of Cisco Unified Communications Manager (CallManager)
- Basic knowledge of Cisco Unified CME

Components Used

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

- Cisco Unified Communications Manager: **4.1(3)SR3b**
- CallManager Express: Cisco IOS® **12.4(9)T2**, **CME Version 4.0(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.

H.323 with CME

Both Cisco Unified CallManager and Cisco Unified CME support H.323, which you can use to create Cisco Unified CallManager–to–Cisco Unified CME links. Cisco Unified CME also supports SIP for VoIP interconnect. SIP has also been introduced as a WAN trunking interface on Cisco Unified CallManager. This document focuses only on the H.323 interconnect option. The information contained in this document applies to the Cisco Unified CME 3.1 and 3.2 releases and the Cisco Unified CallManager 3.3(3) and 4.0. Newer versions can have different behaviors and options than those described here.

Creation of a New CME Site

Create New Region

When you create a new CME site, it can require a new **region** (for Codec selection), a new **location** (for bandwidth control), and a new **device pool**. Some sites can also create local **media resources**. In this section, you are presented with the information to configure the features described in this document.

In order to create a new region, go to **System > Region** from the Cisco Unified Communication Manager Administration page.

In the Region Name field, enter the name that you want to assign to the new region. Choose a value from the drop−down list box for the default codec to use between this region and other regions. Click **Insert**.

In the Audio Codec column, use the drop−down list boxes to choose the audio codec to use for calls within the new region and between the new region and existent regions. The audio codec determines the type of compression and the maximum amount of bandwidth that is allocated for these calls.

Add New Location

This section describes how to add a new location to the Cisco CallManager database. Use locations to implement call admission control in a centralized call−processing system. Call admission control enables you to regulate audio quality and video availability because it limits the amount of bandwidth that is available for audio and video calls over links between the locations.

Perform the procedure below to add a new location.

1. Choose **System > Location**.
2. In order to add a location, use one of these methods:

    ♦ If a location already exists with settings that are similar to the one that you want to add, choose the existent location to display its settings. Click **Copy**, and modify the settings as needed.
    ♦ In order to add a location without the need to copy an existent one, continue with Step 3.
3. In the upper, right corner of the window, click the **Add a New Location** link. Enter the appropriate settings.
4. In order to save the location information in the database, click **Insert**.

Note: When calls cannot use the link for a location, it is possible that bandwidth leakage has occurred that can reduce the allotted bandwidth for the location. You can resynchronize the location bandwidth to the maximum amount that is assigned to this location without the need to reset the Cisco CallManager server. Find the location and click **ReSync Bandwidth** to resynchronize the bandwidth for the chosen location.

Create New Device Pool

Use the Device Pool Settings to define sets of common characteristics for devices such as the Date/Time Group, Region, SRST Reference, Media Resource Group List, etc.

Follow this procedure to add a new device pool.

1. Choose **System > Device Pool**.
2. Use one of these methods to add a device pool:

   ♦ If a device pool already exists with settings that are similar to the one that you want to add, choose the existent device pool to display its settings; click **Copy**, and modify the settings, as needed.
   ♦ In order to add a device pool without copying an existent one, continue with Step 3.
3. In the upper right corner of the window, click the **Add a New Device Pool** link.

4. Enter or edit the appropriate fields and click **Insert** to save the device pool information in the database.

Note: If the local IPT gateway provides DSP (Transcoding or Conferencing) services to local devices, they must also be configured with Media Resources , MRG, and MRGL.

Add Gateway to CallManager

Before you add the gateway, you need to check the interface IP address used by the CME router. Issue these commands in the CME Router to validate the IP address in use by the IOS Telephony–Service.

```
CMErouter#sh telephony-service | inc ^ip
ip source-address 10.252.107.5 port 2000
```

This gateway uses 10.252.107.5 as the IP address.

Inspect which interfaces use the above IP address, as well as the status of the interfaces.

```
CMERouter#sh ip int brief | inc 10.252.107.5
Service-Engine0/0          10.252.107.5    YES TFTP    up                      up
Loopback1                  10.252.107.5    YES TFTP    up                      up
```

Note: The **Service–Engine 0/0** slot in use by Cisco Unity Express runs in the Unnumbered mode.

In order to learn more information about the interface service–engine 0/0, use this command.

```
CMERouter#show runnning intferace service-engine0/0
!
interface Service-Engine0/0
 ip unnumbered Loopback1
 service-module ip address 10.252.107.6 255.255.255.252
 service-module ip default-gateway 10.252.107.5
end
```

Create Gateway from CallManager Administration Page

Follow this procedure to create a H.323 gateway.

1. In order to create a H.323 gateway from the CallManager Administration page, choose **Device> Gateway**Click **Add a New Gateway**.

2. Choose **H.323 Gateway** and click **Next**.

3. Enter a unique name for the Cisco CallManager to use to identify the device. Use either the IP address or the host name as the device name. The new Gateway needs to use distinct site settings, such as Device Pool or Location.

Cisco CallManager Administration
For Cisco IP Telephony Solutions

CISCO SYSTEMS

Gateway Configuration

Product : H.323 Gateway

Gateway : 10.252.107.5
Device Protocol: H.225
Registration: Unknown
IP Address:

Status: Ready

[Update] [Delete] [Reset Gateway]

Device Information

Device Name*	10.252.107.5
Description	shanghailab1
Device Pool*	C1101_CNSHLB_DP
Call Classification*	OnNet
Media Resource Group List	APCN01_MRGL
Location	C1101_CNSHLB28
AAR Group	< None >
Tunneled Protocol	< None >
Signaling Port*	1720

☐ Media Termination Point Required

☑ Retry Video Call as Audio

☑ Wait for Far End H.245 Terminal Capability Set

☐ Path Replacement Support

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain (e.g., "0000FF")	
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

Call Routing Information
Inbound Calls

Significant Digits*	All
Calling Search Space	COS1_Phone_CSS
AAR Calling Search Space	< None >
Prefix DN	

☐ Redirecting Number IE Delivery – Inbound

☐ Enable Inbound FastStart

Outbound Calls

Calling Party Selection*	Originator
Calling Party Presentation*	Default
Called party IE number type unknown*	Cisco CallManager
Calling party IE number type unknown*	Cisco CallManager
Called Numbering Plan*	Cisco CallManager
Calling Numbering Plan*	Cisco CallManager
Caller ID DN	

☐ Display IE Delivery

☐ Redirecting Number IE Delivery – Outbound

☐ Enable Outbound FastStart

Note: After all configuration settings are validated, the H.323 gateway should be Updated and Reset.

Create Route Group for Gateway

Follow this procedure to create a new route group for the new H.323 gateway.

1. In order to create a new route group for the new H.323 gateway, choose **Route Plan > Route/Hunt > Route Group**.

2. Assign a new name for the **Route Group** and add the H.323 gateway to the route group.

Create Call Routing for New Pattern

Call Routing Build Order

The order in which to add the call routing is this:

1. Gateway
2. Route Group
3. Route List
4. Route Pattern

Add New Route List

Follow this procedure to create a new route list for the new dial pattern.

1. In order to create a new route list for the new dial pattern, choose **Route Plan > Route/Hunt > Route List**.
2. Click **Add a New Route List**.

   Use concise and descriptive names for your route lists. The CompanynameLocationCalltype format usually provides a sufficient level of detail and is short enough to enable you to quickly and easily identify a route list.

Note: Two route groups are associated with this route list: one for OnNet calls from the H.323 gateway to the CME router and another for OffNet calls to the CME router through PSTN. OffNet calls need to translate the called number to use the PSTN circuits.

The route list details that are associated with the Failover route group look like this with the calling−party and called−party transformations.

Add New Route Pattern

Follow this procedure to add a new route pattern.

1. In order to add a new route pattern, choose **Route Plan > Route/Hunt > Route Pattern** from the CallManager Administration page.
2. Click **Add a New Route Pattern**.

Route Pattern Configuration

Add a New Route Pattern
Back to Find/List Route Patterns

Route Pattern: New
Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

[Insert]

Pattern Definition

Route Pattern*	2XX
Partition	APCN01_Phones_PT
Description	C1101_CNSHLB - Shangahi Lab
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
MLPP Precedence	Default
Gateway or Route List*	C1101_CNSHLB_RL
Route Option	⊙ Route this pattern
	○ Block this pattern — Not Selected —
Call Classification*	OnNet □ Allow Device Override

□ Provide Outside Dial Tone   □ Allow Overlap Sending   □ Urgent Priority
□ Require Forced Authorization Code
   Authorization Level   0
□ Require Client Matter Code

Calling Party Transformations

□ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation	Default
Calling Name Presentation	Default

Connected Party Transformations

Connected Line ID Presentation	Default
Connected Name Presentation	Default

Called Party Transformations

Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	

ISDN Network-Specific Facilities Information Element

Carrier Identification Code	
Network Service Protocol	— Not Selected —

Network Service	Service Parameter Name	Service Parameter Value
— Not Selected —	< Not Exist >	

* indicates required item.

Note: Make sure the route pattern is in an appropriate partition and any needed Calling Search Spaces (CSS). In this example, we put the route pattern in the same partition as the phones so that no additional CSS configuration is required to make this pattern reachable.

Verify

This section of the document explains how verify the details of active calls and dial–peers.

Verification on CME

Use this section to confirm that your configuration works properly.

Verify the dial–peer configured on the CME.

```
shanghailab1#sh dial-peer voice summary | inc 5678
            AD                                  PRE PASS                OUT
 TAG    TYPE  MIN  OPER PREFIX    DEST-PATTERN   FER THRU SESS-TARGET    STAT PORT
 5000   voip  up   up             [5678]..        1  syst ipv4:172.21.21.21
 5001   pots  up   up             [5678]..        2                      up   0/2/0
 5003   pots  up   up             [5678]..        4                      up   0/2/2
 5004   pots  up   up             [5678]..        5                      up   0/2/3
 5002   pots  up   up             [5678]..        3                      up   0/2/1
```

Note: Make sure that the VoIP dial–peer session target points to the CallManager IP address.

The Output Interpreter Tool (registered customers only) (OIT) supports certain **show** commands. Use the OIT to view an analysis of **show** command output.

Check CallManager for Call Admission Control

Check the CallManager for Call Admission Control (CAC) through the locations parameter. Verify that Call Admission Control monitors the bandwidth in use.

Go to **Start > Programs > Administrative Tool > Performance > Cisco CallManager > Location**.
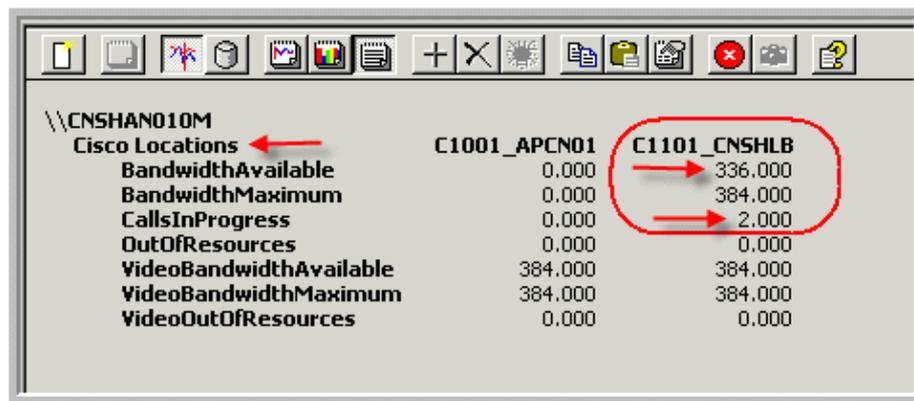
Troubleshoot

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

Related Information

- **How to apply CAC based on CPU & Memory Utilization to prevent gateway overload**
- **Voice Technology Support**
- **Voice and Unified Communications Product Support**
- **Troubleshooting Cisco IP Telephony**

- **Technical Support & Documentation – Cisco Systems**

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