



SIP-Based Trunk Managed Voice Services Solution Design and Implementation Guide

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Introduction

Cisco Unified Communications delivers fully integrated communications systems by enabling data and voice to be transmitted over a single network infrastructure using standards-based Internet Protocol (IP). Leveraging the framework provided by Cisco IP hardware and software products, Cisco Unified Communications delivers unparalleled performance and capabilities to address current and emerging communications needs in service provider, enterprise, and commercial business environments.

This guide discusses a solution network design to enable enterprise Session Initiation Protocol (SIP) trunk deployment with Cisco Unified Communications Manager (Cisco Unified CM) and Cisco Unified Survivable Remote Site Telephony (Cisco Unified SRST), one of the several SIP trunk solutions that Cisco is developing. The model of enterprise SIP trunk development with Cisco Unified CM and Cisco Unified SRST is especially geared for large enterprises with many branch offices. In this distributed model, the service provider (SP) furnishes the SIP trunk services for the enterprise to connect the enterprise headquarter with its enterprise branch offices. At the enterprise headquarter, Cisco Unified CM provides call control for voice services. Remote enterprise branch offices have Cisco Unified SRST deployed for voice services. The Cisco Integrated Services Router (Cisco ISR) running the Cisco Unified Border Element (Cisco UBE) is placed at the edge of the network. Cisco UBE plays an important role in serving multiple functions when connecting to other networks.

This design guide discusses the components deployed in the network, and provides sample router configurations for the Cisco UBE functions tested for the features included in this document.

Use this information to deploy enterprise SIP trunks with Cisco Unified CM and Cisco Unified SRST using service provider networks.

Network Topology

The components of the enterprise SIP trunk deployment with Cisco Unified CM and Cisco Unified SRST network topology is show in [Figure 1](#). The service provider components are listed for completeness only and are not included in this guide.

Enterprise Headquarter

- [Enterprise 1 HQ Cisco UBE Example Configuration, page 29](#)
- [Enterprise 1 HQ Cisco Unified CM Example Configuration, page 32](#)
- [Enterprise 1 HQ Cisco ASA Firewall Example Configuration, page 120](#)
- [Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration, page 119](#)
- [Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration, page 119](#)

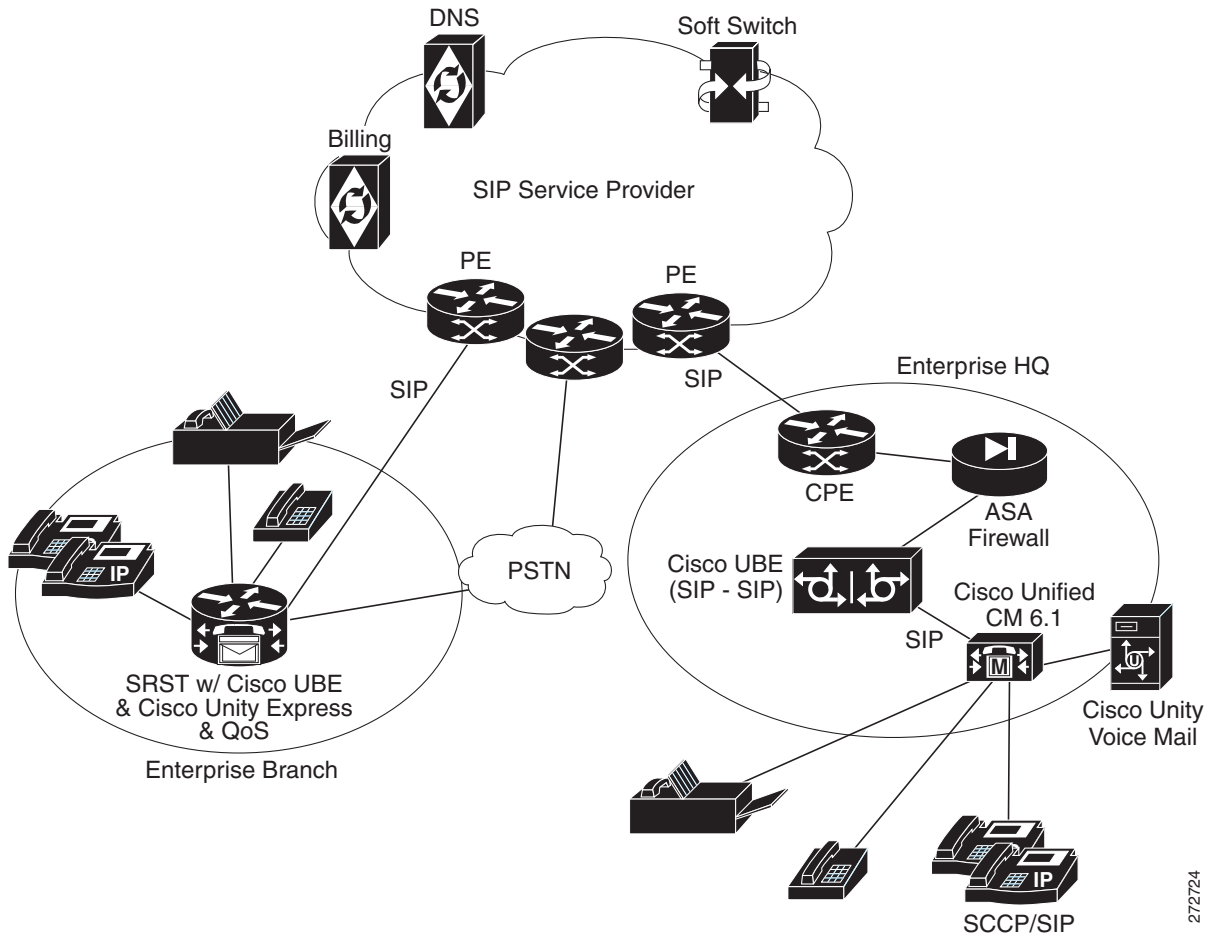
Enterprise Branch

- [Branch 1 Cisco UBE, TDM Gateway, and Cisco Unified SRST Example Configuration, page 121](#)
- [Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration, page 125](#)

Service Provider

- PSTN hop-off gateway
- SIP Call Agent
- Multiprotocol Label Switching (MPLS) core network

Figure 1 Enterprise SIP Trunk Deployments Cisco Unified CM and Cisco Unified SRST with Cisco UBE



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Prerequisites

Prerequisites are grouped into the following sections:

- [Components Used, page 4](#)
- [Cisco IOS Software Releases, page 6](#)
- [Conventions, page 6](#)

Components Used

The information in this guide is based on the software and hardware versions listed in the following sections. The configuration shown in this guide was created through the use of the devices in a specific lab environment. This section includes prerequisites for the following components:

- [Cisco Unified Communications Manager, page 5](#)
- [Cisco Unified Border Element, page 5](#)
- [SCCP Analog Voice Gateway, page 5](#)
- [Voice Mail at the Enterprise Headquarter Site, page 5](#)
- [Cisco Adaptive Security Appliance Firewall Appliance, page 5](#)
- [Cisco Survivable Remote Site Telephony, page 5](#)

Cisco Unified Communications Manager

The Cisco Unified CM at the enterprise headquarter site provides call control to voice services at the headquarter site and the branch offices. The Cisco Unified CM was tested using version 6.1.x.

Cisco Unified Border Element

A Cisco 3800 series platform was tested with Cisco IOS Release 12.4.(20)T1 and Cisco UBE version 1.2. The Cisco 2800 series Integrated Services Router (Cisco ISR) can also be used as a Cisco UBE.

SCCP Analog Voice Gateway

A Cisco VG224 analog voice gateway was used at the enterprise headquarter site to provide connectivity to analog phones and fax machines. The Cisco VG224 analog voice gateway was tested with Cisco IOS Release 12.4(20)T1.

Voice Mail at the Enterprise Headquarter Site

Voice mail at the enterprise headquarter site is provided by the Cisco Unity voice mail server, tested with version 3.2.

Cisco Adaptive Security Appliance Firewall Appliance

A Cisco ASA firewall appliance was placed at the ingress from the service provider servicing the enterprise headquarter site. It was tested with Cisco ASA 8.0(4).



Note

The Cisco UBE at the enterprise headquarter site can also be used to provide Cisco IOS firewall functions. If the Cisco UBE is used to provide Cisco IOS zone-based firewall functions, the Cisco ASA firewall appliance is not needed.

Cisco Survivable Remote Site Telephony

A Cisco Unified SRST router was placed at the enterprise branch site. In addition to the Cisco Unified SRST functions, this router provides Cisco UBE, Cisco IOS firewall, conferencing transcoding, MTP, voice mail using Cisco Unity Express, TDM, and gateway functions. A Cisco 3800 series platform was tested with Cisco IOS Release 12.420T1. Cisco Unity Express was tested with version 3.2. The Cisco 2800 series Integrated Services Router (Cisco ISR) can also be used as an Cisco Unified SRST router.

Cisco IOS Software Releases

The test results described in this guide for the Cisco Unified Border Element were conducted using Cisco IOS Release 12.4(20)T1. We recommend Cisco IOS Release 12.4(20)T1 or later releases for the deployment of the features described in this guide.

Conventions

Refer to [Cisco Technical Tips Conventions](#) for information on document conventions.

Solution Description

The enterprise SIP trunk deployment with the Cisco Unified CM and Cisco Unified SRST solution topology allows the enterprise headquarter site to provide voice services from Cisco Unified CM to remote enterprise branch offices using SIP trunks from service providers. The enterprise branch offices are equipped with Cisco Unified SRST routers.

When Cisco Unified CM fails, but the WAN connection remains active and SRST takes over, the remote phones are able to make WAN calls through SIP to the call agent. If a WAN connectivity failure occurs, the enterprise branch offices can continue to maintain basic IP phone and PSTN services.

The focus of services using this solution are:

- Voice services with call control provided by Cisco Unified CM at the enterprise headquarter site
- Voice services with Cisco Unified SRST at the enterprise branch offices

The following topics describe the solution:

- [Feature Summary, page 6](#)
- [IP Connectivity, page 15](#)
- [Quality of Service, page 16](#)
- [Voice Mail, page 18](#)
- [Dial Plan, page 18](#)
- [Security, page 18](#)
- [Failover and Redundancy, page 19](#)
- [Fax and Modem, page 19](#)
- [Billing and Management, page 19](#)
- [Best Practices for SIP Trunk implementation Using Cisco UBE, page 19](#)
- [Caveats, page 21](#)

Feature Summary

The features listed in this section were tested as part of the solution configuration.

Enterprise Headquarter Site Features

- Cisco Unified Communications Manager call control

- Cisco Unified Border Element
- Cisco ASA Firewall or Cisco IOS Zone-Based Firewall
- Cisco Unity Voice Mail Server
- Analog Phone and Fax Services

Enterprise Branch Offices Features

- Survivable Remote Site Telephony
- Cisco Unified Border Element
- Cisco IOS Firewall
- Cisco Unity Express Voice Mail
- Analog Phone and Fax Services
- PSTN Backup

Service Provider Features

- Multiprotocol Label Switching (MPLS) in the service provider backbone network
- PSTN Hop-Off Services (using service provider shared PSTN gateway)
- Optional Voice Mail Server

Basic Phone Features Served in the Topology

- Basic and Supplementary Calls
- DTMF Relay RFC 2833
- Fax and Modem Passthrough
- Supplementary services: Hold, Transfer, Forward, Conferencing, Transcoding, Music-on-Hold, Delayed Offer, Early Offer
- Calls to service provider PSTN gateway, inbound and outbound
- Voice mail services (Cisco Unity at the enterprise headquarter site and Cisco Unity Express at the enterprise branch offices)

SIP Trunking Design Considerations

SIP trunking design considerations described in the following sections should be assessed when deploying SIP trunks.

- [DTMF Transport, page 8](#)
- [SIP Delayed Offer and Early Offer, page 8](#)
- [Early Media Cut Through, page 9](#)
- [SIP Trunk Transport Protocols, page 9](#)
- [Monitoring SIP Trunk State, page 9](#)

DTMF Transport

There are several ways of transporting DTMF information between SIP endpoints. In general, these methods can be classified as Out of Band (OOB) and In Band (IB) signaling. In Band DTMF transport methods send either raw or signaled DTMF tones within the RTP stream and need to be processed by the endpoints that generate or receive them.

OOB signaling methods transport DTMF tones outside of the RTP stream, either directly to and from the endpoints or using a Call Agent, such as the Communications Manager, which interprets and forwards these tones as required.

OOB SIP DTMF signaling methods include:

- Unsolicited SIP Notify
- INFO method
- Key Press Markup Language (KPML)

KPML (RFC 4730) is the preferred OOB signaling method used by Cisco. KPML is supported on Advanced Cisco 79X1 Series IP Phones, Cisco Unified CM, and Cisco IOS Gateways (Cisco IOS Release 12.4 and later).

Unsolicited Notify is a proprietary DTMF transport method used only on Cisco IOS Gateways (Cisco IOS Release 12.2 and later).

IB DTMF transport methods send DTMF tones as either raw tones in the RTP media stream or as signaled tones in the RTP payload, using RFC 2833.

With SIP product vendors, RFC 2833 has become the predominant method of sending and receiving DTMF tones and is supported by the majority of Cisco voice products.

Because IB signaling methods send DTMF tones in the RTP media stream, the SIP endpoints in a session must either support the transport method used (for example, RFC 2833) or provide a method of intercepting this in band signaling and converting it. That is, if two endpoints are using a B2BUA as the call control agent (such as the Communications Manager) and they negotiate different DTMF transport methods, then the call control agent determines how these DTMF transport differences are handled. With Communications Manager, a DTMF transport mismatch (for example, In Band to Out of Band DTMF) is resolved by inserting a transcoder

SIP Delayed Offer and Early Offer

RFC 3261 defines two ways that Session Description Protocol (SDP) messages can be sent in the offer and answer, commonly known as Delayed Offer and Early Offer, which are mandatory requirements in the specification. In the simplest terms, an initial SIP Invite sent with SDP in the message body defines an Early Offer; whereas, an initial SIP Invite sent without SDP in the message body defines a Delayed Offer. In an Early Offer, the session initiator sends its capabilities in the SDP contained in the initial invite (for example, codecs supported). In a Delayed Offer, the session initiator does not send its capabilities in the initial invite and waits for the called device to send its capabilities first.

Cisco UBE uses the SIP *Offer/Answer* model for establishing SIP sessions, as defined in RFC 3264. In this context, an *Offer* is contained in the SDP fields sent in the body of a SIP message.

**Note**

Service providers sometimes mandate an Early Offer call from the enterprise. In such cases Cisco UBE (Cisco IOS Release 12.4(20)T and later) can be configured to convert the Delayed Offer to the Early Offer.

Early Media Cut Through

The terms Early Offer and Early Media are often confused.

- Early Offer is the call setup where the initial Invite has the SDP Offer.
- Early Media is the preconnect media cut-through.

In certain circumstances, a SIP session can require that a media path be set up prior to completing a connection. To this end, the SIP protocol allows the establishment of Early Media after the initial Offer has been received by an endpoint. The reasons for using Early Media vary.

- The called device might establish an Early Media RTP path to reduce the effects of audio cut-through delay (clipping) for calls experiencing long signaling delays, or to provide a network-based voice message to the caller.
- The calling device might establish an Early Media RTP path to access a DTMF or voice driven IVR system (for example, airlines).

Both Early Offer and Delayed Offer calls support Early Media. Early Offer calls can typically stream Early Media after exchanging two messages (Invite with SDP and Trying). Delayed Offer calls can typically stream Early Media after exchanging four messages (Invite without SDP, 100 Trying, Session Progress with SDP and PRACK).

If Cisco UBE is configured to do DO->EO conversion, ensure that PRACK is enabled on CUCM, for call flows involving early media cut-through (18x w/SDP) to work seamless.

SIP Trunk Transport Protocols

SIP Trunks can use either TCP or UDP as a message transport protocol. As a reliable, connection orientated protocol that maintains the connection state per SIP dialogue, TCP is preferred. However, TCP has a higher segment overhead, uses more bandwidth than UDP, and has a higher packet overhead. These TCP overhead features increase call setup times when compared with UDP, which is connectionless and relies on the SIP stack to maintain its state and reliability.

If your network is prone to packet loss, use TCP. If the networks do not experience packet loss, use UDP.

Monitoring SIP Trunk State

SIP servers can monitor individual SIP dialogues either by using the dialogue's TCP connection or within the SIP stack itself (for example, for UDP based transport). In a Cisco Unified CM environment, use this per-call trunk state tracking feature in conjunction with Cisco Unified CM Route Groups and Route Lists to route calls over multiple SIP trunks. Trunk state is monitored and state changes are detected on a per-call basis. Successive trunk connections are attempted when the first trunk and subsequently selected trunks are down.

To overcome the limitations of per-call, per trunk state detection, the following methods can be used to monitor the state and detect the state changes of each end of a SIP trunk:

- **OPTIONS Method**—The SIP OPTIONS method allows a UA to query another UA or a proxy server as to determine its capabilities. This query allows a client to discover information about the supported methods, content types, extensions, codecs, and so on, without actually placing a call.

Cisco UBE sends an Out of Dialogue OPTIONS message to the device at the far-end of the SIP trunk to determine its state. The OPTIONS method is used as an application-level ping. The returned ping response is generally not as important as the fact that the trunk has confirmed that it is *alive*. Cisco Unified CM SIP trunks support the receipt of OPTIONS messages but do not send OPTIONS messages as keepalives. Cisco Unified CM version 5.x SIP trunks respond to OPTIONS messages with a “405—Method Not Acceptable” response. In Cisco Unified CM version 6.0.1, SIP trunks respond to an OPTIONS message with a “200—OK” response.

- **INVITEs as keepalives**—INVITEs that are sent to unused numbers on the SIP trunk is an alternative to the OPTIONS method as an application-level ping. Similar to the OPTIONS method, the response returned is generally not as important as the fact that the trunk has confirmed that it is *alive*. Cisco Unified CM responds to, but does not send SIP INVITEs as keepalives.

SIP Trunk Redundancy and Load Balancing

Redundancy can be achieved by combining the call admission control (CAC) features of IOS. In general, CAC can be applied based on IP address reachability, Total Memory, Total Calls, Total CPU, IP circuit max-calls, and max-connections. The following show several methods used to achieve redundancy based on:

- [Dial-peer preferences and Dial-peer Hunting](#)
- [DNS SRV](#)
- [GK load balancing for H.323 Networks](#)
- [Route List & Route Group option from CCM](#)

Dial-peer preferences and Dial-peer Hunting

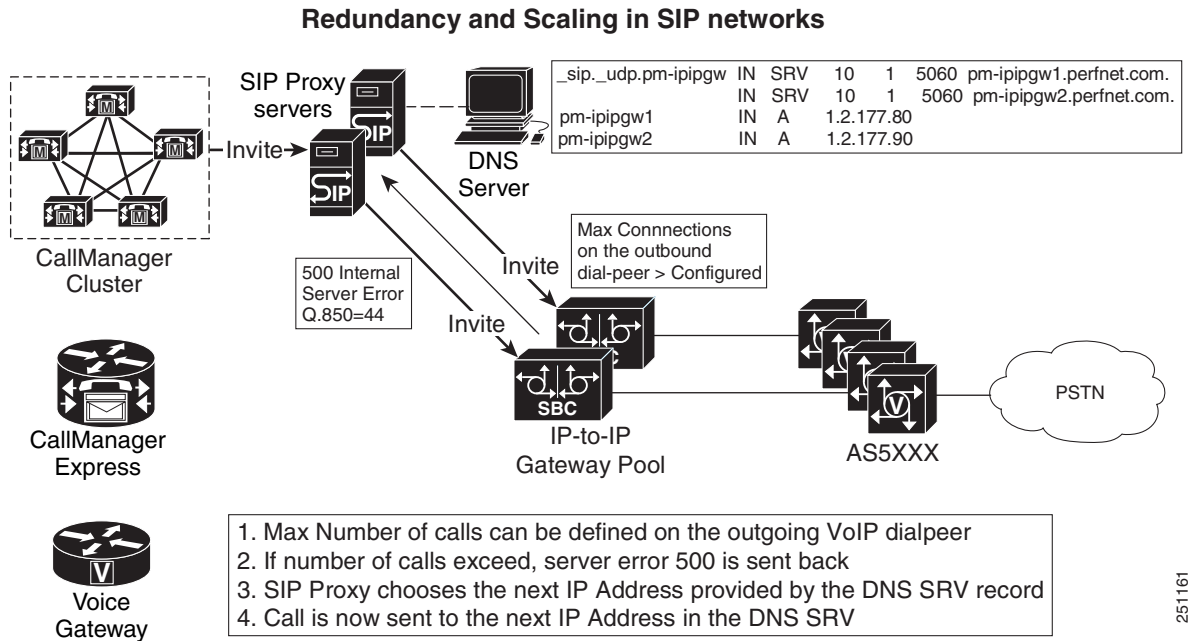
Use the following CLI example to achieve redundancy based on dial-peer preferences and dial-peer hunting:

```
!
dial-peer voice 3670000 voip
description "first hunting for 3670000 to ent2-hq-ipp"
destination-pattern 240367...
session protocol sipv2
session target ipv4:10.10.11.36
codec g711ulaw
!
dial-peer voice 36700 voip
description "second hunting for 3670000 to ent2-hq-ipp"
destination-pattern 240367...
preference 1
session protocol sipv2
session target ipv4:10.10.11.37
codec g711ulaw
!
```

DNS SRV

Use the setup example shown in [Figure 2](#) into achieve redundancy based on DNS SRV.

Figure 2 SIP Network Redundancy and Scaling Based on DNS SRV

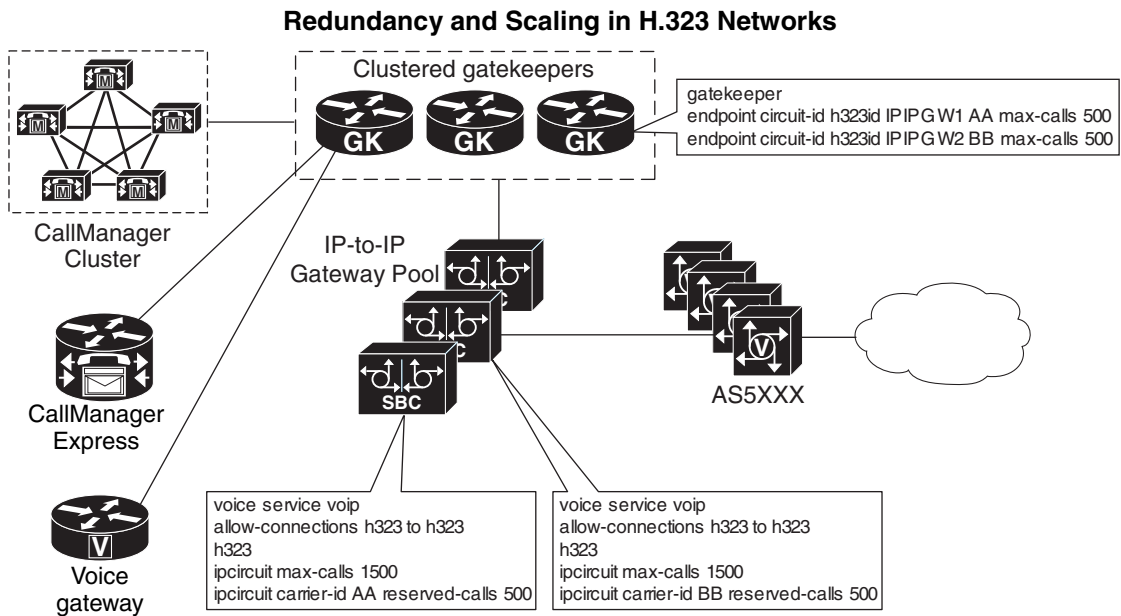


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GK load balancing for H.323 Networks

Use the setup example shown in Figure 3 to achieve redundancy based on GK load balancing for H.323 networks.

Figure 3 Redundancy and Scaling Based on GK Load Balancing for H.323 Networks



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Route List & Route Group option from CCM

To achieve redundancy based on route list and route group using Cisco Unified CM, complete the following steps:

1. Configure one Route Group to each IPIPgw (see [Figure 4](#)).

Figure 4 Configuring Route Groups

Route Group Configuration

[Add new Route Group](#)
[Back to Find/List Route Groups](#)
[Dependency Records](#)

Route Group Members
 15.3.30.60

Route Group: loadbalance-ipipgw60-rg
 Status: Ready
 Update Delete

Route Group Information

Route Group Name* loadbalance
 Distribution Algorithm* Top Down

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices
 (select device, then select port below)

15.3.30.70
pinamojito-ipipgw1-15.5.15.80

Port(s) All Add to Route Group

Current Route Group Members

Reverse Order of Selected Devices

Selected Devices*
 (ordered by highest priority)

15.3.30.60 (All Ports)

Removed Devices
 (to be removed from Route Group when you click Update)

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2. Configure one Route List to club all Route Groups (see [Figure 5](#)).

Figure 5 **Configuring A Route List for Route Groups**

The screenshot displays a web interface titled "Find and List Route Groups" with a link to "Add a New Route Group". It shows search results for "Route Group Name begins with """. The search criteria are "Route Group Name" with a dropdown set to "begins with" and a "Find" button. The results show two items: "loadbalance-ipipgw60-rg" and "loadbalance-ipipgw70-rg". Navigation options include "Delete Selected", "First Previous Next Last", and "Page 1 of 1".

Find and List Route Groups [Add a New Route Group](#)



2 matching record(s) for Route Group Name begins with ""

Find Route Groups where Route Group Name

and show items per page

To list all items, click Find without entering any search text.

Matching record(s) 1 to 2 of 2

<input type="checkbox"/>	Route Group Name
<input type="checkbox"/>	 loadbalance-ipipgw60-rg
<input type="checkbox"/>	 loadbalance-ipipgw70-rg

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- Configure Route List under Route Pattern Gateway or Route List (see [Figure 6_](#).

Figure 6 *Configuring A Route List Under Route Pattern Gateway or Route List*

Route List Configuration

[Add a new Route List](#)
[Back to Find/List Route Lists](#)
[Dependency Records](#)

Route List Details	Route List: loadbalance-ipipgw-rl
<ul style="list-style-type: none"> loadbalance-ipipgw60-rg loadbalance-ipipgw70-rg 	<p>Status: Ready</p> <p style="text-align: right;"> <input type="button" value="Copy"/> <input type="button" value="Update"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> </p>
<h3>Route List Information</h3>	
<p>Route List Name* <input type="text" value="loadbalance-ipipgw-rl"/></p> <p>Description <input type="text" value="loadbalancebetween60-70"/></p> <p>Cisco CallManager Group* <input type="text" value="PUB"/></p> <p>WARNING! The selected Cisco CallManager Group has only one Cisco CallManager configured. For the control process to have redundancy protection, please select a Cisco CallManager Group with more than one Cisco CallManager.</p> <p><input checked="" type="checkbox"/> Enable this Route List (change effective on Update; no reset required)</p>	
<h3>Route List Member Information</h3>	
<input type="button" value="Add Route Group"/>	
<p>Selected Groups* (ordered by highest priority)</p>	<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> loadbalance-ipipgw60-rg[non-QSIG] loadbalance-ipipgw70-rg[non-QSIG] </div> <p style="text-align: center;">▼ ▲</p>
<p>Removed Groups (to be removed from Route List when you click Update)</p>	<div style="border: 1px solid #ccc; padding: 5px; min-height: 100px;"> (Empty) </div>
<p>* indicates required item</p>	

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- Configure Max-Con under IPIPgw dial-peers towards Meeting Place, or Set the Global Call Treatment for total-calls.

Figure 7 **Configuring Max-Con**

Route Pattern Configuration

[Add a New Route Pattern](#)
[Back to Find/List Route Patterns](#)

Route Pattern: 6XXX
Status: Ready
Note: Any update to this Route Pattern automatically resets the associated gateway or Route List

Copy Update Delete

Pattern Definition

Route Pattern* 6XXX
Partition < None >
Description via 15.5.15.60
Numbering Plan* North American Numbering Plan
Route Filter < None >
MLPP Precedence Default
Gateway or Route List* loadbalance-ippgw-rl (Edit)
Route Option
 Route this pattern
 Block this pattern — Not Selected —
Call Classification* OffNet 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

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IP Connectivity

The SIP trunks are typically provided by service providers (SPs). SP voice services are offered using a SIP trunk that uses the same physical IP interface also used to deliver data services. The options for the physical connection of SIP trunks from the SPs are shown in [Table 1](#).

The sample configuration in the [“Configurations” section on page 21](#) shows a Gigabit Ethernet interface.

Some service providers that offer both data and voice services over a single IP interface also offer MPLS services. With MPLS services, voice packets must be sent with an MPLS label so that the service provider can terminate the traffic, and data marked with a different label can be tunneled through the backbone network. Marking voice traffic with an MPLS label requires the Virtual Routing and Forwarding (VRF)-Aware voice feature available on the Cisco ISRs in Cisco IOS Release 12.4(20)T.

Table 1 Cisco CPE Router Network Connectivity Options

Physical Connection	Data Link
Fast Ethernet, Gigabit Ethernet	Metro Ethernet
Broadband Interface (HWIC-CABLE, WIC1-ADSL, WIC1-SHDSDL)	Cable modem, digital subscriber line (DSL), asymmetric digital subscriber line (ADSL)
T1/E1 (WIC-1DSU-T1, VWIC-2MFT-T1, VWIC-2MFT-E1)	Point-to-Point Protocol (PPP), Frame Relay, ATM

Quality of Service

Quality of Service (QoS) is a fundamental requirement for any IP interface that carries voice traffic. Several specific QoS considerations and their configurations are discussed in this section:

- [Congestion Management, page 16](#)
- [Packet Marking, page 17](#)
- [Call Admission Control, page 17](#)
- [Delay, page 17](#)
- [Echo, page 18](#)

Congestion Management

When you use a single connection for both voice and data, you must carefully consider congestion management and bandwidth allocation to prevent data flows from affecting voice quality.

VoIP signaling and media traffic can be identified and classified as priority traffic using the QoS tools available within Cisco IOS software. Use Low Latency Queuing (LLQ) for media traffic streams. During congestion, LLQ queues restrict throughput to the configured bandwidth and packets exceeding this bandwidth are dropped. Therefore, signaling traffic should use class-based weighted fair queuing (CBWFQ), because signaling traffic bursts during call setup and teardown. The configurations for LLQ and CBWFQ are shown in the “[Configurations](#)” section on page 21. See [Quality of Service for Voice Over IP](#) for more information.

You can estimate the bandwidth to allocate to voice traffic by considering:

- Codec used by the calls
- Maximum number of simultaneous calls over the SIP trunk
- Payload size of the packets (that is, the sampling size of the codec)

The service provider can limit the maximum number of calls allowed across the SIP trunk based on the CAC techniques discussed in the “[Billing and Management](#)” section on page 19. This maximum number of calls allowed can be part of the service level agreement (SLA) between the service provider and the end customer.

When a Layer 2 connection technology, like Frame Relay or ATM, is used, additional traffic shaping and traffic management mechanisms must be deployed to ensure QoS on the egress interface. See [Configuring Frame Relay](#) for more information.

Packet Marking

You must set appropriate differentiated services code point (DSCP) values on the media and signaling packets leaving the SIP trunk from the customer premises to receive the desired service level in the service provider's network. By default, Cisco IOS software on the CPE router marks voice media packets, sourced on the router, with DSCP EF (101110) for expedited forwarding and signaling packets, sourced on the router, with DSCP AF31 (011010) for assured forwarding.

QoS policies may use either DSCP or IP precedence to classify voice packets. IP precedence interprets the low order three bits of the 6-bit DSCP value. In this way DSCP EF maps to CS5, while DSCP AD31 maps to CS3, which are appropriate IP precedence settings for voice media and signaling traffic.

Call Admission Control

Different types of Call Admission Control (CAC) are used in this solution. CAC can be based on bandwidth, maximum connections, CPU load, or memory available. CAC can be enabled at Cisco Unified CM or Cisco UBE.

Bandwidth-based CAC monitors the amount of bandwidth available in the network and controls routing of calls accordingly. This provides guaranteed control of bandwidth usage for voice calls. On Cisco Unified CM, bandwidth-based CAC is available and tested.

The number of simultaneous outbound calls can also be limited by the **max-conn** command on the VoIP dial peer used to route calls from the Cisco UBE router to the service provider network. This is the mechanism tested in the configuration example given in this guide.

The Cisco UBE can control the number of calls by setting the CPU load or memory available. This is configurable on the Cisco UBE by setting the threshold such that CAC is triggered when the threshold is reached.

The service provider can also control the total number of inbound and outbound calls from the SIP feature server, which is probably the best place for CAC policies to be implemented.



Note

We recommend also implementing a limit such as that set by the **max-conn** command on the Cisco UBE side to protect against poor voice quality on the IP access link into the customer site if the number of calls exceeds the available bandwidth.

Delay

The telephone industry standard ITU-T G.114 recommends the maximum desired one-way delay for a voice packet be no more than 150 milliseconds (ms). With a round-trip delay of 300 ms or more, users can experience annoying talk-over. In addition to congestion management with proper queuing techniques, you can use link fragmentation and interleaving (LFI) on slower access links to ensure that the end-to-end delay budget for voice packets is met. LFI is usually necessary on links of less than 768K access speeds.

Variable delay in packet rate results in jitter. The jitter buffer in Cisco voice gateways runs in an adaptive mode and can remove the jitter from the packet flow for moderate end-to-end jitter in the network. See [Understanding Jitter in Packet Voice Networks \(Cisco IOS Platforms\)](#) for more information on jitter. Delay can also cause echo.

Echo

Echo is caused by a time-division multiplexing (TDM) connection, or acoustic echo resulting from IP connections and endpoints. An improperly insulated phone, headset, or speakerphone could be the cause of echo experienced across a SIP trunk call. The analog phone user can also hear echo because of a very hot, or very high volume, signal on the TDM interface. [Echo Analysis for Voice over IP](#) explains how to adjust the settings for the voice port to eliminate echo caused by a hot signal and contains details on troubleshooting the source of echo. Delayed echo could be from the PSTN connectivity in the service provider's network. Cancel this echo on the PSTN gateway.

Voice Mail

Voice mail is provided by the Cisco Unity server at the enterprise headquarter site. At the enterprise branch offices, voice mail is provided by Cisco Unity Express embedded in the Cisco Unified SRST router.

The service provider can offer voice mail services using a hosted server. In this configuration, the service provider SIP server is responsible for functions such as call forward busy, call forward no answer, and Message Waiting Indicator (MWI).

Dial Plan

In this solution topology, the voice services are provided by the service provider using a call agent. The dial plan is also controlled by the service provider. The configuration shows the call routing configuration for VoIP dial peers needed on the Cisco UBE.

Security

The following security features are included in the solution network design:

- [Authentication, page 18](#)
- [Encryption of Media and Signaling, page 18](#)
- [Firewall, page 19](#)

Authentication

SIP registration and call method authentication can be provided using Digest Authentication. This method uses a single username and password for the entire SIP trunk, as shown in the [“Configurations” section on page 21](#). The password is encrypted using Message Digest 5 (MD5).

Encryption of Media and Signaling

VPN technology can be used to encrypt the media and signaling streams between the Cisco UBE router and the core network. Cisco UBE also supports Transport Layer Security (TLS) and Secure RTP (SRTP) internally between phones and the router.

Firewall

At the enterprise headquarter site, either the Cisco ASA firewall appliance or Cisco IOS Zone-based firewall can be used to defend against outside attacks from the IP interface entering the headquarter. At the enterprise branch offices, the Cisco IOS Zone-based firewall features in the Cisco Unified SRST router are used. The firewall serves as a checkpoint for the customer LAN traffic exiting from the router to the service provider network.

Access control lists (ACLs) are required to filter out unwanted traffic on physical links to the Internet. These ACLs are used primarily to stop unauthorized access, Denial of Service (DoS) attacks, or distributed DoS (DDoS) attacks that originate from the service provider or a network connected to the service provider, and also to prevent intrusions and data theft.

In this test configuration, the Cisco ASA firewall appliance was used at the enterprise headquarter site and Cisco IOS firewall features were used at the enterprise branch offices.

Failover and Redundancy

If a complete SIP trunk failure or IP interface failure occurs, backup PSTN lines connected directly to Cisco Unified SRST can be used for PSTN access. In the Cisco Unified SRST router configuration shown in the [“Configurations” section on page 21](#), backup PSTN access was tested for alternate call routing when SIP trunk access was down.

Fax and Modem

Fax pass-through and modem pass-through calls were tested between the enterprise headquarter site and branch offices and to the PSTN hop-off gateway. Fax and modem calls were tested with the G.711 codec.

Billing and Management

Typically the service provider is able to do billing without using any information from the managed Cisco UBE router.

Each call through the Cisco UBE router is considered to have two call legs. The start and stop records are generated for each call leg and can be polled through Simple Network Management Protocol (SNMP) using the DIAL-CONTROL-MIB. For more information, see the following documents:

- [CDR Logging with Syslog Servers and Cisco IOS Gateways](#)
- [Equivalent MIB Objects for VoIP show Commands](#)
- [RADIUS VSA Voice Implementation Guide](#)

Best Practices for SIP Trunk implementation Using Cisco UBE

By using the following Cisco UBE configuration methods, you can achieve a more effective SIP trunk topology implementation.

- Configure explicit incoming and outgoing dial-peers for Cisco UBE to apply the appropriate treatment to calls (for example, translations, codec, DTMF-type, SIP Normalization, and so on).
- Configure VoIP dial-peers with appropriate descriptions. For example:

- description *** dial-peer to Service Provider ***
- description *** dial-peer to Publisher Cisco Unified CM ***
- description *** dial-peer to Subscriber Cisco Unified CM ***
- Always use a keepalive mechanism, such as Out of Dialog OPTIONS-ping, over the SIP trunk to detect upstream entity failure before routing calls to the service provider.
- Configure the Cisco UBE for media inactivity based on RTP, or RTCP, or both to accelerate the detection of *hung* calls.
- Because it is the most widely deployed and most interoperable DTMF mechanism for SIP trunks, use RFC 2833 to configure DTMF.
- If Cisco UBE is configured to do Delayed Offer to Early Offer conversions, ensure that PRACK is enabled on Cisco Unified CM, for call flows involving early media cut through (18x w/SDP) to work seamlessly.
- Fine tune the failover timers, especially when using clustered/DNS-SRV addressing.
To ensure minimum Post Dial Delay during failover situations, fine tune the **sip-ua retry xxx parameters**, where *xxx* is the request name and response code. We recommend the value for INVITES as *retry invite 2*.
- Do not configure Cisco HSRP on the router that runs Cisco UBE functionality.
The Layer 3 and Layer 7 embedded SIP addresses can be unpredictable when Cisco HSRP is enabled. Refer to the caveats section for exact Bug-ID's.
- Use SIP profiles to insert or remove elements in the SIP headers.
SIP Profiles is a very powerful SIP message normalization and protocol repair tool that can quickly fix or create a workaround to minor interoperability issues when two SIP implementations communicate with each other. This feature is available in Cisco IOS 12.4(15)XZ and Cisco IOS 12.4(20)T and later.
- If SIP trunk capacity requires a stack of Cisco UBEs to scale capacity, consider using the Cisco Unified SIP Proxy and Cisco UBE scaling architecture at the HQ location.
- Pay close attention to DTMF interoperability and call flows.
Adjust the payload types for DTMF as needed when the default Cisco values are in conflict (for example, PT 96 is used for RFC 2833, which is by default reserved for cisco fax-relay).
- Adjust SIP incoming and outgoing ports as required to accommodate send and listen devices on non-standard SIP ports.
- Always test call flows with supplementary services as they present the most likely interoperability issues.
- Configure ACLs on Cisco UBE to allow traffic only from valid call agents and endpoints to avoid toll-fraud.
You can configure CLI commands such as **allow term**.
- Configure fax traffic on TDM PSTN access if at all possible
- Mark all the outbound voice traffic with the appropriate DSCP values so that it gets the right priority in the service provider network. All other traffic should be appropriately marked.
- Provision backup FXO trunks on the Cisco CPE router to provide emergency PSTN access if the SIP trunk is down.
- The service provider should ensure appropriate call routing for emergency (911) calls using the shared hop-off PSTN gateway.

Caveats

In general, the following global caveats exist with this solution:

- The same static codec must be used on all voice calls. It can be any codec type, but the same codec must be maintained.
- The G.711ua codec must be used for the fax/modem calls in the network.
- Headquarter site or remote branch local calls must be configured with G.711 codecs.
- Voice calls over the WAN must be configured with G.729 codecs.
- Video was not tested as part of this solution.
- H.323 calls were not tested as part of this solution.
- Use of Cisco HSRP is not recommended in this solution as it can cause unexpected results with SIP signaling.

Configurations

The “[Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations](#)” section on page 24 provides configuration examples, screen figures, and other helpful information you need to configure the features on the Cisco UBE router at the edge of the service provider network described in this guide.

**Note**

Use the [Command Lookup Tool \(registered customers only\)](#) or the Cisco IOS master commands list at http://www.cisco.com/en/US/docs/ios/mcl/allreleasemcl/all_book.html for more information on the commands used in this guide.

Configuration Verification

Use the following **show** commands to display and verify your Cisco UBE configuration:

- **show dial-peer voice summary**
- **show sip-ua register status**

The firewall configuration can be verified with the following commands:

- **show ip inspect sessions**
- **show ip inspect statistics**

Troubleshooting

**Note**

See [Important Information on Debug Commands](#) before you use **debug** commands.

Use the following **debug** commands to troubleshoot your configuration:

- **debug ccsip messages**

This command shows all SIP Service Provider Interface (SPI) message tracing. It traces the SIP messages exchanged between the SIP UA client (UAC) and the access server.

- **debug ccsip all**

This command enables all SIP-related debugging including:

- **debug voip app**

This command displays all application debug messages, including Application Framework (AFW) and DSAPP debugs.

- **debug voip ccapi inout**

This command traces the execution path through the call control API, which serves as the interface between the call session application and the underlying network-specific software. You can use the output from this command to understand how calls are being handled by the voice gateway.

- **debug ephone mtp**

This command enables Media Termination Point (MTP) debugging.

- **debug sccp events**

This command displays debugging information for SCCP events and its related applications transcoding and conferencing.

Related Information

The following information is referenced in this guide:

- *Cisco Unified Communications Manager Express 4.1 Multi-party Conferencing Enhancements*
- *CDR Logging with Syslog Servers and Cisco IOS Gateways*
- *Cisco 2800 Series Integrated Services Routers*
- *Cisco 3800 Series Integrated Services Routers*
- *Cisco Cable High-Speed WAN Interface Cards*
- *Cisco High Density Analog and Digital Extension Module for Voice and Fax*
- *Cisco IAD243X Business Class Integrated Access Device*
- *Cisco Systems - Support*
- *“Configuring Conferencing” chapter of the Cisco Unified Communications Manager Express System Administrator Guide*
- *Configuring Frame Relay and Frame Relay Traffic Shaping*
- *Configuring SIP Support for Hookflash*
- *Echo Analysis for Voice over IP*
- *Enterprise QoS Solution Reference Network Design Guide*
- *Equivalent MIB Objects for VoIP show Commands*
- *IP Communications Voice/Fax Network Module*
- *Quality of Service for Voice Over IP*
- *RADIUS VSA Voice Implementation Guide*
- *Service Provider Quality-of-Service Overview*
- *Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)*

Obtaining Documentation and Submitting a Service Request

For information on obtaining documentation, submitting a service request, and gathering additional information, see the monthly *What's New in Cisco Product Documentation*, which also lists all new and revised Cisco technical documentation, at:

<http://www.cisco.com/en/US/docs/general/whatsnew/whatsnew.html>

Subscribe to the *What's New in Cisco Product Documentation* as a Really Simple Syndication (RSS) feed and set content to be delivered directly to your desktop using a reader application. The RSS feeds are a free service and Cisco currently supports RSS Version 2.0.

Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

This appendix contains configuration examples to configure a SIP-based managed voice services solution using the Cisco Unified Border Element, Cisco Unified Communications Manager, Cisco Unity, and Cisco Unity Express, depending on your configuration requirements.

- [Overview of Test Configurations, page 24](#)
- [High-Level Operation, page 25](#)
- [Test Topology, page 28](#)
- [Example Configuration Details, page 29](#)
- [Enterprise 1 HQ Cisco UBE Example Configuration, page 29](#)
- [Enterprise 1 HQ Cisco Unified CM Example Configuration, page 32](#)
- [Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration, page 119](#)
- [Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration, page 119](#)
- [Enterprise 1 HQ Cisco ASA Firewall Example Configuration, page 120](#)
- [Branch 1 Cisco UBE, TDM Gateway, and Cisco Unified SRST Example Configuration, page 121](#)
- [Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration, page 125](#)

Overview of Test Configurations

The following main components are used in the Voice Enterprise 1 configuration.

Enterprise 1 HQ Components

The main components of the Enterprise 1 Headquarters (HQ) include:

- Cisco Unified CM (version 6.1)
- SCCP IP Phones
- VG224 (version 12.4(20)T1) analog lines for Fax/Modem support
- Cisco UBE (Cisco IOS Release 12.4(20)T1)

Enterprise 1 and Branch 1 Components

The main components of the Enterprise 1 and Branch 1 include:

- Cisco UBE/Cisco Unified SRST/Analog lines for Fax/Modem
- SCCP IP Phones

Caveats

The following caveats apply to the SIP-based Trunk Voice Enterprise 1 solution:

Global Caveats

In general, the following global caveats exist with this solution:

- The same static codec must be used on all voice calls. It can be any codec type, but the same codec must be maintained.
- The G.711ua codec must be used for the fax/modem calls in the network.
- Headquarter site or remote branch local calls must be configured with G.711 codecs.
- Voice calls over the WAN must be configured with G.729 codecs.
- Video was not tested as part of this solution.
- H.323 calls were not tested as part of this solution.
- Use of Cisco HSRP is not recommended in this solution as it can cause unexpected results with SIP signaling.

Cisco Unified CM 6.1.0.9901-372 Caveats

1. Cisco Unified CM version 6.1 does not support Early Offer g729r8; Delayed Offer is configured on Cisco Unified CM, and Early Offer is enforced on Cisco UBEs.
2. Cisco Unified CM does not support the midcall audio codec change (CSCsr03120).
3. Enhance SIP Trunk display to minimize confusion (CSCsv80045).

High-Level Operation

Anyone trying to configure the Voice Enterprise 1 topology should be very familiar with networking in general and the specific configurations of the following Cisco applications:

- Cisco Unified CM
- Cisco ASA 8.0(4) Firewall
- Cisco Unity
- Cisco Unity Express

Call Flow Within Enterprise 1

All endpoints (Cisco Unified CM, HQ/Branch Cisco UBEs, IP phones, and so on) in the Voice Enterprise 1 network are configured to be routable. Calls within the enterprise use SCCP/MGCP for call control.

During normal operation, call flow from HQ to Branch 1 are as follows:

IP/VG224 FXS Phone (over SCCP) > Cisco Unified CM (over SCCP/MGCP) > IP/Branch Cisco UBE FXS Phone

During normal operation, Branch 1 call flows to HQ is in the reverse direction.

HQ Call Flow to Enterprise Offsite Remote Endpoint

During normal operation, call flow from HQ to outside of the enterprise is as follows:

IP/VG224 FXS phone (over SCCP) > Cisco Unified CM (over SIP) > HQ Cisco UBE (over SIP) > Service Provider SIP Proxy Server

During normal operation, external call flow to the enterprise HQ is in the reverse direction.

Branch 1 Call Flow to Enterprise Offsite Remote Endpoint

Call flow from Branch 1 to outside of the enterprise would be as follows:

IP/Branch Cisco UBE FXS phone (over SCCP/MGCP) > Cisco Unified CM (over SIP) > Branch Cisco UBE (over SIP) > Service Provider SIP Proxy Server

For normal operation, external call flow to the enterprise Branch 1 is in the reverse direction.



Note

Between Cisco Unified CM and Branch Cisco UBE, signaling and voice RTP packets must pass through the enterprise HQ Cisco UBE, and it is not shown in the call flow because it is transparent.

Cisco Unified CM is used to control the number of uplink calls (CAC—bandwidth) for both the enterprise HQ and branch.

For purposes of security, the Cisco ASA can be placed at the front end of the HQ Cisco UBE.

High-Level Configuration Summaries

The following topics summarize the scope of a current enterprise solution.

Protocols

The following is a list of protocols used between components:

- SCCP: Cisco Unified CM and all IP Phones
- SCCP: Cisco Unified CM and Cisco VG224
- MGCP: Cisco Unified CM and Cisco UBE/Cisco Unified SRST TDM
- SIP–SIP: Cisco Unified CM HQ/Branch Cisco UBE and WAN (External to Enterprise)

Codecs

The following is a list of codecs used between components:

- g711ulaw: HQ/Branch IP Phone to IP Phone local calls
- G729r8: HQ/Branch IP Phone to remote endpoint across WAN
- Pass-through g711ulaw: HQ/Branch Fax/Modem to Fax/Modem local calls
- Pass-through g711ulaw:HQ/Branch Fax/Modem to remote endpoint Fax/Modem across WAN

**Note**

Cisco Unified CM (version 6.1) does not support Early Offer g729r8. HQ/Branch Cisco UBEs are therefore configured to overcome this lack of support by using the Early Offer g729r8 for voice calls across the WAN to remote SIP endpoints. Remote voice calls terminating at the enterprise are forced to use g729r8. Cisco UBEs are also configured to force the pass-through of g711ulaw for Fax/Modem calls in both directions.

DSP Farms

Separate DSP farms are installed and configured on the enterprise HQ and Branch Cisco UBEs. Although only conference resources are used for these solutions, MTP and Transcoder resources are also configured and are registered to Cisco Unified CM for example purposes only.

Supplementary Services

The following is a list of supplementary services.

- CALL FORWARD
- CALL TRANSFER—Attended and Blind
- CALL HOLD, MUSIC on HOLD
- HARDWARE CONFERENCING

Call Admission Control

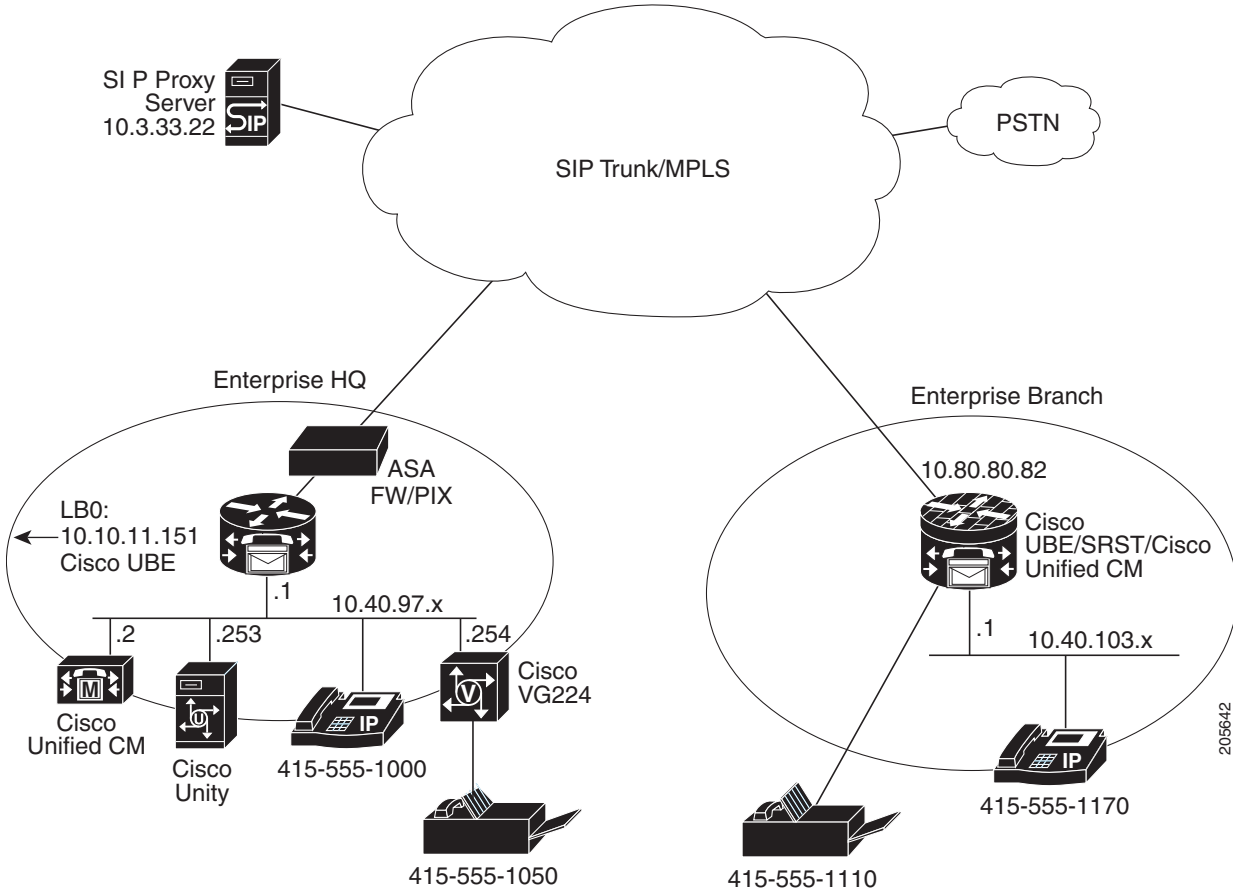
The call admission control (CAC) restrictions that are imposed by Cisco Unified CM for the whole enterprise are as follows:

1. **BANDWIDTH**—With Static Location. Cisco Unified CM restricts max voice and fax/modem calls to configured bandwidth threshold for both enterprise HQ and the Branch uplinks under “Location/Audio calls information.”
2. **NUMBER of CALLS**—The Branch Cisco UBE must be configured to activate when in Cisco Unified SRST mode only, which means that the max-calls/bandwidth threshold should be larger than the setting for Cisco Unified CM. Cisco Unified CM would be the triggering mechanism under normal circumstances.
3. **CPU%**—Cisco UBE at the enterprise HQ and the Branch restrict the maximum voice and fax/modem calls to configured CPU% threshold.
4. **MEMORY**—Cisco UBE at the enterprise HQ and the Branch restrict the maximum voice and fax/modem calls to the configured available memory threshold.

Test Topology

Figure 8 shows the setup test topology used in example configurations described in the following sections.

Figure 8 Test Topology



Example Configuration Details

The IP addresses used with SIP in the network are as follows:

- HQ Cisco UBE: 10.10.11.151
- Cisco Unified CM: 10.40.97.2
- Service Provider SIP Proxy Server: 10.3.33.22
- Br1 Cisco UBE: 10.80.80.82

The selection of the static codec for either a voice or fax call is implemented by tightly integrating the configurations of Cisco Unified CM and site Cisco UBE. For the DO-to-EO to originate from the originator's local Cisco UBE and for the correct codec to be used with the Service Provider SIP proxy server, the following configuration example has been set up:

1. When the enterprise HQ IP Phone initiates the long-distance call pattern 91xxxxxxxxx, through Route Pattern/Location/Partition/Trunk configurations on Cisco Unified CM, SIP INVITE with destination 61xxxxxxxxx is forwarded to the HQ Cisco UBE. A new SIP leg with the destination number 1xxxxxxxxx and codec g729r8 is offered to the service provider's SIP proxy server by the HQ Cisco UBE after translation and forced EO manipulation.
2. When the enterprise HQ FXS phone initiates the long-distance call pattern 91xxxxxxxxx, through Route Pattern/Location/Partition/Trunk configurations on Cisco Unified CM, SIP INVITE with destination 71xxxxxxxxx is forwarded to the HQ Cisco UBE. A new SIP leg with the destination number 1xxxxxxxxx and codec g711u is offered to the service provider's SIP proxy server by the HQ Cisco UBE after translation and forced EO manipulation.
3. When the Branch 1 IP Phone initiates the long-distance call pattern 91xxxxxxxxx, through Route Pattern/Location/Partition/Trunk configurations on Cisco Unified CM, SIP INVITE with destination 61xxxxxxxxx is forwarded to the Branch 1 Cisco UBE. A new SIP leg with the destination number 1xxxxxxxxx and codec g729r8 is offered to the service provider's SIP proxy server by the Branch 1 Cisco UBE after translation and forced EO manipulation.
4. When Branch 1 FXS phone initiates the long-distance call pattern 91xxxxxxxxx, through Route Pattern/Location/Partition/Trunk configurations on Cisco Unified CM, SIP INVITE with destination 71xxxxxxxxx is forwarded to the Branch 1 Cisco UBE. A new SIP leg with the destination number 1xxxxxxxxx and codec g711u is offered to the service provider's SIP proxy server by the Branch 1 Cisco UBE after translation and forced EO manipulation.

Calls terminating at the enterprise are also tightly controlled as to whether they are IP phone (g729r8) or FXS phone (g711u), where the latter is mainly used for fax/modem purposes. Received calls that do not match these criteria are rejected.

The dial-plan for the enterprise HQ and the Branch sites can be any global numbering plan. In the following example, the same area code was used for the enterprise HQ 1 and the Branch 1.

Enterprise 1 HQ Cisco UBE Example Configuration

The following is a command-line interface (CLI) configuration example for the enterprise 1 HQ Cisco Unified Border Element for the test topology described in [Figure 8](#).

```
Ent1_HQ_CUBE1#
!
voice-card 0
 dspfarm
 dsp services dspfarm
!
```

```

voice service voip
address-hiding
allow-connections sip to sip
fax protocol pass-through g711ulaw
modem passthrough nse codec g711ulaw
sip
bind control source-interface Loopback0
bind media source-interface Loopback0
min-se 2000
header-passing error-passthru
options-ping 1200
listen-port non-secure 5090
midcall-signaling passthru
!
voice translation-rule 1
rule 1 /^61/ /1/
rule 2 /^71/ /1/
!
voice translation-profile OUTGOING-SIP-TRK-DIGIT-STRIP
translate called 1
!
!
interface Loopback0
ip address 10.10.11.151 255.255.255.255
!
interface GigabitEthernet0/0
ip address 10.40.97.1 255.255.255.0
duplex full
speed 100
media-type rj45
no keepalive
!
interface GigabitEthernet0/1
ip address 10.40.99.2 255.255.255.0
duplex full
speed 100
media-type rj45
no keepalive
!
ip rtcp report interval 9000
!
sccp local GigabitEthernet0/0
sccp ccm 10.40.97.2 identifier 5 priority 1 version 6.0
sccp
!
sccp ccm group 10
associate ccm 5 priority 1
associate profile 10 register MTP111222333
associate profile 12 register CON111222333
associate profile 11 register XCODE111222333
!
dspfarm profile 11 transcode
codec g711ulaw
codec g729r8
maximum sessions 10
associate application SCCP
!
dspfarm profile 12 conference
description conference bridge
codec g711ulaw
codec g729r8
maximum sessions 10
associate application SCCP
!

```

```
dspfarm profile 10 mtp
  codec g711ulaw
  maximum sessions software 5
  associate application SCCP
!
dial-peer voice 2000 voip
  description *** Voice: LAN to WAN - Incoming Dial-Peer ***
  huntstop
  codec g729r8
  session protocol sipv2
  incoming called-number 6T
  dtmf-relay rtp-nte digit-drop
  no vad
!
dial-peer voice 2001 voip
  description *** Voice: LAN to WAN - Outgoing Dial-Peer ***
  translation-profile outgoing OUTGOING-SIP-TRK-DIGIT-STRIP
  huntstop
  destination-pattern 6T
  codec g729r8
  voice-class sip early-offer forced
  max-redirects 5
  session protocol sipv2
  session target ipv4:10.3.33.22
  dtmf-relay rtp-nte digit-drop
  no vad
!
dial-peer voice 2100 voip
  description *** Voice: WAN to LAN - Incoming Dial-Peer ***
  huntstop
  codec g729r8
  session protocol sipv2
  incoming called-number 415T
  dtmf-relay rtp-nte digit-drop
  no vad
!
dial-peer voice 2101 voip
  description *** Voice: WAN to LAN - Outgoing Dial-Peer ***
  huntstop
  destination-pattern 415T
  codec g729r8
  max-redirects 5
  session protocol sipv2
  session target ipv4:10.40.97.2
  dtmf-relay rtp-nte digit-drop
  no vad
!
dial-peer voice 3000 voip
  description *** Fax: LAN to WAN - Incoming Dial-Peer ***
  huntstop
  session protocol sipv2
  incoming called-number 7T
  dtmf-relay rtp-nte digit-drop
  codec g711ulaw
  no vad
!
dial-peer voice 3001 voip
  description *** Fax: LAN to WAN - Outgoing Dial-Peer ***
  translation-profile outgoing OUTGOING-SIP-TRK-DIGIT-STRIP
  huntstop
  destination-pattern 7T
  voice-class sip early-offer forced
  max-redirects 5
  session protocol sipv2
```

```

session target ipv4:10.3.33.22
dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
!
dial-peer voice 3100 voip
description *** Fax: WAN to LAN - Incoming Dial-Peer ***
huntstop
session protocol sipv2
incoming called-number 415555105[0,1]
dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
!
dial-peer voice 3101 voip
description *** Fax: WAN to LAN - Outgoing Dial-Peer ***
huntstop
destination-pattern 415555105[0,1]
max-redirects 5
session protocol sipv2
session target ipv4:10.40.97.2
dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 180
!
sip-ua
keepalive target ipv4:10.3.33.22
authentication username yyyy password 7 xxxxxxxxxxxx
no remote-party-id
retry invite 2
retry bye 2
retry cancel 2
timers keepalive active 600
reason-header override
g729-annexb override
!
Ent1_HQ_CUBE1#

```

Enterprise 1 HQ Cisco Unified CM Example Configuration

The following example shows the required field and parameter entries for example configuration of the Cisco Unified CM for the topology shown in [Figure 8](#). Parameters are entered using the Cisco Unified CM GUI. The example parameters windows entries are shown in following sections:

- [Configuring the Cisco Unified CM System Parameters, page 33](#)
- [Configuring the Cisco Unified CM Call Routing Parameters, page 63](#)
- [Configuring the Cisco Unified CM Media Resources Parameters, page 78](#)
- [Configuring the Cisco Unified CM Voice Mail Parameters, page 95](#)
- [Configuring the Cisco Unified CM Device Parameters, page 102](#)

Configuring the Cisco Unified CM System Parameters

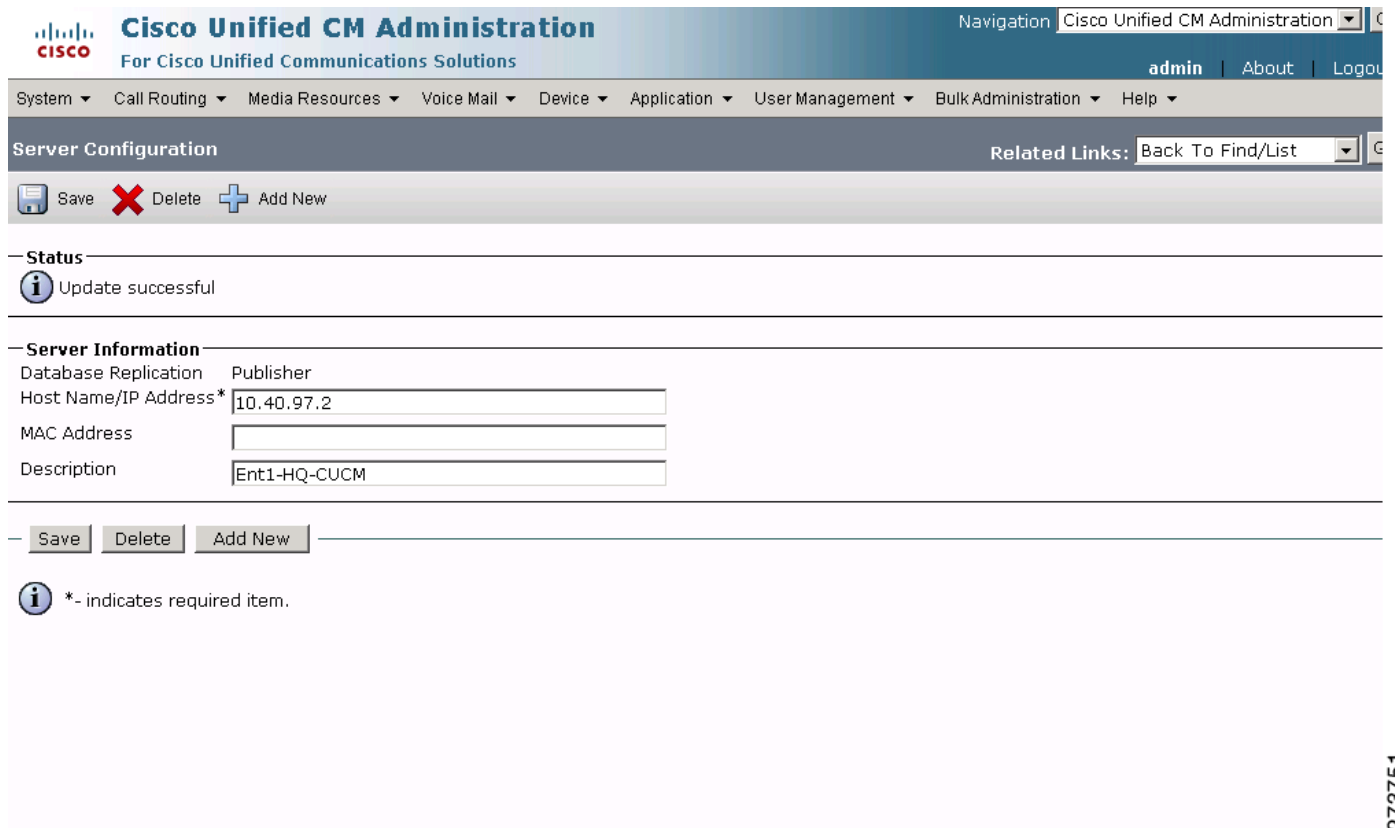
Use the Cisco Unified Communications Manager Administration window to configure system parameters. The system parameter example configurations are shown in the following sections:

- [System: Server Parameters, page 33](#)
- [System: Region Parameters, page 34](#)
- [System: Device Pool Parameters, page 47](#)
- [System: Location Parameters, page 56](#)

System: Server Parameters

To configure the system server parameters for the Cisco Unified CM, click on **System > Server** menu in the Cisco Unified CM Administration window.

Figure 9 System Server Enterprise 1 HQ Cisco Unified CM Administration Window



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Server Configuration Related Links: Back To Find/List

Save Delete Add New

Status
Update successful

Server Information

Database Replication	Publisher
Host Name/IP Address*	10.40.97.2
MAC Address	
Description	Ent1-HQ-CUCM

Save Delete Add New

*- indicates required item.

273751

System: Region Parameters

To configure the system region parameters for the Cisco Unified CM, click **System > Region** menu in the Cisco Unified CM Administration window.

Figure 10 System Region Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for managing regions. The top navigation bar includes the Cisco logo and the text 'Cisco Unified CM Administration For Cisco Unified Communications Solutions'. The main menu shows 'System' selected, leading to 'Find and List Regions'. Below the menu, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'. A status bar indicates '12 records found'. The main table lists 12 regions, each with a checkbox and a name. The regions are: Default, Region Br1 Phones Analog, Region Br1 DSPfarm, Region Br1 DSPfarm Conference, Region Br1 DSPfarm Transcoder, Region Br1 Phones IP, Region HQ DSPfarm, Region HQ DSPfarm Conference, Region HQ DSPfarm Transcoder, Region HQ Phones Analog, Region HQ Phones IP, and Region Wan. The table has a search filter set to 'begins with' and a 'Rows per Page' dropdown set to 50. At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

<input type="checkbox"/>	Name ^
<input type="checkbox"/>	Default
<input type="checkbox"/>	Region Br1 Phones Analog
<input type="checkbox"/>	Region Br1 DSPfarm
<input type="checkbox"/>	Region Br1 DSPfarm Conference
<input type="checkbox"/>	Region Br1 DSPfarm Transcoder
<input type="checkbox"/>	Region Br1 Phones IP
<input type="checkbox"/>	Region HQ DSPfarm
<input type="checkbox"/>	Region HQ DSPfarm Conference
<input type="checkbox"/>	Region HQ DSPfarm Transcoder
<input type="checkbox"/>	Region HQ Phones Analog
<input type="checkbox"/>	Region HQ Phones IP
<input type="checkbox"/>	Region Wan

273752

Figure 11 System Region Default Cisco Unified CM Administration Window

Region Configuration

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.711	384	Use System Default
Region_HQ_Phones_IP	G.729	384	Use System Default
Region_Wan	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region_Br1_Phones_Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New

i *- indicates required item.

i **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273753

Figure 12 System Region-Region Branch 1 Phones Analog Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. At the top, there's a navigation bar with 'Cisco Unified CM Administration' and a user 'admin'. Below that is a menu bar with options like 'System', 'Call Routing', 'Media Resources', etc. The main content area is titled 'Region Configuration' and shows a 'Region Information' section with the name 'Region Br1 Phones Analog'. Below this is a 'Region Relationships' table. The table has columns for 'Region', 'Audio Codec', 'Video Call Bandwidth', and 'Link Loss Type'. The rows show relationships with 'Region_Br1_Phones_IP', 'Region_HQ_Phones_Analog', 'Region_HQ_Phones_IP', 'Region_Wan', and 'Region Br1 Phones Analog', all with 'G.711' audio codec and '384' video call bandwidth. Below the table is a 'NOTE: Regions(s) not displayed' and 'Use System Default' for each column. At the bottom, there's a 'Modify Relationship to other Regions' section with a dropdown menu for 'Regions' (showing 'Default', 'Region Br1 Phones Analog', etc.), a dropdown for 'Audio Codec' (set to 'Keep Current Setting'), radio buttons for 'Video Call Bandwidth' (set to 'Keep Current Setting'), and a dropdown for 'Link Loss Type' (set to 'Keep Current Setting').

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_Phones_IP	G.711	384	Use System Default
Region_HQ_Phones_Analog	G.711	384	Use System Default
Region_HQ_Phones_IP	G.711	384	Use System Default
Region_Wan	G.711	384	Use System Default
Region Br1 Phones Analog	G.711	384	Use System Default

- *- indicates required item.
- **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273754

Figure 13 System Region-Region Branch 1 DSP Farm Cisco Unified CM Administration Window

Region Configuration Related Links: [Back To Find/List](#)

Save ✖ Delete ↺ Reset + Add New

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_DSPfarm	G.729	384	Use System Default
Region_Br1_Phones_IP	G.711	384	Use System Default
Region_HQ_Phones_IP	G.729	384	Use System Default
Region_Wan	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region Br1 Phones Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New



*- indicates required item.



**The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273755

Figure 14 System Region-Region Branch 1 DSP Farm Conference Cisco Unified CM Administration Window

Region Configuration

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Region Configuration | Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_Br1_DSPfarm_Conference

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_Phones_IP	G.711	384	Use System Default
Region_HQ_Phones_IP	G.729	384	Use System Default
Region_Wan	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
Default Region_Br1_Phones_Analog Region_Br1_DSPfarm Region_Br1_DSPfarm_Conference Region_Br1_DSPfarm_Transcoder	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> [] kbps	Keep Current Setting

Save Delete Reset Add New

i *- indicates required item.

i **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273756

Figure 15 System Region-Region Branch 1 DSP Farm Transcoder Cisco Unified CM Administration Window

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_DSPfarm_Transcoder	G.711	384	Use System Default
Region_Br1_Phones_IP	G.711	384	Use System Default
Region_Wan	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region Br1 Phones Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New

i *- indicates required item.

i **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273757

Figure 16 System Region-Region Branch 1 Phones IP Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the user 'admin'. Below the navigation bar is a menu with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Region Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. Below this are buttons for Save, Delete, Reset, and Add New. The 'Region Information' section shows the 'Name*' as 'Region_Br1_Phones_IP'. The 'Region Relationships' section contains a table with columns for Region, Audio Codec, Video Call Bandwidth, and Link Loss Type. Below the table, a note states 'NOTE: Region(s) not displayed' for each column. The 'Modify Relationship to other Regions' section features a 'Regions' dropdown menu, an 'Audio Codec' dropdown set to 'Keep Current Setting', a 'Video Call Bandwidth' section with radio buttons for 'Keep Current Setting', 'Use System Default', 'None', and a text input for 'kbps', and a 'Link Loss Type' dropdown set to 'Keep Current Setting'. At the bottom, there are buttons for Save, Delete, Reset, and Add New, followed by two informational notes.

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_DSPfarm	G.711	384	Use System Default
Region_Br1_DSPfarm_Conference	G.711	384	Use System Default
Region_Br1_DSPfarm_Transcoder	G.711	384	Use System Default
Region_Br1_Phones_IP	G.711	384	Use System Default
Region_HQ_DSPfarm	G.729	384	Use System Default
Region_HQ_DSPfarm_Conference	G.729	384	Use System Default
Region_HQ_Phones_IP	G.729	384	Use System Default
Region_Wan	G.729	384	Use System Default
Region Br1 Phones Analog	G.711	384	Use System Default

NOTE: Region(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region Br1 Phones Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New

i *- indicates required item.

i **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

Figure 17 System Region-Region HQ DSP Farm Cisco Unified CM Administration Window

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_Phones_IP	G.729	384	Use System Default
Region_HQ_DSPfarm	G.729	384	Use System Default
Region_HQ_Phones_IP	G.711	384	Use System Default
Region_Wan	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region Br1 Phones Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New



*- indicates required item.



**The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273759

Figure 18 System Region-Region HQ DSP Farm Conference Cisco Unified CM Administration Window

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_Phones_IP	G.729	384	Use System Default
Region_HQ_Phones_IP	G.711	384	Use System Default
Region_Wan	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region Br1 Phones Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New

- *- indicates required item.
- **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273760

Figure 19 System Region-Region HQ DSP Farm Transcoder Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The page title is "Cisco Unified CM Administration" with the subtitle "For Cisco Unified Communications Solutions". The navigation menu includes "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The current page is "Region Configuration" for "Region_HQ_DSPfarm_Transcoder".

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_HQ_DSPfarm_Transcoder	G.711	384	Use System Default
Region_HQ_Phones_IP	G.711	384	Use System Default
Region_Wan	G.729	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region_Br1_Phones_Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New

i *- indicates required item.

i **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273761

Figure 20 System Region-Region HQ Phones Analog Cisco Unified CM Administration Window

Region Configuration

Save Delete Reset Add New

Region Information

Name* Region_HQ_Phones_Analog

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_HQ_Phones_IP	G.711	384	Use System Default
Region_Wan	G.711	384	Use System Default
Region Br1 Phones Analog	G.711	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
Default Region Br1 Phones Analog Region_Br1_DSPfarm Region_Br1_DSPfarm_Conference Region_Br1_DSPfarm_Transcoder	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="radio"/> [] kbps	Keep Current Setting

Save Delete Reset Add New

- *- indicates required item.
- **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273762

Figure 21 System Region-Region HQ Phones IP Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The page title is "Region Configuration" for "Region_HQ_Phones_IP". The interface includes a navigation menu, a toolbar with "Save", "Delete", "Reset", and "Add New" buttons, and a "Region Information" section with a text input field containing "Region_HQ_Phones_IP".

The "Region Relationships" section contains a table with the following data:

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.729	384	Use System Default
Region_Br1_DSPfarm	G.729	384	Use System Default
Region_Br1_DSPfarm_Conference	G.729	384	Use System Default
Region_Br1_Phones_IP	G.729	384	Use System Default
Region_HQ_DSPfarm	G.711	384	Use System Default
Region_HQ_DSPfarm_Conference	G.711	384	Use System Default
Region_HQ_DSPfarm_Transcoder	G.711	384	Use System Default
Region_HQ_Phones_Analog	G.711	384	Use System Default
Region_HQ_Phones_IP	G.711	384	Use System Default
Region_Wan	G.729	384	Use System Default
Region Br1 Phones Analog	G.711	384	Use System Default

Below the table, a note states: "NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default".

The "Modify Relationship to other Regions" section includes a table with columns for "Regions", "Audio Codec", "Video Call Bandwidth", and "Link Loss Type". The "Regions" column has a dropdown menu with options: Default, Region Br1 Phones Analog, Region_Br1_DSPfarm, Region_Br1_DSPfarm_Conference, and Region_Br1_DSPfarm_Transcoder. The "Audio Codec" column has a dropdown menu with "Keep Current Setting". The "Video Call Bandwidth" column has radio buttons for "Keep Current Setting", "Use System Default", "None", and a text input field for "kbps". The "Link Loss Type" column has a dropdown menu with "Keep Current Setting".

At the bottom of the interface, there are "Save", "Delete", "Reset", and "Add New" buttons.

*- indicates required item.

**The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273763

Figure 22 System Region-Region WAN Cisco Unified CM Administration Window

Region Information

Name*

Region Relationships

Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	G.729	384	Use System Default
Region_Br1_DSPfarm	G.729	384	Use System Default
Region_Br1_DSPfarm_Conference	G.729	384	Use System Default
Region_Br1_DSPfarm_Transcoder	G.729	384	Use System Default
Region_Br1_Phones_IP	G.729	384	Use System Default
Region_HQ_DSPfarm	G.729	384	Use System Default
Region_HQ_DSPfarm_Conference	G.729	384	Use System Default
Region_HQ_DSPfarm_Transcoder	G.729	384	Use System Default
Region_HQ_Phones_Analog	G.711	384	Use System Default
Region_HQ_Phones_IP	G.729	384	Use System Default
Region_Wan	G.729	384	Use System Default
Region Br1 Phones Analog	G.711	384	Use System Default

NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
<input type="text" value="Default"/> <input type="text" value="Region Br1 Phones Analog"/> <input type="text" value="Region_Br1_DSPfarm"/> <input type="text" value="Region_Br1_DSPfarm_Conference"/> <input type="text" value="Region_Br1_DSPfarm_Transcoder"/>	<input type="text" value="Keep Current Setting"/>	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text" value=""/> kbps	<input type="text" value="Keep Current Setting"/>

Save Delete Reset Add New

Footnote:

- *- indicates required item.
- **The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.

273764

System: Device Pool Parameters

To configure the system device pool parameters for the Cisco Unified CM, click **System > Device Pool** menu in the Cisco Unified CM Administration window.

Figure 23 System Device Pool Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. At the top, there is a navigation menu with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. Below this is a section titled 'Find and List Device Pools' with buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'. A status bar indicates '8 records found'. The main area displays a table of device pools with the following columns: Name, Cisco Unified CM Group, Region, Date/Time Group, and Copy. The table lists 8 device pools, all with a 'Default' group and 'CMLocal' date/time group. At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

<input type="checkbox"/>	Name ^	Cisco Unified CM Group	Region	Date/Time Group	Copy
<input type="checkbox"/>	Default	Default	Default	CMLocal	
<input type="checkbox"/>	DevicePool_Br1_Analog_Phones	Default	Region_Br1_Phones_Analog	CMLocal	
<input type="checkbox"/>	DevicePool_Br1_DSPfarm	Default	Region_Br1_DSPfarm	CMLocal	
<input type="checkbox"/>	DevicePool_Br1_IP_Phones	Default	Region_Br1_Phones_IP	CMLocal	
<input type="checkbox"/>	DevicePool_HQ_Analog_Phones	Default	Region_HQ_Phones_Analog	CMLocal	
<input type="checkbox"/>	DevicePool_HQ_DSPfarm	Default	Region_HQ_DSPfarm	CMLocal	
<input type="checkbox"/>	DevicePool_HQ_IP_Phones	Default	Region_HQ_Phones_IP	CMLocal	
<input type="checkbox"/>	DevicePool_WAN	Default	Region_Wan	CMLocal	

273765

Figure 24 System Device Pool Default Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Device Pool Configuration | Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Device Pool Information
Device Pool: Default (3 members**)

Device Pool Settings

Device Pool Name*	Default
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Reverted Call Focus Priority	Default

Roaming Sensitive Settings

Date/Time Group*	CMLocal
Region*	Default
Media Resource Group List	< None >
Location	< None >
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >

Device Mobility Related Information****

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >

Save Delete Copy Reset Add New

*- indicates required item.

**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***leave blank to use default.

****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273766

Figure 25 System Device Pool-DevicePool Branch 1 Analog Phones Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Device Pool Information

Device Pool: DevicePool_Br1_Analog_Phones (2 members**)

Device Pool Settings

Device Pool Name* DevicePool_Br1_Analog_Phones

Cisco Unified Communications Manager Group* Default

Calling Search Space for Auto-registration < None >

Reverted Call Focus Priority Default

Roaming Sensitive Settings

Date/Time Group* CMLocal

Region* Region Br1 Phones Analog

Media Resource Group List Br1 HW MRGL

Location Hub_Br1

Network Locale < None >

SRST Reference* SRST_Ent1_Br1

Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Save Delete Copy Reset Add New

*- indicates required item.

**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***leave blank to use default.

****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273767

Figure 26 System Device Pool-DevicePool Branch 1 DSP Farm Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Device Pool Information

Device Pool: DevicePool_Br1_DSPfarm (3 members)**

Device Pool Settings

Device Pool Name* DevicePool_Br1_DSPfarm

Cisco Unified Communications Manager Group* Default

Calling Search Space for Auto-registration < None >

Reverted Call Focus Priority Default

Roaming Sensitive Settings

Date/Time Group* CMLocal

Region* Region_Br1_DSPfarm

Media Resource Group List Br1 HW MRGL

Location Hub_Br1

Network Locale < None >

SRST Reference* Disable

Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Save Delete Copy Reset Add New

*- indicates required item.

**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***leave blank to use default.

****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273768

Figure 27 System Device Pool-DevicePool Branch 1 IP Phones Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Device Pool Information
Device Pool: DevicePool_Br1_IP_Phones (5 members**)

Device Pool Settings

Device Pool Name*	DevicePool_Br1_IP_Phones
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Reverted Call Focus Priority	Default

Roaming Sensitive Settings

Date/Time Group*	CMLocal
Region*	Region_Br1_Phones_IP
Media Resource Group List	Br1 HW MRGL
Location	Hub_Br1
Network Locale	< None >
SRST Reference*	SRST_Ent1_Br1
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >

Device Mobility Related Information****

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >

Save Delete Copy Reset Add New

*- indicates required item.

**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***leave blank to use default.

****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273769

Figure 28 System Device Pool-DevicePool HQ Analog Phones Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Device Pool Information

Device Pool: DevicePool_HQ_Analog_Phones (3 members**)

Device Pool Settings

Device Pool Name* DevicePool_HQ_Analog_Phones

Cisco Unified Communications Manager Group* Default

Calling Search Space for Auto-registration < None >

Reverted Call Focus Priority Default

Roaming Sensitive Settings

Date/Time Group* CMLocal

Region* Region_HQ_Phones_Analog

Media Resource Group List HQ HW MRGL

Location Hub_HQ

Network Locale < None >

SRST Reference* SRST_Ent1_Br1

Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >





Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Save Delete Copy Reset Add New

-  *- indicates required item.
-  **Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.
-  ***leave blank to use default.
-  ****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273770

Figure 29 System Device Pool-DevicePool HQ DSP Farm Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Device Pool Configuration | Related Links: Back To Find/List

Save | Delete | Copy | Reset | Add New

Status

Status: Ready

Device Pool Information

Device Pool: DevicePool_HQ_DSPfarm (3 members**)

Device Pool Settings

Device Pool Name* DevicePool_HQ_DSPfarm

Cisco Unified Communications Manager Group* Default

Calling Search Space for Auto-registration < None >

Reverted Call Focus Priority Default

Roaming Sensitive Settings

Date/Time Group* CMLocal

Region* Region_HQ_DSPfarm

Media Resource Group List HQ HW MRGL

Location Hub_HQ

Network Locale < None >

SRST Reference* Disable

Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Save | Delete | Copy | Reset | Add New

*- indicates required item.

**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***leave blank to use default.

****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273771

Figure 30 System Device Pool-DevicePool HQ IP Phones Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Device Pool Configuration | Related Links: Back To Find/List

Save | Delete | Copy | Reset | Add New

Status
Status: Ready

Device Pool Information
Device Pool: DevicePool_HQ_IP_Phones (12 members**)

Device Pool Settings

Device Pool Name*	DevicePool_HQ_IP_Phones
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Reverted Call Focus Priority	Default

Roaming Sensitive Settings

Date/Time Group*	CMLocal
Region*	Region_HQ_Phones_IP
Media Resource Group List	HQ HW MRGL
Location	Hub_HQ
Network Locale	< None >
SRST Reference*	SRST_Ent1_Br1
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >

Device Mobility Related Information****

Device Mobility Calling Search Space	< None >
AAR Calling Search Space	< None >
AAR Group	< None >

Save | Delete | Copy | Reset | Add New

- *- indicates required item.
- **Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.
- ***leave blank to use default.
- ****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273772

Figure 31 System DevicePool-DevicePool WAN Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Device Pool Information

Device Pool: DevicePool_WAN (2 members**)

Device Pool Settings

Device Pool Name* DevicePool_WAN

Cisco Unified Communications Manager Group* Default

Calling Search Space for Auto-registration < None >

Reverted Call Focus Priority Default

Roaming Sensitive Settings

Date/Time Group* CMLocal

Region* Region_Wan

Media Resource Group List HQ HW MRGL

Location Hub_HQ

Network Locale < None >

SRST Reference* SRST_Ent1_Br1

Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Save Delete Copy Reset Add New

*- indicates required item.

**Number of devices that have to be reset when this device pool is updated. To see a detailed list of these devices and other dependencies, click on Dependency Records.

***leave blank to use default.

****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273773

System: Location Parameters

To configure the system location parameters for the Cisco Unified CM, click **System > Location** menu in the Cisco Unified CM Administration window.

Figure 32 System Location Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface for the 'System Location' configuration. The page title is 'Cisco Unified CM Administration' and the user is logged in as 'admin'. The navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Find and List Locations' and includes a search bar and action buttons: Add New, Select All, Clear All, and Delete Selected. Below the search bar, the status indicates '5 records found'. The table below shows the following data:

Location	Audio Bandwidth	Video Bandwidth	Copy
Hub_Br1	85	NONE	
Hub_HQ	110	NONE	
Hub_None	UNLIMITED	UNLIMITED	
Trunk_Br1	85	NONE	
Trunk_HQ	110	NONE	

At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

273774

Figure 33 System Location Hub Branch 1 Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration web interface. At the top, there is a navigation bar with the Cisco logo and the text "Cisco Unified CM Administration For Cisco Unified Communications Solutions". The current page is "System Location Configuration".

Status: Status: Ready

Location Information: Name* Hub_Br1

Audio Calls Information: Audio Bandwidth* Unlimited 85 kbps
 If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

Video Calls Information: Video Bandwidth* None Unlimited kbps

Location RSVP Settings:

Location	RSVP Setting
NOTE: Location(s) not displayed	Use System Default

Modify Setting(s) to Other Locations:

Location	RSVP Setting
<ul style="list-style-type: none"> Hub_Br1 Hub_HQ Hub_None Trunk_Br1 Trunk_HQ 	Use System Default

Buttons: Save, Delete, Copy, Add New, Resync Bandwidth

Legend: *- indicates required item.

273775

Figure 34 System Location Hub HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Location Configuration Related Links: Back To Find/List ▾

Save Delete Copy Add New Resync Bandwidth

Status

Status: Ready

Location Information

Name* Hub_HQ

Audio Calls Information

Audio Bandwidth* Unlimited 110 kbps

If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

Video Calls Information

Video Bandwidth* None Unlimited kbps

Location RSVP Settings

Location	RSVP Setting
NOTE: Location(s) not displayed	Use System Default

Modify Setting(s) to Other Locations

Location	RSVP Setting
Hub_Br1 Hub_HQ Hub_None Trunk Br1 Trunk HQ	Use System Default ▾

Save Delete Copy Add New Resync Bandwidth

i *- indicates required item.

273776

Figure 35 System Location Hub None Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration web interface. The page title is "Cisco Unified CM Administration" with the subtitle "For Cisco Unified Communications Solutions". The navigation menu includes "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The current page is "Location Configuration" for the location "Hub_None".

Status: Status: Ready

Location Information: Name* Hub_None

Audio Calls Information: Audio Bandwidth* Unlimited [] kbps
If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

Video Calls Information: Video Bandwidth* None Unlimited [] kbps

Location RSVP Settings:

Location	RSVP Setting
NOTE: Location(s) not displayed	Use System Default

Modify Setting(s) to Other Locations:

Location	RSVP Setting
<ul style="list-style-type: none"> Hub_Br1 Hub_HQ Hub_None Trunk_Br1 Trunk_HQ 	Use System Default

Buttons: Save, Copy, Add New, Resync Bandwidth

i *- indicates required item.

273777

Figure 36 System Location-Location Trunk Branch 1 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Location Configuration | Related Links: Back To Find/List

Save | Delete | Copy | Add New | Resync Bandwidth

Status
Status: Ready

Location Information
Name* Trunk Br1

Audio Calls Information
Audio Bandwidth* Unlimited 85 kbps
If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

Video Calls Information
Video Bandwidth* None Unlimited kbps

Location RSVP Settings

Location	RSVP Setting
NOTE: Location(s) not displayed	Use System Default

Modify Setting(s) to Other Locations

Location	RSVP Setting
<ul style="list-style-type: none"> Hub_Br1 Hub_HQ Hub_None Trunk Br1 Trunk HQ 	Use System Default

Save | Delete | Copy | Add New | Resync Bandwidth

*- indicates required item.

273778

Figure 37 System Location-Location Trunk HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Location Configuration Related Links: Back To Find/List

Save Delete Copy Add New Resync Bandwidth

Status
Status: Ready

Location Information
Name* Trunk HQ

Audio Calls Information
Audio Bandwidth* Unlimited 110 kbps
If the audio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps.

Video Calls Information
Video Bandwidth* None Unlimited kbps

Location RSVP Settings

Location	RSVP Setting
NOTE: Location(s) not displayed	Use System Default

Modify Setting(s) to Other Locations

Location	RSVP Setting
Hub_Br1 Hub_HQ Hub_None Trunk_Br1 Trunk_HQ	Use System Default

Save Delete Copy Add New Resync Bandwidth

*- indicates required item.

273779

System: SRST Parameters

To configure the system SRST parameters for the Cisco Unified CM, click **System > SRST** menu in the Cisco Unified CM Administration window.

Figure 38 System SRST-SRST Enterprise 1 Branch 1 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

SRST Reference Configuration Related Links: Back To Find/List ▾

Save Delete Copy Reset Add New

Status
Status: Ready

SRST Reference Status
SRST Reference: SRST_Ent1_Br1 (used by 13 devices)

SRST Reference Information

Name*	SRST_Ent1_Br1
Port*	2000
IP Address*	10.40.103.1
SIP Network/IP Address	
SIP Port*	5060
SRST Certificate Provider Port*	2445
<input type="checkbox"/> Is SRST Secure?	

Save Delete Copy Reset Add New

i *- indicates required item.

273780

Configuring the Cisco Unified CM Call Routing Parameters

Use the Cisco Unified Communications Manager Administration window to configure call routing parameters. Call routing parameter example configurations are shown in the following sections:

- [Call Routing: Route/Hunt Parameters, page 63](#)
- [Call Routing: Class of Control Parameters, page 68](#)

Call Routing: Route/Hunt Parameters

To configure call routing route/hunt parameters for the Cisco Unified CM, click **Call Routing > Route/Hunt** menu in the Cisco Unified CM Administration window.

Figure 39 Call Routing Route/Hunt Route Pattern Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The main content area is titled 'Find and List Route Patterns'. Below the title are controls for adding, selecting, clearing, and deleting records. A status bar indicates '4 records found'. The main table is titled 'Route Patterns (1 - 4 of 4)' and has a 'Rows per Page' dropdown set to 50. The table has the following data:

Pattern	Description	Partition	Route Filter	Associated Device	Copy
9.1XXXXXXXXXX	RP Ent1-HQ IP Phone LongDistance	Partition-HQ_Phones_IP		10.10.11.151	📄
9.1XXXXXXXXXX	RP Ent1-HQ Analog Phone LongDistance	Partition-HQ_Phones_Analog		10.10.11.151	📄
9.1XXXXXXXXXX	RP Ent1-Br1 Analog Phone LongDistance	Partition-Br1_Phones_Analog		10.80.80.82	📄
9.1XXXXXXXXXX	RP Ent1-Br1 IP Phone LongDistance	Partition-Br1_Phones_IP		10.80.80.82	📄

At the bottom of the table are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

273703

Figure 40 Call Routing RouteHunt Route Pattern RP Ent 1 HQ IP Phone LongDistance Cisco Unified CM Admin Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition

Route Pattern* 9.1XXXXXXXXXX

Route Partition Partition-HQ_Phones_IP

Description RP Ent1-HQ IP Phone LongDistance

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Gateway/Route List* 10.10.11.151 (Edit)

Route Option
 Route this pattern
 Block this pattern No Error

Call Classification* OffNet

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 PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls) 6

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

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

Save Delete Copy Add New

i *- indicates required item.

Figure 41 Call Routing RouteHunt Route Pattern RP Ent1 HQ Analog Phone LongDistance Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

 Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Links: Back To Find/List

Save
 Delete
 Copy
 Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override
 Provide Outside Dial Tone
 Allow Overlap Sending
 Urgent Priority

Require Forced Authorization Code

Authorization Level*

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*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text"/>

*- indicates required item.

273705

Figure 42 Call Routing RouteHunt Route Pattern RP Ent1 Br1 Analog Phone LongDistance Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition

Route Pattern* 9.1XXXXXXXXXX

Route Partition Partition-Br1_Phones_Analog

Description RP Ent1-Br1 Analog Phone LongDistance

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Gateway/Route List* 10.80.80.82 (Edit)

Route Option
 Route this pattern
 Block this pattern No Error

Call Classification* OffNet

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 41555XXXX

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 PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls) 7

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

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

Save Delete Copy Add New

i *- indicates required item.

273706

Figure 43 Call Routing RouteHunt Route Pattern RP Ent1 Br1 IP Phone LongDistance Administration Window

Cisco Unified CM Administration
Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern*

Route Partition

Description

Numbering Plan

Route Filter

MLPP Precedence*

Gateway/Route List* [\(Edit\)](#)

Route Option

Route this pattern

Block this pattern

Call Classification*

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

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*

Calling Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input type="text" value="-- Not Selected --"/>	<input type="text" value="< Not Exist >"/>	<input type="text" value=""/>

Save

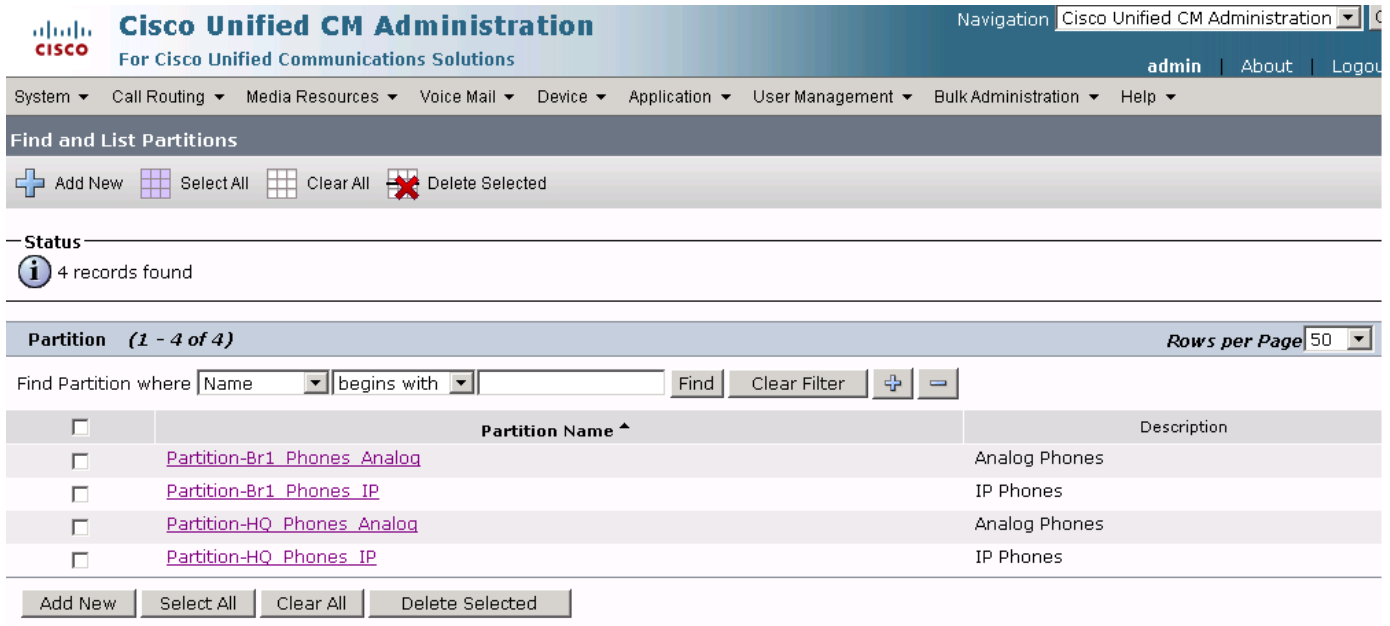
*- indicates required item.

273707

Call Routing: Class of Control Parameters

To configure the call routing class of control parameters for the Cisco Unified CM, click on **Call Routing > Class of Control** menu in the Cisco Unified CM Administration window.

Figure 44 Call Routing Class of Control Partition Cisco Unified CM Administration Window



The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Call Routing' menu is expanded, showing 'Find and List Partitions'. Below this, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'. A status bar indicates '4 records found'. The main table displays the following data:

Partition	(1 - 4 of 4)	Rows per Page	50
Find Partition where	Name	begins with	
<input type="checkbox"/>	Partition-Br1_Phones_Analog	Analog Phones	
<input type="checkbox"/>	Partition-Br1_Phones_IP	IP Phones	
<input type="checkbox"/>	Partition-HQ_Phones_Analog	Analog Phones	
<input type="checkbox"/>	Partition-HQ_Phones_IP	IP Phones	

At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', and 'Delete Selected'.

273708

Figure 45 Call Routing Class of Control Partition-Partition Br1 Phones Analog Administration Window

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation dropdown menu. Below the navigation bar is a breadcrumb trail: System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > Help. The main content area is titled 'Partition Configuration' and includes a 'Related Links' dropdown menu set to 'Back To Find/List'. Below the title bar are action buttons: Save, Delete, Reset, and Add New. The 'Status' section shows 'Status: Ready'. The 'Partition Information' section contains the following fields: Name* (Partition-Br1_Phones_Analog), Description (Analog Phones), Time Schedule (< None >), and Time Zone (radio buttons for 'Originating Device' and 'Specific Time Zone' with 'Greenwich Standard Time' selected). At the bottom of the form are buttons for Save, Delete, Reset, and Add New. A note at the bottom left states: '*- indicates required item.'

273709

Figure 46 Call Routing Class of Control Partition-Partition Br1 Phones IP Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the text 'Cisco Unified CM Administration', and a user profile 'admin'. Below this is a menu bar with options: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Partition Configuration' and includes a 'Related Links' section with a 'Back To Find/List' button. Action buttons for 'Save', 'Delete', 'Reset', and 'Add New' are visible. The configuration form is divided into sections: 'Status' (Ready) and 'Partition Information'. The 'Partition Information' section contains fields for Name, Description, Time Schedule, and Time Zone. The 'Name' field is marked with an asterisk and contains 'Partition-Br1_Phones_IP'. The 'Description' field contains 'IP Phones'. The 'Time Schedule' dropdown is set to '< None >'. The 'Time Zone' section has two radio buttons: 'Originating Device' (selected) and 'Specific Time Zone' (set to 'Greenwich Standard Time'). At the bottom, there are 'Save', 'Delete', 'Reset', and 'Add New' buttons, and a note: '*- indicates required item.'

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Partition Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Partition Information

Name* Partition-Br1_Phones_IP

Description IP Phones

Time Schedule < None >

Time Zone
 Originating Device
 Specific Time Zone Greenwich Standard Time

Save Delete Reset Add New

*- indicates required item.

273710

Figure 47 Call Routing Class of Control Partition-Partition HQ Phones Analog Administration Window

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the subtitle 'For Cisco Unified Communications Solutions'. The user is logged in as 'admin'. The main menu contains 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is 'Partition Configuration', with a 'Related Links' dropdown set to 'Back To Find/List'. Below the navigation is a toolbar with 'Save', 'Delete', 'Reset', and 'Add New' buttons. The 'Status' section shows 'Status: Ready'. The 'Partition Information' section contains the following fields: 'Name*' (Partition-HQ_Phones_Analog), 'Description' (Analog Phones), 'Time Schedule' (< None >), and 'Time Zone' (radio buttons for 'Originating Device' and 'Specific Time Zone' (Greenwich Standard Time)). A second toolbar at the bottom of the form also contains 'Save', 'Delete', 'Reset', and 'Add New' buttons. A note at the bottom left states '*- indicates required item.'

Save **Delete** **Reset** **Add New**

Status

Status: Ready

Partition Information

Name* Partition-HQ_Phones_Analog

Description Analog Phones

Time Schedule < None >

Time Zone Originating Device Specific Time Zone Greenwich Standard Time

Save **Delete** **Reset** **Add New**

*- indicates required item.

273711

Figure 48 Call Routing Class of Control Partition-Partition HQ Phones IP Cisco Unified CM Administration Window

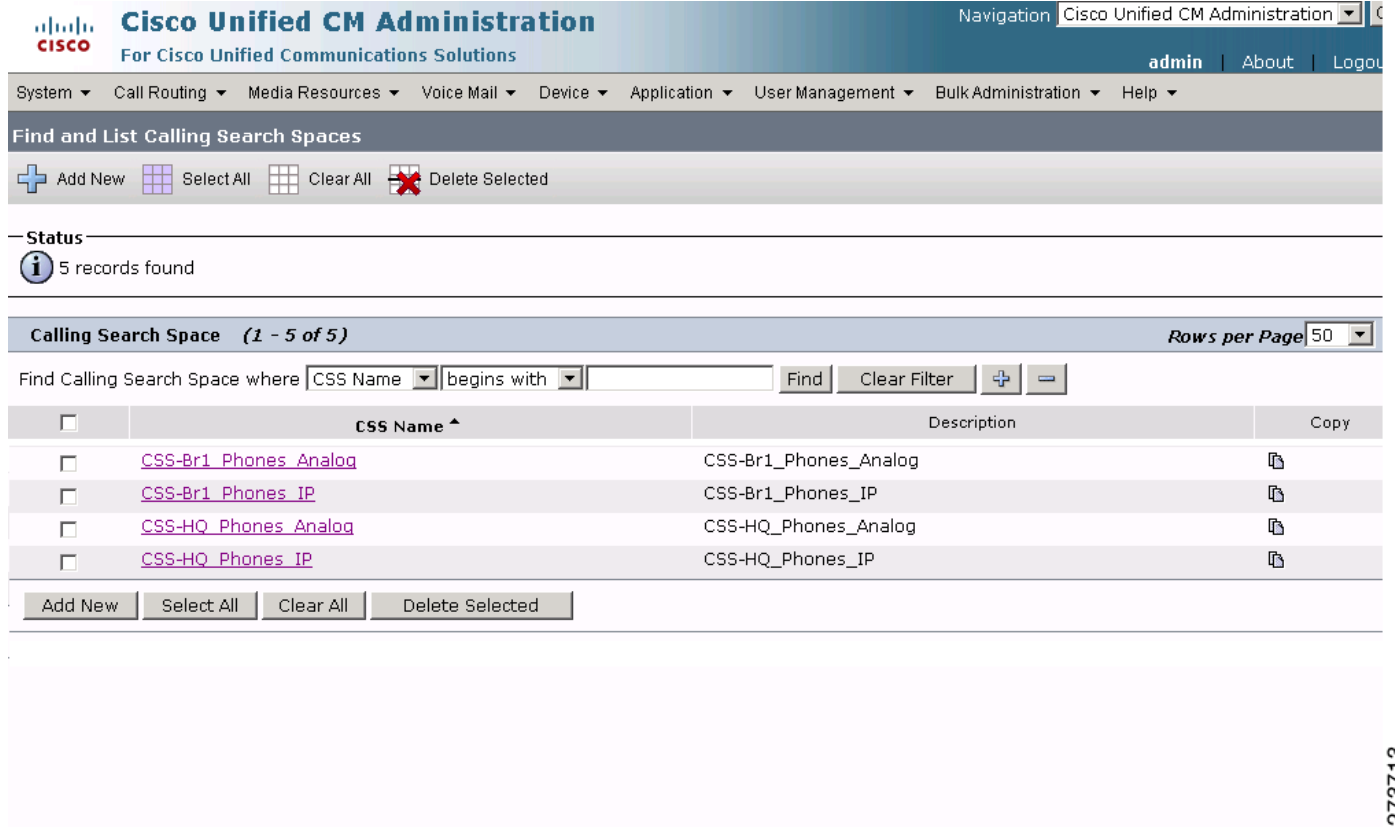
The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the text 'Cisco Unified CM Administration', and a user profile 'admin'. Below this is a menu with options: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Partition Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. Action buttons for 'Save', 'Delete', 'Reset', and 'Add New' are visible. The configuration details are as follows:

- Status:** Ready
- Partition Information:**
 - Name*: Partition-HQ_Phones_IP
 - Description: IP Phones
 - Time Schedule: < None >
 - Time Zone:
 - Originating Device
 - Specific Time Zone: Greenwich Standard Time

At the bottom, there are buttons for 'Save', 'Delete', 'Reset', and 'Add New'. A note indicates that an asterisk (*) denotes a required item.

273712

Figure 49 Call Routing Class of Control CSS Cisco Unified CM Administration Window



The screenshot shows the Cisco Unified CM Administration interface. The main heading is "Cisco Unified CM Administration" with the tagline "For Cisco Unified Communications Solutions". The navigation menu includes "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The current page is "Find and List Calling Search Spaces".

At the top, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below this, a status bar indicates "5 records found".

The main content area is titled "Calling Search Space (1 - 5 of 5)" and includes a search filter: "Find Calling Search Space where CSS Name begins with". The search results are displayed in a table with the following columns: "CSS Name", "Description", and "Copy".

<input type="checkbox"/>	CSS Name ^	Description	Copy
<input type="checkbox"/>	CSS-Br1_Phones_Analog	CSS-Br1_Phones_Analog	
<input type="checkbox"/>	CSS-Br1_Phones_IP	CSS-Br1_Phones_IP	
<input type="checkbox"/>	CSS-HQ_Phones_Analog	CSS-HQ_Phones_Analog	
<input type="checkbox"/>	CSS-HQ_Phones_IP	CSS-HQ_Phones_IP	

At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

273713

Figure 50 Call Routing Class of Control CSS-CSS Branch 1 Phones Analog Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for configuring a Calling Search Space (CSS). The page title is "Calling Search Space Configuration" and the status is "Ready". The configuration details are as follows:

- Name***: CSS-Br1_Phones_Analog
- Description**: CSS-Br1_Phones_Analog
- Route Partitions for this Calling Search Space**:
 - Available Partitions****: (Empty list)
 - Selected Partitions**:
 - Partition-Br1_Phones_Analog
 - Partition-Br1_Phones_IP
 - Partition-HQ_Phones_Analog
 - Partition-HQ_Phones_IP

At the bottom, there are buttons for Save, Delete, Copy, and Add New. Informational notes state:

- *- indicates required item.
- **Selected Partitions are ordered by highest priority

273714

Figure 51 Call Routing Class of Control CSS-CSS Branch 1 Phones IP Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Calling Search Space Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Status: Ready

Calling Search Space Information
Name* CSS-Br1_Phones_IP
Description CSS-Br1_Phones_IP

Route Partitions for this Calling Search Space
Available Partitions**

Selected Partitions
Partition-Br1_Phones_IP
Partition-Br1_Phones_Analog
Partition-HQ_Phones_Analog
Partition-HQ_Phones_IP

Save Delete Copy Add New

i *- indicates required item.
i **Selected Partitions are ordered by highest priority

273715

Figure 52 Call Routing Class of Control CSS-CSS HQ Phones Analog Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for configuring a Calling Search Space (CSS). The page title is "Calling Search Space Configuration" and the specific configuration is for "CSS-HQ_Phones_Analog".

Status: Ready

Calling Search Space Information:

- Name*: CSS-HQ_Phones_Analog
- Description: CSS-HQ_Phones_Analog

Route Partitions for this Calling Search Space:

- Available Partitions**: (Empty list)
- Selected Partitions:
 - Partition-HQ_Phones_Analog
 - Partition-Br1_Phones_Analog
 - Partition-Br1_Phones_IP
 - Partition-HQ_Phones_IP

At the bottom of the configuration area, there are buttons for Save, Delete, Copy, and Add New. Below the configuration area, there are two informational messages:

- *- indicates required item.
- **Selected Partitions are ordered by highest priority

273716

Figure 53 Call Routing Class of Control CSS-CSS HQ Phones IP Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface. At the top, the navigation bar includes 'Navigation Cisco Unified CM Administration' and user information 'admin | About | Logout'. A main menu contains 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is titled 'Calling Search Space Configuration' with a 'Related Links: Back To Find/List' dropdown. Below the title is a toolbar with 'Save', 'Delete', 'Copy', and 'Add New' icons. The 'Status' section shows 'Status: Ready'. The 'Calling Search Space Information' section has 'Name*' set to 'CSS-HQ_Phones_IP' and 'Description' set to 'CSS-HQ_Phones_IP'. The 'Route Partitions for this Calling Search Space' section shows an empty 'Available Partitions**' list and a 'Selected Partitions' list containing 'Partition-HQ_Phones_IP', 'Partition-Br1_Phones_Analog', 'Partition-Br1_Phones_IP', and 'Partition-HQ_Phones_Analog'. A second toolbar at the bottom contains 'Save', 'Delete', 'Copy', and 'Add New' buttons. Information icons provide details: '*' indicates a required item, and '**Selected Partitions are ordered by highest priority'.

273717

Configuring the Cisco Unified CM Media Resources Parameters

Use the Cisco Unified Communications Manager Administration window to configure the media resources parameters. The media resources parameter example configurations are shown in the following sections:

- [Media Resources: Annunciator Parameters, page 78](#)
- [Media Resources: Conference Bridge Parameters, page 79](#)
- [Media Resources: Media Termination Point Parameters, page 82](#)
- [Media Resources: Music on Hold Server Parameters, page 85](#)
- [Media Resources: Transcoder Parameters, page 86](#)
- [Media Resources: Media Resource Group Parameters, page 89](#)
- [Media Resources: Media Resource Group List Parameters, page 92](#)

Media Resources: Annunciator Parameters

To configure the media resources annunciator parameters for the Cisco Unified CM, click **Media Resources** > **Annunciator** menu in the Cisco Unified CM Administration window.

Figure 54 Media Resources Annunciator ANN 2 Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface for configuring an Annunciator. The page title is "Cisco Unified CM Administration" and the sub-header is "Annunciator Configuration". The navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "admin".

Status: Status: Ready

Device Information:

Registration Registered with Cisco Unified Communications Manager 40.40.97.2

IP Address 10.40.97.2

Server* 10.40.97.2

Name* ANN_2

Description ANN_2_Ent1-HQ-CUCM

Device Pool* DevicePool_HQ_IP_Phones

Location* Hub_HQ

Buttons: Save, Reset

Legend: * - indicates required item.

273734

Media Resources: Conference Bridge Parameters

To configure the media resources conference bridge parameters for the Cisco Unified CM, click **Media Resources > Conference Bridge** menu in the Cisco Unified CM Administration window.

Figure 55 Media Resources Conference Bridges Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The 'Media Resources' menu is expanded, and 'Conference Bridges' is selected. The page title is 'Find and List Conference Bridges'. Below the title, there are buttons for 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected'. A status bar indicates '3 records found'. The main content area shows a table of conference bridges with the following data:

Conference Bridge Name	Description	Device Pool	Status	IP Address	Copy
CFB_2	CFB_2-Ent1-HQ	Default	Registered with 10.40.97.2	10.40.97.2	
CON001AA29DF631	CFB-Ent1-Br1	DevicePool_Br1_DSPfarm	Registered with 10.40.97.2	10.40.103.1	
CON111222333	CFB-Ent1-HQ	DevicePool_HQ_DSPfarm	Registered with 10.40.97.2	10.40.97.1	

At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected'.

273735

Figure 56 Media Resources Conference Bridges CFB Enterprise 1 Branch 1 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Conference Bridge Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Conference Bridge Information
Conference Bridge : CON001AA29DF631 (CFB-Ent1-Br1)
Registration Registered with Cisco Unified Communications Manager 10.40.97.2
IP Address 10.40.103.1

IOS Conference Bridge Info
Conference Bridge Type* Cisco IOS Enhanced Conference Bridge
Conference Bridge Name* CON001AA29DF631
Description CFB-Ent1-Br1
Device Pool* DevicePool_Br1_DSPfarm
Common Device Configuration < None >
Location* Hub_Br1
Device Security Mode* Non Secure Conference Bridge

Save Delete Copy Reset Add New

*- indicates required item.

273736

Figure 57 Media Resources Conference Bridges CFB Enterprise 1 HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Conference Bridge Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Conference Bridge Information

Conference Bridge : CON111222333 (CFB-Ent1-HQ)
 Registration Registered with Cisco Unified Communications Manager 10.40.97.2
 IP Address 10.40.97.1

IOS Conference Bridge Info

Conference Bridge Type* Cisco IOS Enhanced Conference Bridge
 Conference Bridge Name* CON111222333
 Description CFB-Ent1-HQ
 Device Pool* DevicePool_HQ_DSPfarm
 Common Device Configuration < None >
 Location* Hub_HQ
 Device Security Mode* Non Secure Conference Bridge

Save Delete Copy Reset Add New

i *- indicates required item.

273737

Media Resources: Media Termination Point Parameters

To configure the media resources media termination point parameters for the Cisco Unified CM, click **Media Resources > Media Termination Point** menu in the Cisco Unified CM Administration window.

Figure 58 Media Resources Media Termination Point Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The main navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is titled 'Find and List Media Termination Points'. Below the title, there are buttons for 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected'. A status bar indicates '3 records found'. The main content area shows a table of Media Termination Points with the following data:

Name	Description	Device Pool	Status	IP Address	Copy
MTP001AA29DF631	MTP-Ent1-Br1	DevicePool_Br1_DSPfarm	Registered with 10.40.97.2	10.40.103.1	Copy
MTP111222333	MTP-Ent1-HQ	DevicePool_HQ_DSPfarm	Registered with 10.40.97.2	10.40.97.1	Copy
MTP_2	MTP_2-Ent1-HQ	Default	Registered with 10.40.97.2	10.40.97.2	Not Allowed

At the bottom of the table, there are buttons for 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected'.

273738

Figure 59 Media Resources Media Termination Point MTP Enterprise 1 Branch 1 Administration Window

The screenshot shows the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation dropdown menu currently set to "Cisco Unified CM Administration". Below this is a secondary navigation menu with options: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Media Termination Point Configuration" and includes a "Related Links" dropdown menu set to "Back To Find/List".

Below the navigation is a toolbar with icons for Save, Delete, Copy, Reset, and Add New. The "Status" section shows "Status: Ready". The "Media Termination Point Information" section contains the following details:

- Registration: Registered with Cisco Unified Communications Manager 10.40.97.2
- IP Address: 10.40.103.1
- Media Termination Point Type*: Cisco IOS Enhanced Software Media Termination Point
- Media Termination Point Name*:
- Description:
- Device Pool*:

At the bottom of the configuration section, there are buttons for Save, Delete, Copy, Reset, and Add New. A note below the buttons states: "i *- indicates required item."

273739

Figure 60 Media Resources Media Termination Point MTP Enterprise 1 HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Media Termination Point Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Media Termination Point Information

Registration	Registered with Cisco Unified Communications Manager 10.40.97.2
IP Address	10.40.97.1
Media Termination Point Type*	Cisco IOS Enhanced Software Media Termination Point
Media Termination Point Name*	<input type="text" value="MTP111222333"/>
Description	<input type="text" value="MTP-Ent1-HQ"/>
Device Pool*	<input type="text" value="DevicePool_HQ_DSPfarm"/>

Save Delete Copy Reset Add New

*- indicates required item.

273740

Media Resources: Music on Hold Server Parameters

To configure the media resources music on hold server parameters for the Cisco Unified CM, click **Media Resources > Music On Hold Server** menu in the Cisco Unified CM Administration window.

Figure 61 Media Resources Music on Hold Server MOH Enterprise 1 HQ Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface for configuring a Music On Hold (MOH) Server. The page title is "Music On Hold (MOH) Server Configuration". The navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "admin".

Status: Status: Ready

Device Information:

Registration	Registered with Cisco Unified Communications Manager 10.40.97.2
IP Address	10.40.97.2
Host Server*	10.40.97.2
Music On Hold Server Name*	MOH-Ent1
Description	MOH_Ent1-HQ
Device Pool*	Default
Location*	Hub_HQ
Maximum Half Duplex Streams*	250
Maximum Multicast Connections*	30
Fixed Audio Source Device	
Run Flag*	Yes

Multicast Audio Source Information:

Enable Multicast Audio Sources on this MOH Server

Base Multicast IP Address* 0.0.0.0

Base Multicast Port Number* 0 (Even numbers only)

Increment Multicast on* Port Number IP Address

Selected Multicast Audio Sources:

There are no Music On Hold Audio Sources selected for Multicasting. Click Configure Audio Sources in the top right corner of the page to select Multicast Audio Sources.

Buttons: Save, Reset

Footnote: *- indicates required item.

273741

Media Resources: Transcoder Parameters

To configure the media resources transcoder parameters for the Cisco Unified CM, click **Media Resources > Transcoder** menu in the Cisco Unified CM Administration window.

Figure 62 Media Resources Transcoder Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The current page is titled "Find and List Transcoders" and shows 2 records found. The table below lists the transcoders:

Name	Description	Device Pool	Status	IP Address	Copy
XCD001AA29DF631	XCODE-Ent1-Br1	DevicePool_Br1_DSPfarm	Registered with 10.40.97.2	10.40.103.1	
XCODE111222333	XCODE-Ent1-HQ	DevicePool_HQ_DSPfarm	Registered with 10.40.97.2	10.40.97.1	

273742

Figure 63 Media Resources Transcoder XCODE Enterprise 1 Branch 1 Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below this is a secondary menu with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Transcoder Configuration" and includes a toolbar with icons for Save, Delete, Copy, Reset, and Add New. The configuration details are as follows:

- Transcoder Information:**
 - Transcoder: XCD001AA29DF631 (XCODE-Ent1-Br1)
 - Registration: Registered with Cisco Unified Communications Manager 10.40.97.2
 - IP Address: 10.40.103.1
- IOS Transcoder Info:**
 - Transcoder Type*: Cisco IOS Enhanced Media Termination Point
 - Description: XCODE-Ent1-Br1
 - Device Name*: XCD001AA29DF631
 - Device Pool*: DevicePool_Br1_DSPfarm (with a [View Details](#) link)
 - Common Device Configuration: < None > (with a [View Details](#) link)
 - Special Load Information: Leave blank to use default

At the bottom of the configuration area, there are buttons for Save, Delete, Copy, Reset, and Add New. A note below the buttons states: "i *- indicates required item."

273743

Figure 64 Media Resources Transcoder XCODE Enterprise 1 HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Transcoder Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Transcoder Information

Transcoder: XCODE111222333 (XCODE-Ent1-HQ)
 Registration Registered with Cisco Unified Communications Manager 10.40.97.2
 IP Address 10.40.97.1

IOS Transcoder Info

Transcoder Type* Cisco IOS Enhanced Media Termination Point
 Description XCODE-Ent1-HQ
 Device Name* XCODE111222333
 Device Pool* DevicePool_HQ_DSPfarm [View Details](#)
 Common Device Configuration < None > [View Details](#)
 Special Load Information Leave blank to use default

Save Delete Copy Reset Add New

i *- indicates required item.

273744

Media Resources: Media Resource Group Parameters

To configure the media resources media resource group parameters for the Cisco Unified CM, click **Media Resources > Media Resource Group** menu in the Cisco Unified CM Administration window.

Figure 65 Media Resources-Media Resource Group Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for managing Media Resource Groups. The page title is "Find and List Media Resource Groups". The navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The user is logged in as "admin".

The status bar indicates "2 records found". The table below shows the details of the two Media Resource Groups:

	Name ^	Description	Multicast	Copy
<input type="checkbox"/>	Br1_HW_MRG	Ent 1 Br1	false	
<input type="checkbox"/>	HQ_HW_MRG	Ent 1 HQ	false	

At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

273745

Figure 66 Media Resources-Media Resource Group Enterprise 1 Branch 1 Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration web interface. At the top, the navigation bar includes the Cisco logo, the text 'Cisco Unified CM Administration', and a user profile 'admin'. Below this is a menu bar with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Media Resource Group Configuration' and includes a 'Related Links' section with a 'Back To Find/List' button. A toolbar contains icons for Save, Delete, Copy, Reset, and Add New. The configuration details are as follows:

- Status:** Status: Ready
- Media Resource Group Status:** Media Resource Group: Br1_HW_MRG (used by 11 devices)
- Media Resource Group Information:**
 - Name*: Br1_HW_MRG
 - Description: Ent 1 Br1
- Devices for this Group:**
 - Available Media Resources**: ANN_2, CFB_2, CON111222333, MTP111222333, MTP_2
 - Selected Media Resources*: CON001AA29DF631 (CFB), MOH-Ent1 (MOH), MTP001AA29DF631 (MTP), XCD001AA29DF631 (XCODE)
- Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

At the bottom of the configuration area, there is another toolbar with buttons for Save, Delete, Copy, Reset, and Add New.

*- indicates required item.

**Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)

273746

Figure 67 Media Resources-Media Resource Group Enterprise 1 HQ Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group. The page title is "Media Resource Group Configuration" and the group name is "HQ_HW_MRG".

Status: Ready

Media Resource Group Status: Media Resource Group: HQ_HW_MRG (used by 19 devices)

Media Resource Group Information:

- Name*: HQ_HW_MRG
- Description: Ent 1 HQ

Devices for this Group:

- Available Media Resources**: ANN_2, CFB_2, CON001AA29DF631, MTP001AA29DF631, MTP_2
- Selected Media Resources*: CON111222333 (CFB), MOH-Ent1 (MOH), MTP111222333 (MTP), XCODE111222333 (XCODE)

Use Multicast for MOH Audio (If at least one multicast MOH resource is available)

Buttons: Save, Delete, Copy, Reset, Add New

Legend:

- *- indicates required item.
- **Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)

273747

Media Resources: Media Resource Group List Parameters

To configure the media resources media resource group list parameters for the Cisco Unified CM, click **Media Resources > Media Resource Group List** menu in the Cisco Unified CM Administration window.

Figure 68 Media Resources-Media Resource Group List Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The main heading is "Cisco Unified CM Administration" with the subtitle "For Cisco Unified Communications Solutions". The navigation menu includes "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The current page is "Find and List Media Resource Group Lists".

At the top, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below this, a status bar indicates "2 records found".

The main content area shows a table titled "Media Resource Group List (1 - 2 of 2)". The table has columns for "Name" and "Copy". The first row is "Br1 HW MRGL" and the second row is "HQ HW MRGL".

At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

	Name ^	Copy
<input type="checkbox"/>	Br1 HW MRGL	
<input type="checkbox"/>	HQ HW MRGL	

273748

Figure 69 Media Resources-Media Resource Group List Branch 1 HW MRGL Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface for configuring a Media Resource Group List. The page title is "Media Resource Group List Configuration" and the breadcrumb trail is "System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > Help". The "Name*" field is populated with "Br1 HW MRGL". Under "Media Resource Groups for this List", the "Available Media Resource Groups" list contains "HQ_HW_MRG" and the "Selected Media Resource Groups" list contains "Br1_HW_MRG". The "Status" is "Ready" and the "Media Resource Group List Status" is "Media Resource Group List: Br1 HW MRGL (used by 11 devices)".

Navigation: Cisco Unified CM Administration

System: System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Media Resource Group List Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Media Resource Group List Status
Media Resource Group List: Br1 HW MRGL (used by 11 devices)

Media Resource Group List Information
Name* Br1 HW MRGL

Media Resource Groups for this List
Available Media Resource Groups: HQ_HW_MRG
Selected Media Resource Groups: Br1_HW_MRG

Save Delete Copy Reset Add New

i *- indicates required item.

273749

Figure 70 Media Resources-Media Resource Group List HQ HW MRGL Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Media Resource Group List Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Media Resource Group List Status
Media Resource Group List: HQ HW MRGL (used by 19 devices)

Media Resource Group List Information
Name* HQ HW MRGL

Media Resource Groups for this List
Available Media Resource Groups: Br1_HW_MRG

Selected Media Resource Groups: HQ_HW_MRG

Save Delete Copy Reset Add New

*- indicates required item.

273750

Configuring the Cisco Unified CM Voice Mail Parameters

Use the Cisco Unified Communications Manager Administration window to configure the voice mail parameters. The voice mail parameter example configurations are shown in the following sections:

- [Voice Mail: Cisco Voice Mail Port Parameters, page 95](#)
- [Voice Mail: Message Waiting Parameters, page 97](#)
- [Voice Mail: Voice Mail Pilot Parameters, page 100](#)
- [Voice Mail: Voice Mail Profile Parameters, page 101](#)

Voice Mail: Cisco Voice Mail Port Parameters

To configure the voice mail Cisco voice mail port parameters for the Cisco Unified CM, click **Voice Mail** > **Cisco Voice Mail Port** menu in the Cisco Unified CM Administration window.

Figure 71 Voice Mail Cisco Voice Mail Port Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. At the top, there is a navigation bar with 'Cisco Unified CM Administration' and 'admin | About | Logout'. Below this is a menu bar with 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main content area is titled 'Find and List Voice Mail Ports' and includes a toolbar with 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected' buttons. A status bar indicates '5 records found'. Below this is a search bar with 'Find Voice Mail Port where' and a dropdown menu set to 'Device Name'. The search criteria are 'begins with' and an empty text box. A 'Find' button and 'Clear Filter' button are also present. Below the search bar is a table with the following data:

Device Name	Description	Device Pool	Device Security Mode	Calling Search Space	Ext.	Partition	Status	IP Address	Copy
CiscoUM1-VI1	Voicemail for Enterprise1	DevicePool_HQ_IP_Phones	Non Secure Voice Mail Port	CSS-HQ_Phones_IP	1090	Partition-HQ_Phones_IP	Registered with 10.40.97.2	10.40.97.253	
CiscoUM1-VI2	Voicemail for Enterprise1	DevicePool_HQ_IP_Phones	Non Secure Voice Mail Port	CSS-HQ_Phones_IP	1091	Partition-HQ_Phones_IP	Registered with 10.40.97.2	10.40.97.253	
CiscoUM1-VI3	Voicemail for Enterprise1	DevicePool_HQ_IP_Phones	Non Secure Voice Mail Port	CSS-HQ_Phones_IP	1092	Partition-HQ_Phones_IP	Registered with 10.40.97.2	10.40.97.253	
CiscoUM1-VI4	Voicemail for Enterprise1	DevicePool_HQ_IP_Phones	Non Secure Voice Mail Port	CSS-HQ_Phones_IP	1093	Partition-HQ_Phones_IP	Registered with 10.40.97.2	10.40.97.253	
CiscoUM1-VI5	Voicemail for Enterprise1	DevicePool_HQ_IP_Phones	Non Secure Voice Mail Port	CSS-HQ_Phones_IP	1094	Partition-HQ_Phones_IP	Registered with 10.40.97.2	10.40.97.253	

At the bottom of the table, there is another toolbar with 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected' buttons. The page number '273781' is visible on the right side.

Figure 72 Voice Mail-Voice Mail Port CiscoUM1 VI1 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Voice Mail Port Configuration Related Links: [Back To Find/List](#)

Save Delete Copy Reset Add New

Status

Status: Ready

Device Information

Registration Registered with Cisco Unified Communications Manager 10.40.97.2

IP Address 10.40.97.253

Port Name* CiscoUM1-VI1

Description Voicemail for Enterprise1

Device Pool* DevicePool_HQ_IP_Phones

Common Device Configuration < None >

Calling Search Space CSS-HQ_Phones_IP

AAR Calling Search Space < None >

Location* Hub_HQ

Device Security Mode* Non Secure Voice Mail Port

Directory Number Information

Directory Number* 1090

Partition Partition-HQ_Phones_IP

Calling Search Space CSS-HQ_Phones_IP

AAR Group < None >

Internal Caller ID Display VoiceMail

Internal Caller ID Display (ASCII format) VoiceMail

External Number Mask 41555XXXX

Save Delete Copy Reset Add New

i *- indicates required item.

273782

Voice Mail: Message Waiting Parameters

To configure the voice mail message waiting parameters for the Cisco Unified CM, click **Voice Mail > Message Waiting** menu in the Cisco Unified CM Administration window.

Figure 73 Voice Mail Message Waiting Cisco Unified CM Administration Window

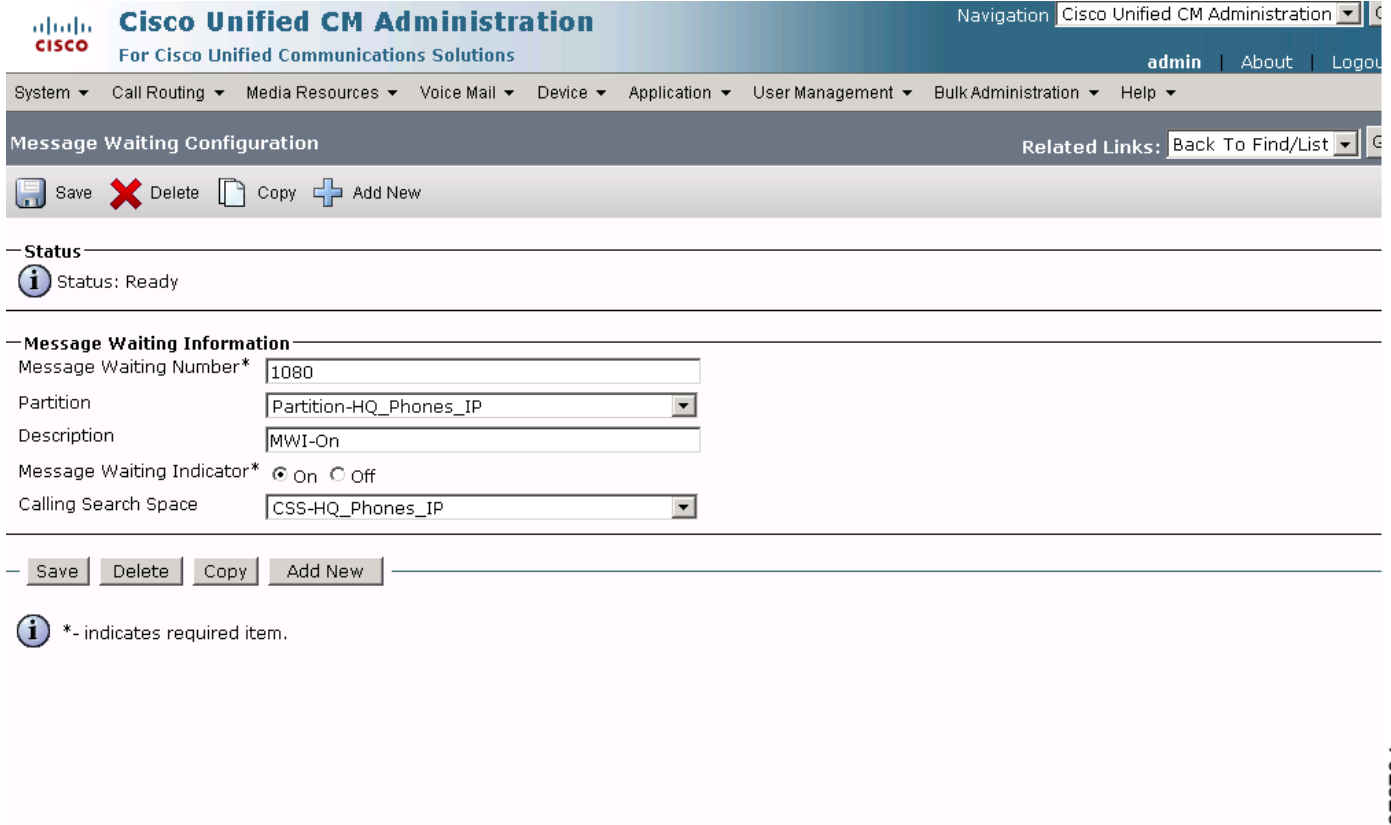
The screenshot shows the Cisco Unified CM Administration interface for configuring Message Waiting Numbers. The page title is "Find and List Message Waiting Numbers". Below the title are action buttons: Add New, Select All, Clear All, and Delete Selected. A status bar indicates "2 records found". The main content area shows a table of Message Waiting Numbers with the following data:

Directory Number	Description	Partition	Calling Search Space	Copy
1080	MWI-On	Partition-HQ_Phones_IP	CSS-HQ_Phones_IP	
1081	MWI-Off	Partition-HQ_Phones_IP	CSS-HQ_Phones_IP	

At the bottom of the table are buttons for Add New, Select All, Clear All, and Delete Selected.

273783

Figure 74 Voice Mail Message Waiting MWI ON Cisco Unified CM Administration Window



Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Message Waiting Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status

Status: Ready

Message Waiting Information

Message Waiting Number*

Partition

Description

Message Waiting Indicator* On Off

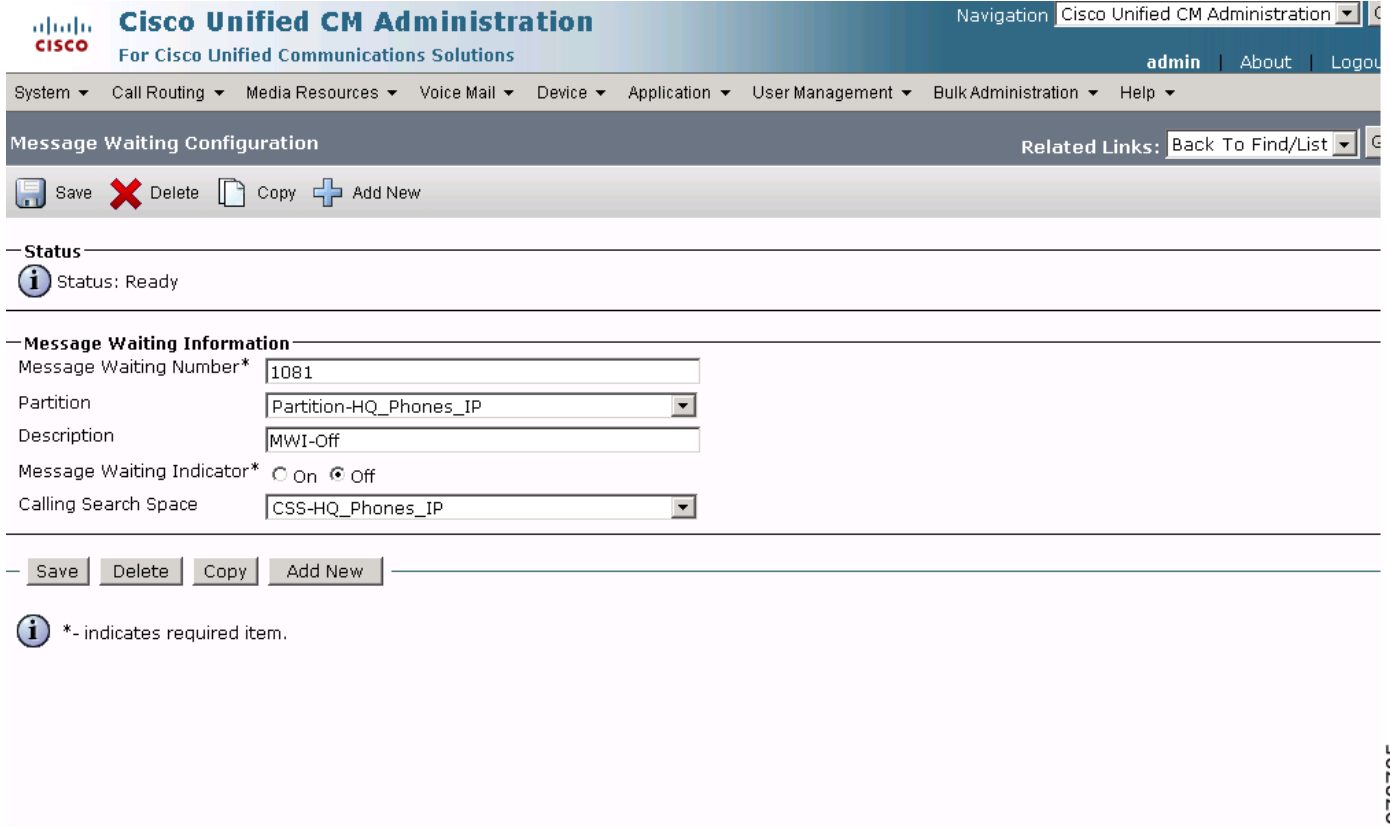
Calling Search Space

Save Delete Copy Add New

*- indicates required item.

273784

Figure 75 Voice Mail Message Waiting MWI Off Cisco Unified CM Administration Window



273785

Voice Mail: Voice Mail Pilot Parameters

To configure the voice mail voice mail pilot parameters for the Cisco Unified CM, click **Voice Mail > Voice Mail Pilot** menu in the Cisco Unified CM Administration window.

Figure 76 Voice Mail-Voice Mail Pilot 1099 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Voice Mail Pilot Configuration Related Links: Back To Find/List ▾

Save **X** Delete **+** Add New

Status

i Status: Ready

Voice Mail Pilot Information

Voice Mail Pilot Number:

Calling Search Space:

Description:

Make this the default Voice Mail Pilot for the system

Save Delete Add New

i *- indicates required item.

273786

Voice Mail: Voice Mail Profile Parameters

To configure the voice mail voice mail profile parameters for the Cisco Unified CM, click **Voice Mail > Voice Mail Profile** menu in the Cisco Unified CM Administration window.

Figure 77 Voice Mail-Voice Mail Profile VM Profile Enterprise 1 HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Voice Mail ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

Voice Mail Profile Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

i Status: Ready

Voice Mail Profile Information

Voice Mail Profile VM-Profile-Ent1-HQ (used by 15 devices)

Voice Mail Profile Name* VM-Profile-Ent1-HQ

Description Default voice messaging profile

Voice Mail Pilot** 1099/CSS-HQ_Phones_IP

Voice Mail Box Mask

Make this the default Voice Mail Profile for the System

Save Delete Copy Reset Add New

i *- indicates required item.

i **- The Voice Mail Pilot is comprised of the Voice Mail Pilot Number and it's corresponding Calling Search Space Name (< Voice Mail Pilot Number >/< Calling Search Space >).

273787

Configuring the Cisco Unified CM Device Parameters

Use the Cisco Unified Communications Manager Administration window to configure the device parameters. The device parameter example configurations are shown in the following sections:

- [Device: Gateway Parameters, page 102](#)
- [Device: Phone Parameters, page 109](#)
- [Device: Trunk Parameters, page 114](#)

Device: Gateway Parameters

To configure the device gateway parameters for the Cisco Unified CM, click **Device > Gateway** menu in the Cisco Unified CM Administration window.

Figure 78 Device Gateway Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for managing gateways. The page includes a navigation menu and a search bar. The main content area shows a table of gateways with the following data:

Device Name	Description	Device Pool	Calling Search Space	Extension	Partition	Route Group	Priority	Port	Device Type	Status	IP Address
Ent1_Br1.Ent1.com	Ent1_Br1								Cisco 3845	See Endpoints	
SKIGWOC863972F5	Ent1-HQ-VG224								VG224	See Endpoints	

273718

Figure 79 Device Gateway Enterprise 1 Branch 1 Enterprise 1.com Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Gateway Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Gateway Details

Product	Cisco 3845
Gateway	Ent1_Br1.Ent1.com
Protocol	MGCP
Domain Name*	Ent1_Br1.Ent1.com
Description	Ent1_Br1
Cisco Unified Communications Manager Group*	Default

Configured Slots, VICs and Endpoints

Module in Slot 0 < None >

Module in Slot 1 < None >

Module in Slot 2 < None >

Module in Slot 3 < None >

Module in Slot 4 NM-HDV2-2PORT-T1

Subunit 0 VIC2-2FXS Begin Port 0 4/0/ 0 POTS 4/0/ 1 POTS

Subunit 1 < None > Begin Port 0

Product Specific Configuration Layout

Global ISDN Switch Type 4ESS ?

Switchback Timing* Graceful

Switchback uptime-delay (min) 10

Switchback schedule (hh:mm) 12:00

Type Of DTMF Relay* Current GW Config

Save Delete Reset Add New

*- indicates required item.

273719

Figure 80 Device Gateway Enterprise 1 Branch 1 Enterprise 1.com pots 1110 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Gateway Configuration | Related Links: Back to MGCP Configuration

Save | Delete | Reset | Add New

Status
Status: Ready

Directory Number Information

- 7718 Line [1] - 1110 in Partition-
- 7719
- Br1_Phones_Analog

Device Information

Product	Cisco MGCP FXS Port
Gateway	Ent1_Br1.Ent1.com
Device Protocol	Analog Access
Registration	Registered with Cisco Unified Communications Manager 10.40.97.2
IP Address	10.40.103.1
End-Point Name *	AALN/S4/SU0/0@Ent1_Br1.Ent1.com
Description	Ent1_Br1_FXS
Device Pool*	DevicePool_Br1_Analog_Phones
Common Device Configuration	< None >
Media Resource Group List	Br1 HW MRGL
Calling Search Space	CSS-Br1_Phones_Analog
AAR Calling Search Space	< None >
Location*	Hub_Br1
AAR Group	< None >
Network Locale	< None >

Transmit UTF-8 for Calling Party Name

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain	< None >
MLPP Indication	Not available on this device
MLPP Preemption	Not available on this device

Port Information (POTS)

Port Direction*	Bothways
Prefix DN	
Num Digits*	4
Expected Digits*	0
SMDI Port Number(0-4096)*	0

Unattended Port

Save | Delete | Reset | Add New

*- indicates required item.

** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

273720

Figure 81 Device Gateway Enterprise 1 Branch 1 Enterprise 1.com pots 1110 Line Administration Window

Directory Number Configuration

Save Delete Reset Add New

Status: Ready

Directory Number Information

Directory Number* 1110
 Route Partition Partbon-Br1_Phones_Analog
 Description 1110
 Alerting Name Ent1_Br1_1110
 ASCII Alerting Name Ent1_Br1_1110
 Associated Devices AALN/54/SU0/0@Ent1_Br1.Ent1.com
 Edit Device Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile <None > (Choose <None> to use system default)
 Calling Search Space CSS-Br1_Phones_Analog
 Presence Group* Standard Presence group
 User Hold MOH Audio Source 1-SampleAudioSource
 Network Hold MOH Audio Source 1-SampleAudioSource

AAR Settings

Voice Mail	AAR Destination Mask	AAR Group
<input type="checkbox"/> or		<None >

Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy		Use System Default
Forward All <input type="checkbox"/> or		<None >
Secondary Calling Search Space for Forward All		<None >
Forward Busy Internal <input type="checkbox"/> or		<None >
Forward Busy External <input type="checkbox"/> or		<None >
Forward No Answer Internal <input type="checkbox"/> or		<None >
Forward No Answer External <input type="checkbox"/> or		<None >
Forward No Coverage Internal <input type="checkbox"/> or		<None >
Forward No Coverage External <input type="checkbox"/> or		<None >
Forward on CTI Failure <input type="checkbox"/> or		<None >
Forward Unregistered Internal <input type="checkbox"/> or		<None >
Forward Unregistered External <input type="checkbox"/> or		<None >
No Answer Ring Duration (seconds)		
Call Pickup Group		<None >

MLPP Alternate Party Settings

Target (Destination)
 MLPP Calling Search Space <None >
 MLPP No Answer Ring Duration (seconds)

Line Settings for All Devices

Hold Reversion Ring Duration (seconds) Setting the Hold Reversion Ring Duration to zero will disable the feature
 Hold Reversion Notification Interval (seconds) Setting the Hold Reversion Notification Interval to zero will disable the feature

Line 1 on Device AALN/54/SU0/0@Ent1_Br1.Ent1.com

Display (Internal Caller ID) Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
 ASCII Display (Internal Caller ID)
 External Phone Number Mask 415371XXXX

Multiple Call/Call Waiting Settings on Device AALN/54/SU0/0@Ent1_Br1.Ent1.com

Note: The range to select the Max Number of calls is: 1-2
 Maximum Number of Calls* 2
 Busy Trigger* 1 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device AALN/54/SU0/0@Ent1_Br1.Ent1.com

Caller Name
 Caller Number
 Redirected Number
 Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Add New

* Indicates required item.
 ** Changes to Line or Directory Number settings require restart.

273721

Figure 82 Device Gateway Enterprise 1 HQ VG224 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Gateway Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status

Status: Ready

Gateway Details

Product: VG224
 Gateway: SKIGW0C863972F5
 Protocol: SCCP
 Mac Address (Last 10 Characters)*: 0C863972F5
 Description: Ent1-HQ-VG224
 Cisco Unified Communications Manager Group*: Default

Configured Slots, VICs and Endpoints

Module in Slot 2: ANALOG

Subunit 0: 24FXS-SCCP

2/0/0	2/0/1	2/0/2	2/0/3	2/0/4	2/0/5
2/0/6	2/0/7	2/0/8	2/0/9	2/0/10	2/0/11
2/0/12	2/0/13	2/0/14	2/0/15	2/0/16	2/0/17
2/0/18	2/0/19	2/0/20	2/0/21	2/0/22	2/0/23

Save Delete Reset Add New

*- indicates required item.

273722

Figure 83 Device Gateway Enterprise 1 HQ VG224 ANA 1050 Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for configuring a phone. The top navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main content area is titled 'Phone Configuration' and includes a 'Related Links' section with a 'Back to Gateway' button. Below this are action buttons: Save, Delete, Copy, Reset, and Add New.

Status: Status: Ready

Association Information

- 1 [Line \[1\] - 1050 in Partition-HQ_Phones_Analog](#)
- Unassigned Associated Items -----
- 2 [Line \[2\] - Add a new DN](#)

Phone Type

Product Type: Analog Phone
Device Protocol: SCCP

Device Information

Registration: Registered with Cisco Unified Communications Manager 10.40.97.2
 IP Address: 10.40.97.254
 MAC Address*: 0C863972F5400
 Description: 415555XXXX
 Device Pool*: DevicePool_HQ_Analog_Phones [View Details](#)
 Common Device Configuration: < None > [View Details](#)
 Phone Button Template*: Standard Analog
 Common Phone Profile*: Standard Common Phone Profile
 Calling Search Space: CSS-HQ_Phones_Analog
 Media Resource Group List: HQ HW MRGL
 Location*: Hub_HQ
 User Locale: < None >
 Network Locale: < None >
 Device Mobility Mode*: Default [View Current](#)
 Owner User ID: < None >

Is Active
 Ignore Presentation Indicators (internal calls only)
 Allow Control of Device from CTI
 Logged Into Hunt Group
 Remote Device

Protocol Specific Information

Presence Group*: Standard Presence group
 Device Security Profile*: Analog Phone - Standard SCCP Non-Secure Pr
 SUBSCRIBE Calling Search Space: < None >
 Unattended Port

MLPP Information

MLPP Domain: < None >
 MLPP Indication*: Default
 MLPP Preemption*: Default

At the bottom of the configuration area are buttons for Save, Delete, Copy, Reset, and Add New.

i *- indicates required item.

i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

i ***Note: Security Profile Contains Addition CAPF Settings.

273723

Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 84 Device-Gateway Enterprise 1 HQ VG224 ANA 1050 Line Cisco Unified CM Administration Window

Cisco Unified CM Administration Navigation: Cisco Unified CM Administration

System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > Help

Directory Number Configuration Related Links: Configure Device (ANOCB63972F5400)

Save Delete Reset Add New

Status: Ready

Directory Number Information

Directory Number* 1050
Route Partition Partition-HQ_Phones_Analog
Description 1050
Alerting Name
ASCII Alerting Name
 Allow Control of Device from CTI
Associated Devices ANOCB63972F5400
Edit Device
Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile < None > (Choose <None> to use system default)
Calling Search Space CSS-HQ_Phones_Analog
Presence Group* Standard Presence group
User Hold MOH Audio Source 1-SampleAudioSource
Network Hold MOH Audio Source 1-SampleAudioSource

AAR Settings

Voice Mail	AAR Destination Mask	AAR Group
AAR <input type="checkbox"/> or		< None >

Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy		Use System Default
Forward All <input type="checkbox"/> or		< None >
Secondary Calling Search Space for Forward All		< None >
Forward Busy Internal <input type="checkbox"/> or		< None >
Forward Busy External <input type="checkbox"/> or		< None >
Forward No Answer Internal <input type="checkbox"/> or		< None >
Forward No Answer External <input type="checkbox"/> or		< None >
Forward No Coverage Internal <input type="checkbox"/> or		< None >
Forward No Coverage External <input type="checkbox"/> or		< None >
Forward on CTI Failure <input type="checkbox"/> or		< None >
Forward Unregistered Internal <input type="checkbox"/> or		< None >
Forward Unregistered External <input type="checkbox"/> or		< None >
No Answer Ring Duration (seconds)		
Call Pickup Group		< None >

MLPP Alternate Party Settings

Target (Destination)
MLPP Calling Search Space < None >
MLPP No Answer Ring Duration (seconds)

Line Settings for All Devices

Hold Reversion Ring Duration (seconds) Setting the Hold Reversion Ring Duration to zero will disable the feature
Hold Reversion Notification Interval (seconds) Setting the Hold Reversion Notification Interval to zero will disable the feature

Line 1 on Device ANOCB63972F5400

Display (Internal Caller ID) Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
ASCII Display (Internal Caller ID)
External Phone Number Mask 41555XXXX
Monitoring Calling Search Space < None >

Multiple Call/Call Waiting Settings on Device ANOCB63972F5400

Note: The range to select the Max Number of calls is: 1-2
Maximum Number of Calls* 1
Busy Trigger* 1 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device ANOCB63972F5400

Caller Name
 Caller Number
 Redirected Number
 Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Add New

* - indicates required item.
** - Changes to Line or Directory Number settings require restart.

273724

Device: Phone Parameters

To configure the device phone parameters for the Cisco Unified CM, click **Device > Phone** menu in the Cisco Unified CM Administration window.

Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 85 Device Phone 415551000 Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for configuring a device phone. The main configuration area is divided into several sections:

- Association Information:** Lists associated items such as Line 1 (Line 1), Line 2 (Line 2), and various call park and forward settings.
- Device Information:** Includes fields for Registration, IP Address, Device Name, Device Pool, Common Device Configuration, Phone Button Template, Common Phone Profile, Calling Search Space, and various call handling options like Call Park, Call Forward, and Call Transfer.
- Product Specific Information:** Contains settings for Packet Capture Mode, Presence Group, Device Security Profile, and Unattended Port.
- Configuration Authority Proxy Function (CAPF) Information:** Includes fields for Certificate Operation, Authentication Mode, and Key Size.
- Extension Module Information:** Lists modules and their associated line names.
- External Data Location Information (Leave blank to use default):** Includes fields for Directory, Messages, Services, and Authentication Server.
- Extension Information:** Includes checkboxes for Enable Extension Mobility and Log Out Profile.
- MFP Information:** Includes fields for MFP Extension, MFP Indication, and MFP Interruption.
- Do Not Disturb:** Includes checkboxes for Do Not Disturb, DND Option, and DND Incoming Call Alert.
- Secure Shell Information:** Includes fields for Secure Shell User and Secure Shell Password.
- Product Specific Configuration Legend:** A large section with numerous checkboxes and dropdown menus for features like Speakerphone, Forwarding Delay, PC Port, Settings Access, and various display and recording options.

273725

* indicates required item.
 ** device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
 ***note: Security Profile Contains Additional CAPF settings.

Figure 86 Device Phone 1000 Cisco Unified CM Administration Window

Directory Number Information

Directory Number* 1000
 Route Partition Partition+HQ_Phones_IP
 Description 1000
 Alerting Name
 ASCII Alerting Name
 Allow Control of Device from CTI
 Associated Devices SEP00187371C3FA

Directory Number Settings

Voice Mail Profile < None >
 Calling Search Space CSS-HQ_Phones_IP
 Presence Group Standard Presence group
 User Hold MOH Audio Source 1-SampleAudioSource
 Network Hold MOH Audio Source 1-SampleAudioSource
 Auto Answer* Auto Answer Off

AAR Settings

AAR or AAR Destination Mask < None > AAR Group < None >
 Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

Calling Search Space Activation Policy	Destination	Calling Search Space
Forward All <input type="checkbox"/> or <input type="checkbox"/>		Use System Default
Secondary Calling Search Space for Forward All		< None >
Forward Busy Internal <input checked="" type="checkbox"/> or <input type="checkbox"/>		< None >
Forward Busy External <input checked="" type="checkbox"/> or <input type="checkbox"/>		CSS-HQ_Phones_IP
Forward No Answer Internal <input checked="" type="checkbox"/> or <input type="checkbox"/>		CSS-HQ_Phones_IP
Forward No Answer External <input checked="" type="checkbox"/> or <input type="checkbox"/>		CSS-HQ_Phones_IP
Forward No Coverage Internal <input checked="" type="checkbox"/> or <input type="checkbox"/>		< None >
Forward No Coverage External <input checked="" type="checkbox"/> or <input type="checkbox"/>		< None >
Forward on CTI Failure <input checked="" type="checkbox"/> or <input type="checkbox"/>		< None >
Forward Unregistered Internal <input checked="" type="checkbox"/> or <input type="checkbox"/>		CSS-HQ_Phones_IP
Forward Unregistered External <input checked="" type="checkbox"/> or <input type="checkbox"/>		CSS-HQ_Phones_IP

No Answer Ring Duration (seconds) 5
 Call Pickup Group < None >

MPP Alternate Party Settings

Target (Destination)
 MPP Calling Search Space < None >
 MPP No Answer Ring Duration (seconds)

Line Settings for All Devices

Hold Reversion Ring Duration (seconds)
 Hold Reversion Notification Interval (seconds)

Line 1 on Device SEP00187371C3FA

Display (Internal Caller ID) Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
 ASCII Display (Internal Caller ID)
 Line Text Label
 ASCII Line Text Label
 External Phone Number Mask #15551XXXX
 Visual Message Waiting Indicator Policy* Use System Policy
 Audible Message Waiting Indicator Policy* Off
 Ring Setting (Phone Idle)* Use System Default
 Ring Setting (Phone Active)* Use System Default Applies to this line when any line on the phone has a call in progress.
 Call Pickup Group Audio Alert Setting (Phone Idle)* Use System Default
 Call Pickup Group Audio Alert Setting (Phone Active)* Use System Default
 Recording Option* Call Recording Disabled
 Recording Profile < None >
 Monitoring Calling Search Space < None >

Multiple Call/Call Waiting Settings on Device SEP00187371C3FA

Note: The range to select the Max Number of calls is: 1-200
 Maximum Number of Calls* 4
 Busy Trigger* 2 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP00187371C3FA

Caller Name
 Caller Number
 Redirected Number
 Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Add New

① * indicates required item.
 ② ** Changes to Line or Directory Number settings require restart.

273726

Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 87 Device Phone 415551170 Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for configuring a device phone. The main content area is divided into several sections:

- Association Information:** Lists various phone services and features, such as Call Forward, Call Park, and Call Transfer, with checkboxes for enabling or disabling them.
- Device Information:** Contains fields for registration details, IP address, MAC address, device name, and various configuration options like Common Device Configuration, Feature Set, and Software Template.
- Protocol Specific Information:** Includes settings for Packet Capture Mode, Presence Group, Device Security Profile, and Unattended Port.
- Certificate Authority Proxy Function (CAPF) Information:** Shows options for Certificate Operation, Authentication Mode, and Key Size.
- Expansion Module Information:** Lists Module 1 and Module 2 load names.
- External Data Location Information:** Provides fields for external data locations like Directory, Messages, and Services.
- Extension Information:** Includes checkboxes for Extension Mobility and fields for Log In and Log Out times.
- MRP Information:** Contains fields for MRP Domain, MRP Indication, and MRP Preemption.
- Do Not Disturb:** Includes checkboxes for Do Not Disturb, DND Option, and DND Incoming Call Alert.
- Device Shell Information:** Includes fields for Secure Shell User and Secure Shell Password.
- Product Specific Configuration List:** A large list of checkboxes for various product-specific features, such as Enable Speakerphone, Forwarding Delay, PC Port, and various discovery protocols.

273727

* Indicates required item.
 ** Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
 ***Note: Security Profile Contains Addition CAPF Settings.

Figure 88 Device Phone 1170 Cisco Unified CM Administration Window

Cisco Unified CM Administration

Navigation: Cisco Unified CM Administration > admin > About > Logout

System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > Help

Directory Number Configuration Related Links: Configure Device (SEP0019E0ABBE78)

Save Delete Reset Add New

Status: Ready

Directory Number Information

Directory Number* 1170

Route Partition Partition-BR1_Phones_IP

Description 1170

Alerting Name

ASCII Alerting Name

Allow Control of Device from CTI

Associated Devices SEP0019E0ABBE78 Edit Device Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile <None> (Choose <None> to use system default)

Calling Search Space CSS-BR1_Phones_IP

Presence Group* Standard Presence group

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

Auto Answer* Auto Answer with Speakerphone

AAR Settings

Voice Mail	AAR Destination Mask	AAR Group
<input type="checkbox"/> or		<None>

Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

Voice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy		Use System Default
Forward All	<input type="checkbox"/> or	<None>
Secondary Calling Search Space for Forward All		<None>
Forward Busy Internal	<input type="checkbox"/> or	<None>
Forward Busy External	<input type="checkbox"/> or	<None>
Forward No Answer Internal	<input type="checkbox"/> or	<None>
Forward No Answer External	<input type="checkbox"/> or	<None>
Forward No Coverage Internal	<input type="checkbox"/> or	<None>
Forward No Coverage External	<input type="checkbox"/> or	<None>
Forward on CTI Failure	<input type="checkbox"/> or	<None>
Forward Unregistered Internal	<input type="checkbox"/> or	<None>
Forward Unregistered External	<input type="checkbox"/> or	<None>
No Answer Ring Duration (seconds)		
Call Pickup Group	<None>	

MLPP Alternate Party Settings

Target (Destination)

MLPP Calling Search Space <None>

MLPP No Answer Ring Duration (seconds)

Line Settings for All Devices

Hold Reversion Ring Duration (seconds) the feature. Setting the Hold Reversion Ring Duration to zero will disable the feature.

Hold Reversion Notification Interval (seconds) the feature. Setting the Hold Reversion Notification Interval to zero will disable the feature.

Line 1 on Device SEP0019E0ABBE78

Display (Internal Caller ID) name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller. Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.

ASCII Display (Internal Caller ID)

Line Text Label

ASCII Line Text Label

External Phone Number Mask FLSSESXXXX

Visual Message Waiting Indicator Policy* Use System Policy

Audible Message Waiting Indicator Policy* Off

Ring Setting (Phone Idle)* Use System Default

Ring Setting (Phone Active) Use System Default Applies to this line when any line on the phone has a call in progress.

Call Pickup Group Use System Default

Audio Alert Setting (Phone Idle) Use System Default

Call Pickup Group Audio Alert Setting (Phone Active) Use System Default

Recording Option* Call Recording Disabled

Recording Profile <None>

Monitoring Calling Search Space <None>

Multiple Call/Call Waiting Settings on Device SEP0019E0ABBE78

Note: The range to select the Max Number of calls is: 1-200

Maximum Number of Calls* 6

Busy Trigger* 6 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP0019E0ABBE78

Caller Name

Caller Number

Redirected Number

Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Add New

* - indicates required item.

** - Changes to Line or Directory Number settings require restart.

273728

Device: Trunk Parameters

To configure the device trunk parameters for the Cisco Unified CM, click **Device > Trunk** menu in the Cisco Unified CM Administration window.

Figure 89 Device Trunk Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The main content area is titled "Find and List Trunks". Below this title is a toolbar with buttons for "Add New", "Select All", "Clear All", "Delete Selected", and "Reset Selected". A status bar indicates "4 records found". Below the status bar is a search section with a dropdown menu for "Find Trunks where" (set to "Device Name"), a "begins with" dropdown, a search input field, and "Find", "Clear Filter", and "Reset" buttons. Below the search section is a table of trunks.

<input type="checkbox"/>	Name ^	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Security Profile
<input type="checkbox"/>	10.10.11.151	Ent1-HQ-CUBE1	CSS-HQ_Phones_IP	DevicePool_WAN	9.1XXXXXXXXXX	Partition-HQ_Phones_Analog			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	10.10.11.151	Ent1-HQ-CUBE1	CSS-HQ_Phones_IP	DevicePool_WAN	9.1XXXXXXXXXX	Partition-HQ_Phones_IP			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	10.80.80.82	Ent1-Br1-CUBE1	CSS-Br1_Phones_IP	DevicePool_WAN	9.1XXXXXXXXXX	Partition-Br1_Phones_IP			SIP Trunk	Non Secure SIP Trunk Profile
<input type="checkbox"/>	10.80.80.82	Ent1-Br1-CUBE1	CSS-Br1_Phones_IP	DevicePool_WAN	9.1XXXXXXXXXX	Partition-Br1_Phones_Analog			SIP Trunk	Non Secure SIP Trunk Profile

At the bottom of the table, there is another toolbar with buttons for "Add New", "Select All", "Clear All", "Delete Selected", and "Reset Selected".

273729

Figure 90 Device Trunk Enterprise 1 HQ CUBE1 Phones Analog Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface for configuring a SIP Trunk. The page is titled "Trunk Configuration" and includes a navigation menu at the top with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. Below the navigation is a toolbar with "Save", "Delete", "Reset", and "Add New" buttons. The main configuration area is divided into several sections:

- Status:** Shows "Status: Ready".
- Device Information:**
 - Product: SIP Trunk
 - Device Protocol: SIP
 - Device Name*: 10.10.11.151
 - Description: Ent1-HQ-CUBE1
 - Device Pool*: DevicePool_WAN
 - Common Device Configuration: < None >
 - Call Classification*: Use System Default
 - Media Resource Group List: HQ HW MRGL
 - Location*: Trunk HQ
 - AAR Group: < None >
 - Packet Capture Mode*: None
 - Packet Capture Duration: 0
 - Options: Media Termination Point Required, Retry Video Call as Audio, Transmit UTF-8 for Calling Party Name, Unattended Port
- Multilevel Precedence and Preemption (MLPP) Information:**
 - MLPP Domain: < None >
- Call Routing Information:**
 - Inbound Calls:**
 - Significant Digits*: 4
 - Connected Line ID Presentation*: Default
 - Connected Name Presentation*: Default
 - Calling Search Space: CSS-HQ_Phones_IP
 - AAR Calling Search Space: < None >
 - Prefix DN:
 - Redirecting Diversion Header Delivery - Inbound
 - Outbound Calls:**
 - Calling Party Selection*: Last Redirect Number (External)
 - Calling Line ID Presentation*: Default
 - Calling Name Presentation*: Default
 - Caller ID DN:
 - Caller Name:
 - Redirecting Diversion Header Delivery - Outbound
- SIP Information:**
 - Destination Address*: 10.10.11.151
 - Destination Address is an SRV
 - Destination Port*: 5090
 - MTP Preferred Originating Codec*: 711ulaw
 - Presence Group*: Standard Presence_group
 - SIP Trunk Security Profile*: Non Secure SIP Trunk Profile
 - Rerouting Calling Search Space: < None >
 - Out-Of-Dialog Refer Calling Search Space: < None >
 - SUBSCRIBE Calling Search Space: < None >
 - SIP Profile*: Standard SIP Profile
 - DTMF Signaling Method*: No Preference

At the bottom of the configuration area, there are buttons for "Save", "Delete", "Reset", and "Add New". A small information icon is visible in the bottom left corner.

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Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 91 Device Trunk Enterprise 1 HQ CUBE1 Phones IP Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Device Information

Product: SIP Trunk
 Device Protocol: SIP
 Device Name*: 10.10.11.151
 Description: Ent1-HQ-CUBE1
 Device Pool*: DevicePool_WAN
 Common Device Configuration: < None >
 Call Classification*: Use System Default
 Media Resource Group List: HQ HW MRGL
 Location*: Trunk HQ
 AAR Group: < None >
 Packet Capture Mode*: None
 Packet Capture Duration: 0

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information
 MLPP Domain: < None >

Call Routing Information

Inbound Calls

Significant Digits*: 4
 Connected Line ID Presentation*: Default
 Connected Name Presentation*: Default
 Calling Search Space: CSS-HQ_Phones_IP
 AAR Calling Search Space: < None >
 Prefix DN:
 Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*: Last Redirect Number (External)
 Calling Line ID Presentation*: Default
 Calling Name Presentation*: Default
 Caller ID DN:
 Caller Name:
 Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*: 10.10.11.151
 Destination Address is an SRV
 Destination Port*: 5090
 MTP Preferred Originating Codec*: 711ulaw
 Presence Group*: Standard Presence group
 SIP Trunk Security Profile*: Non Secure SIP Trunk Profile
 Rerouting Calling Search Space: < None >
 Out-Of-Dialog Refer Calling Search Space: < None >
 SUBSCRIBE Calling Search Space: < None >
 SIP Profile*: Standard SIP Profile
 DTMF Signaling Method*: No Preference

Save Delete Reset Add New

***** - indicates required item.
****** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

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Figure 92 Device Trunk Enterprise 1 Branch 1 CUBE1 Phones Analog Cisco Unified CM Administration Window

Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

System | Call Routing | Media Resources | Voice Mail | Device | Application | User Management | Bulk Administration | Help

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
 Status: Ready

Device Information

Product: SIP Trunk
 Device Protocol: SIP
 Device Name*: 10.80.80.82
 Description: Ent1-Br1-CUBE1
 Device Pool*: DevicePool_WAN
 Common Device Configuration: < None >
 Call Classification*: Use System Default
 Media Resource Group List: Br1 HW MRGL
 Location*: Trunk Br1
 AAR Group: < None >
 Packet Capture Mode*: None
 Packet Capture Duration: 0

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information
 MLPP Domain: < None >

Call Routing Information

Inbound Calls

Significant Digits*: 4
 Connected Line ID Presentation*: Default
 Connected Name Presentation*: Default
 Calling Search Space: CSS-Br1_Phones_IP
 AAR Calling Search Space: < None >
 Prefix DN:
 Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*: Originator
 Calling Line ID Presentation*: Default
 Calling Name Presentation*: Default
 Caller ID DN:
 Caller Name:
 Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*: 10.80.80.82
 Destination Address is an SRV
 Destination Port*: 5060
 MTP Preferred Originating Codec*: 711ulaw
 Presence Group*: Standard Presence_group
 SIP Trunk Security Profile*: Non Secure SIP Trunk Profile
 Rerouting Calling Search Space: < None >
 Out-Of-Dialog Refer Calling Search Space: < None >
 SUBSCRIBE Calling Search Space: < None >
 SIP Profile*: Standard SIP Profile
 DTMF Signaling Method*: No Preference

Save Delete Reset Add New

i *- indicates required item.
i **- Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

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Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 93 Device Trunk Enterprise 1 Branch 1 CUBE1 Phones IP Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin About Logout

System Call Routing Media Resources Voice Mail Device Application User Management Bulk Administration Help

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Device Information

Product: SIP Trunk
 Device Protocol: SIP
 Device Name*: 10.80.80.82
 Description: Ent1-Br1-CUBE1
 Device Pool*: DevicePool_WAN
 Common Device Configuration: < None >
 Call Classification*: Use System Default
 Media Resource Group List: Br1 HW MRGL
 Location*: Trunk Br1
 AAR Group: < None >
 Packet Capture Mode*: None
 Packet Capture Duration: 0

Media Termination Point Required
 Retry Video Call as Audio
 Transmit UTF-8 for Calling Party Name
 Unattended Port

Multilevel Precedence and Preemption (MLPP) Information
 MLPP Domain: < None >

Call Routing Information

Inbound Calls

Significant Digits*: 4
 Connected Line ID Presentation*: Default
 Connected Name Presentation*: Default
 Calling Search Space: CSS-Br1_Phones_IP
 AAR Calling Search Space: < None >
 Prefix DN:

Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*: Originator
 Calling Line ID Presentation*: Default
 Calling Name Presentation*: Default
 Caller ID DN:
 Caller Name:

Redirecting Diversion Header Delivery - Outbound

SIP Information

Destination Address*: 10.80.80.82
 Destination Address is an SRV
 Destination Port*: 5060
 MTP Preferred Originating Codec*: 711ulaw
 Presence Group*: Standard Presence group
 SIP Trunk Security Profile*: Non Secure SIP Trunk Profile
 Rerouting Calling Search Space: < None >
 Out-Of-Dialog Refer Calling Search Space: < None >
 SUBSCRIBE Calling Search Space: < None >
 SIP Profile*: Standard SIP Profile
 DTMF Signaling Method*: No Preference

Save Delete Reset Add New

***** - indicates required item.
****** - Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.

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Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration

To integrate the Cisco Unity version 5.0 with Cisco Unified CM configuration, see the *Cisco Unified Communications Manager SCCP Integration Guide for Cisco Unity Release 5.0*.

Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration

The following is a command-line interface (CLI) configuration example for the enterprise 1 HQ the Cisco VG224 Analog Phone Gateway for the test topology described in [Figure 8](#).

```
Ent1_HQ_VG224#
!
stcapp ccm-group 1
stcapp
!
voice service voip
  fax protocol pass-through g711ulaw
  modem passthrough nse codec g711ulaw
!
interface FastEthernet0/0
  ip address 10.40.97.254 255.255.0.0
  load-interval 30
  duplex full
  speed 100
!
interface FastEthernet0/1
  no ip address
  shutdown
  duplex auto
  speed auto
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 FastEthernet0/0
!
voice-port 2/0
  timeouts ringing infinity
  caller-id enable
!
voice-port 2/1
  timeouts ringing infinity
  caller-id enable
!
sccp local FastEthernet0/0
sccp ccm 10.40.97.2 identifier 10
sccp
!
sccp ccm group 1
  associate ccm 10 priority 1
!
dial-peer voice 1 pots
  service stcapp
  port 2/0
!
dial-peer voice 2 pots
  service stcapp
```

```

port 2/1
!
Ent1_HQ_VG224#

```

Enterprise 1 HQ Cisco ASA Firewall Example Configuration

The following is a command-line interface (CLI) configuration example for the enterprise 1 HQ the Cisco ASA 8.0(4) 5500 Series Adaptive Security Appliances firewall for the test topology described in [Figure 8](#).

```

Ent1-HQ-ASA#
!
interface Vlan65
 nameif inside
 security-level 100
 ip address 10.40.99.1 255.255.255.0
!
interface Vlan70
 nameif outside
 security-level 0
 ip address 10.40.98.2 255.255.255.0
!
interface Ethernet0/0
 description *** To WAN ***
 switchport access vlan 70
!
interface Ethernet0/1
 description *** To LAN ***
 switchport access vlan 65
!
ftp mode passive
access-list 100 extended permit icmp any any
access-list 100 extended permit icmp any any echo
access-list 100 extended permit icmp any any echo-reply
access-list 100 extended permit tcp any host 40.40.97.2 eq 2000
access-list 100 extended permit udp any host 40.40.97.2 eq sip
access-list 100 extended permit tcp any host 40.40.97.2 range h323 h323
access-list 100 extended permit tcp any host 10.10.11.151 eq 5090
access-list 100 extended permit udp any host 10.10.11.151 eq 5090
access-list 100 extended permit tcp any host 40.40.97.2 eq 2428
access-list 100 extended permit udp any host 40.40.97.2 eq 2427
pager lines 24
logging enable
logging buffered debugging
logging asdm informational
mtu inside 1500
mtu outside 1500
icmp unreachable rate-limit 1 burst-size 1
asdm image disk0:/asdm-524.bin
no asdm history enable
arp timeout 14400
access-group 100 in interface outside
!
timeout xlate 3:00:00
timeout conn 1:00:00 half-closed 0:10:00 udp 0:02:00 icmp 0:00:02
timeout sunrpc 0:10:00 h323 0:05:00 h225 1:00:00 mgcp 0:05:00 mgcp-pat 0:05:00
timeout sip 0:30:00 sip_media 0:02:00 sip-invite 0:03:00 sip-disconnect 0:02:00
timeout sip-provisional-media 0:02:00 uauth 0:05:00 absolute
http server enable
no snmp-server location
no snmp-server contact

```



```

snmp-server enable traps snmp authentication linkup linkdown coldstart
telnet timeout 5
ssh timeout 5
console timeout 0
!
class-map sipoutin
  match port udp eq 5090
class-map inspection_default
  match default-inspection-traffic
!
policy-map type inspect dns preset_dns_map
  parameters
    message-length maximum 512
policy-map global_policy
  class inspection_default
    inspect dns preset_dns_map
    inspect ftp
    inspect rsh
    inspect rtsp
    inspect esmtp
    inspect sqlnet
    inspect skinny
    inspect sunrpc
    inspect xdmcp
    inspect sip
    inspect netbios
    inspect tftp
policy-map outsidein
  class sipoutin
    inspect sip
  class inspection_default
    inspect skinny
!
service-policy global_policy interface inside
service-policy outsidein interface outside
prompt hostname context
Cryptochecksum:xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx
: end
Ent1-HQ-ASA#

```

Branch 1 Cisco UBE, TDM Gateway, and Cisco Unified SRST Example Configuration

The following is a command-line interface (CLI) configuration example for the branch 1 Cisco Unified Border Element, TDM Switching in the Cisco AS5000 Gateway, and Cisco Unified SRST for the test topology described in [Figure 8](#).

```

Ent1_Br1#
!
voice-card 4
  dspfarm
  dsp services dspfarm
!
voice service voip
  address-hiding
  allow-connections sip to sip
  no supplementary-service sip moved-temporarily
  no supplementary-service sip refer
  supplementary-service media-renegotiate

```

```

fax protocol pass-through g711ulaw
modem passthrough nse codec g711ulaw
sip
  min-se 90
  header-passing error-passthru
  midcall-signaling passthru
!
voice translation-rule 1
  rule 1 /^61/ /1/
  rule 2 /^71/ /1/
!
voice translation-profile OUTGOING-SIP-TRK-DIGIT-STRIP
  translate called 1
!
interface Loopback0
  ip address 10.10.11.154 255.255.255.255
!
interface GigabitEthernet0/0
  no ip address
  shut
  duplex auto
  speed auto
  media-type rj45
!
interface GigabitEthernet0/1
  description *** To Local LAN ***
  no ip address
  ip virtual-reassembly
  load-interval 30
  duplex auto
  speed auto
  media-type rj45
!
interface GigabitEthernet0/1.1
  encapsulation dot1Q 103
  ip address 10.40.103.1 255.255.255.0
  ip helper-address 10.40.97.2
  ip virtual-reassembly
!
interface Serial4/0:0
  description *** To WAN ***
  ip address 10.80.80.82 255.255.255.252
  ip virtual-reassembly
  encapsulation frame-relay
  load-interval 30
  cdp enable
  frame-relay map ip 10.80.80.81 202
  frame-relay interface-dlci 202
  no frame-relay inverse-arp NOVELL 202
  no frame-relay inverse-arp APPLETALK 202
  no frame-relay inverse-arp DECNET 202
  frame-relay lmi-type ansi
  frame-relay local-dlci 202
!
interface Serial4/0:23
  no ip address
  encapsulation hdlc
  isdn switch-type primary-net5
  isdn incoming-voice voice
  no cdp enable
!
call treatment on
call threshold global cpu-avg low 68 high 75
call threshold global total-mem low 75 high 85

```

```
call threshold global total-calls low 1 high 12
!
!
voice-port 2/1/0
!
voice-port 2/1/1
!
voice-port 4/0/0
!
voice-port 4/0/1
!
voice-port 4/0:23
!
ccm-manager mgcp
!
mgcp
mgcp call-agent 10.40.97.2 2427 service-type mgcp version 0.1
mgcp dtmf-relay voip codec all mode out-of-band
mgcp sdp simple
mgcp fax t38 inhibit
mgcp bind control source-interface GigabitEthernet0/1.1
mgcp bind media source-interface GigabitEthernet0/1.1
!
mgcp profile default
!
sccp local GigabitEthernet0/1.1
sccp ccm 10.40.97.2 identifier 1 priority 1 version 6.0
sccp ip precedence 3
sccp
!
sccp ccm group 1
  bind interface GigabitEthernet0/1.1
  associate ccm 1 priority 1
  associate profile 3 register XCD001AA29DF631
  associate profile 2 register CON001AA29DF631
  associate profile 1 register MTP001AA29DF631
  keepalive retries 1
  keepalive timeout 10
  switchover method immediate
  switchback method immediate
!
dspfarm profile 3 transcode
  description transcode bridge
  codec g711ulaw
  codec g729r8
  maximum sessions 5
  associate application SCCP
!
dspfarm profile 2 conference
  description conference bridge
  codec g711ulaw
  codec g729r8
  maximum sessions 4
  associate application SCCP
!
dspfarm profile 1 mtp
  codec g729r8
  maximum sessions software 5
  associate application SCCP
!
!
dial-peer voice 2000 voip
  description *** Voice: LAN to WAN - Incoming Dial-Peer ***
  huntstop
```

```

    codec g729r8
    session protocol sipv2
    incoming called-number 6T
    dtmf-relay rtp-nte digit-drop
    no vad
!
dial-peer voice 2001 voip
description *** Voice: LAN to WAN - Outgoing Dial-Peer ***
translation-profile outgoing OUTGOING-SIP-TRK-DIGIT-STRIP
huntstop
destination-pattern 6T
codec g729r8
voice-class sip early-offer forced
max-redirects 5
session protocol sipv2
session target ipv4:10.3.33.22
dtmf-relay rtp-nte digit-drop
no vad
!
dial-peer voice 2100 voip
description *** Voice: WAN to LAN - Incoming Dial-Peer ***
huntstop
codec g729r8
session protocol sipv2
incoming called-number 415T
dtmf-relay rtp-nte digit-drop
no vad
!
dial-peer voice 2101 voip
description *** Voice: WAN to LAN - Outgoing Dial-Peer ***
huntstop
destination-pattern 415T
codec g729r8
max-redirects 5
session protocol sipv2
session target ipv4:10.40.97.2
dtmf-relay rtp-nte digit-drop
no vad
!
dial-peer voice 3000 voip
description *** Fax: LAN to WAN - Incoming Dial-Peer ***
huntstop
session protocol sipv2
incoming called-number 7T
dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
!
dial-peer voice 3001 voip
description *** Fax: LAN to WAN - Outgoing Dial-Peer ***
translation-profile outgoing OUTGOING-SIP-TRK-DIGIT-STRIP
huntstop
destination-pattern 7T
voice-class sip early-offer forced
max-redirects 5
session protocol sipv2
session target ipv4:10.3.33.22
dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
!
dial-peer voice 3100 voip
description *** Fax: WAN to LAN - Incoming Dial-Peer ***
huntstop

```

```
session protocol sipv2
incoming called-number 4155551111[0,1]
dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
!
dial-peer voice 3101 voip
description *** Fax: WAN to LAN - Outgoing Dial-Peer ***
huntstop
destination-pattern 4155551111[0,1]
max-redirects 5
session protocol sipv2
session target ipv4:10.40.97.2
dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
!
dial-peer voice 1 pots
service mgcpapp
port 4/0/0
!
dial-peer voice 2 pots
service mgcpapp
port 4/0/1
!
dial-peer hunt 3
sip-ua
authentication username yyyyyy password 7 xxxxxxxxxxxx
no remote-party-id
retry invite 2
retry response 5
retry bye 2
retry cancel 2
retry register 10
retry options 1
g729-annexb override
!
call-manager-fallback
video
max-conferences 10 gain -6
transfer-system full-consult
log table max-size 1000
ip source-address 10.40.103.1 port 2000
max-ephones 50
max-dn 50
system message primary Ent1_Br1
dialplan-pattern 1 415555.... extension-length 4
transfer-pattern .T
!
Ent1_Br1#
```

Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration

To integrate the Branch 1 Cisco Unity Express with Cisco Unified CM configuration, see the [CallManager for Cisco Unity Express Configuration Example](#).

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