



# 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. This guide is an update to the existing SRND and validates the Cisco UBE functions on the second generation Cisco Integrated Services Router (Cisco ISR-G2) 29xx/39xx and 3945E platforms, All other solution components remain unchanged.

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 31](#)
- [Enterprise 1 HQ Cisco Unified CM Example Configuration, page 34](#)
- [Enterprise 1 HQ Cisco ASA Firewall Example Configuration, page 123](#)
- [Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration, page 122](#)
- [Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration, page 122](#)

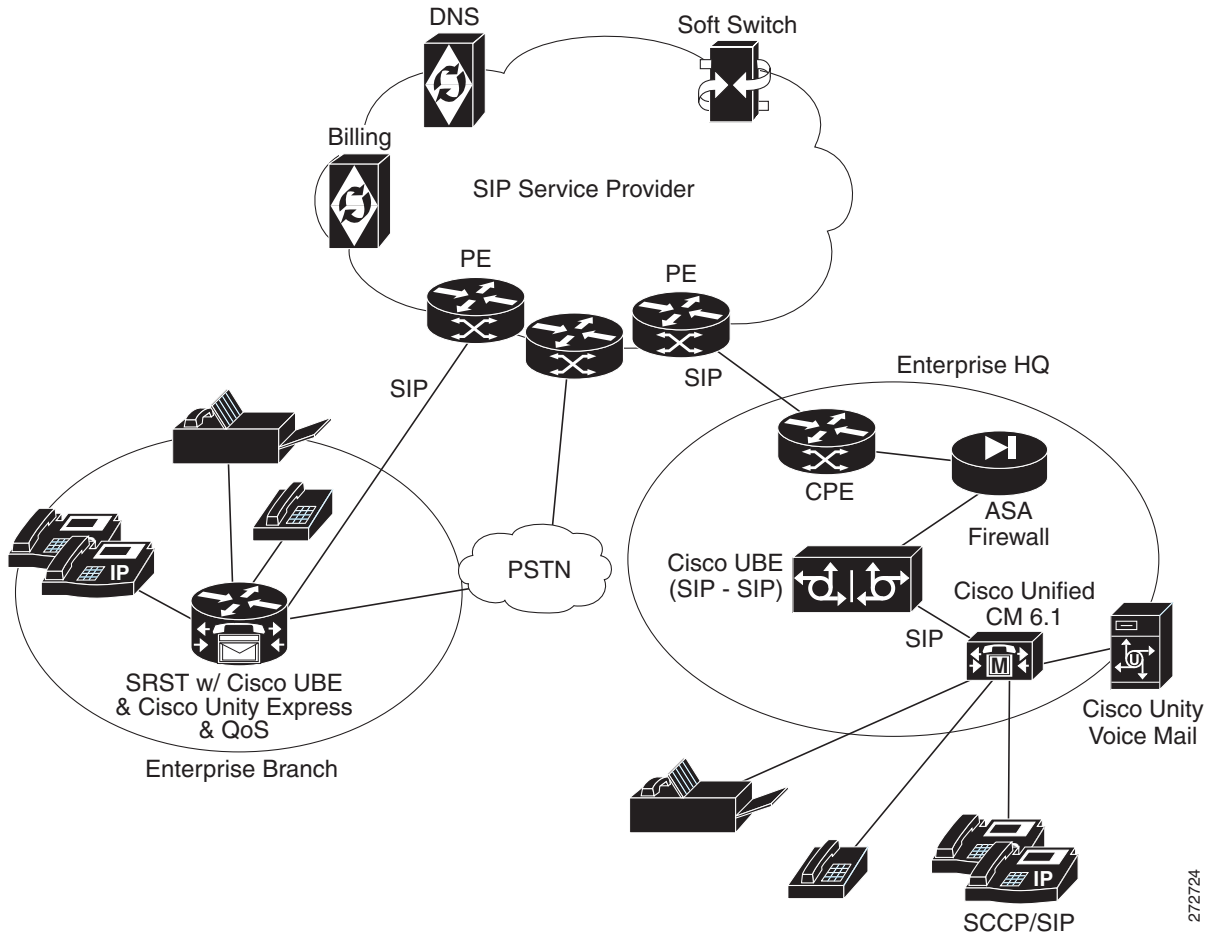
### Enterprise Branch

- [Branch 1 Cisco UBE, TDM Gateway, and Cisco Unified SRST Example Configuration, page 124](#)
- [Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration, page 128](#)

**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



# Prerequisites

Prerequisites are grouped into the following sections:

- [Components Used, page 5](#)
- [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 6](#)

### 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 6.1.3.

### Cisco Unified Border Element

A Cisco 3945 and 3945A series platforms were tested with Cisco IOS Release 15.1(1)T and Cisco UBE version 1.4. The Cisco 2900 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 15.1(1)T.

### Voice Mail at the Enterprise Headquarter Site

Voice mail at the enterprise headquarter site is provided by the Cisco Unity voice mail server, which was 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 is 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 15.1(1)T. 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.

## SIP Call Agent

Various SIP call agents can be used for the feature functionality discussed in this guide. For testing purposes, a BroadSoft call agent release 14 SP7 was used.

The BroadSoft call agent uses the BroadWorks platform. The typical deployment is comprised of three servers installed in a clustered or redundant format:

## PSTN Hop-Off Gateway

A Cisco AS5000 series gateway with PRI trunks was used and tested with Cisco IOS Release 15.1(1)T. You can use other software releases later than Cisco IOS Release 15.1(1)T or other gateway platforms.

## Cisco IOS Software Releases

The test results described in this guide for the Cisco Unified Border Element were conducted using Cisco IOS Release 15.1(1)T. We recommend Cisco IOS Release 15.1(1)T 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 7](#)
- [IP Connectivity, page 16](#)
- [Quality of Service, page 17](#)
- [Voice Mail, page 19](#)
- [Dial Plan, page 19](#)
- [Security, page 19](#)
- [Failover and Redundancy, page 20](#)
- [Fax and Modem, page 20](#)
- [Billing and Management, page 20](#)
- [Best Practices for SIP Trunk Implementation Using Cisco UBE, page 21](#)
- [Caveats, page 22](#)

## 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 9](#)
- [Early Media Cut Through, page 9](#)
- [SIP Trunk Transport Protocols, page 10](#)
- [Monitoring SIP Trunk State, page 10](#)

## 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. IB 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 15.1 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, IB to OOB 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 15.1(1) 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 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](#)
- [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:

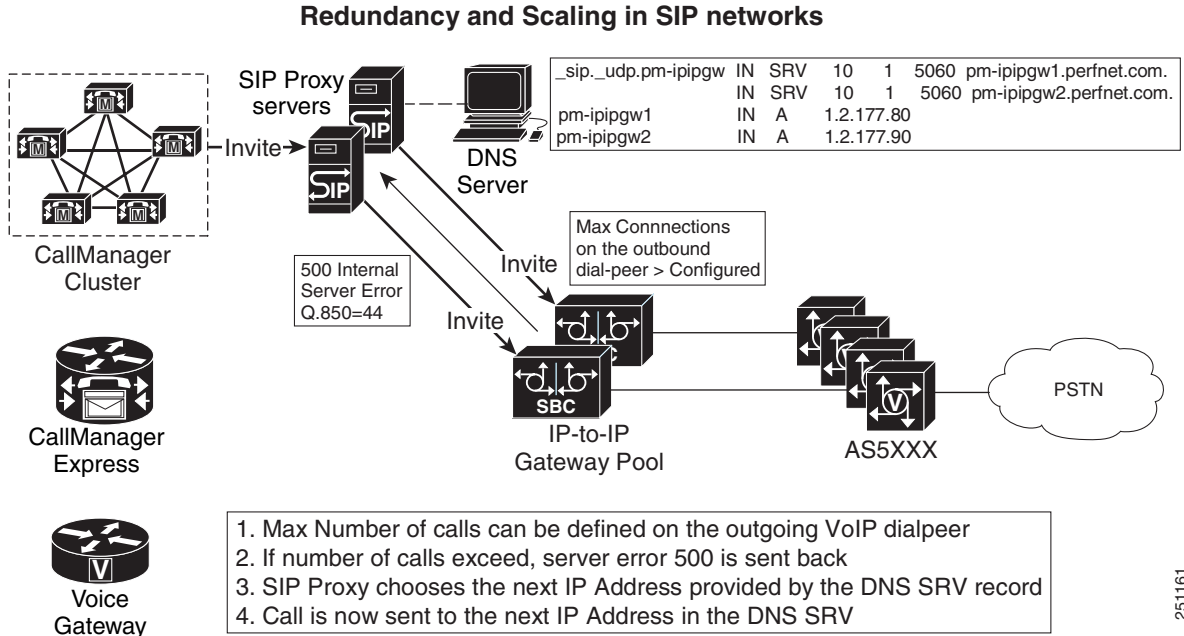
```
!
```

```
dial-peer voice 3670000 voip
description "first hunting for 3670000 to ent2-hq-ipip"
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-ipip"
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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### 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 3](#)).

**Figure 3**      **Configuring Route Groups**

The screenshot shows the 'Route Group Configuration' page for a route group named 'loadbalance-ipipgw60-rg'. The page is divided into several sections:

- Route Group Members:** A list of members with IP address 15.3.30.60.
- Route Group Information:**
  - Route Group Name\*: loadbalance-ipipgw60-rg
  - Distribution Algorithm\*: Top Down
- Route Group Member Information:**
  - Find Devices to Add to Route Group: A search box for 'Device Name contains' with a 'Find' button.
  - Available Devices (select device, then select port below): A list showing '15.3.30.70' and 'pinamajito-ipipgw1-15.5.15.80'.
  - Port(s): A dropdown menu set to 'All' and an 'Add to Route Group' button.
- Current Route Group Members:**
  - Reverse Order of Selected Devices: A button to reverse the order.
  - Selected Devices\* (ordered by highest priority): A list showing '15.3.30.60 (All Ports)'.
  - Removed Devices (to be removed from Route Group when you click Update): An empty list.

Navigation links at the top right include: [Add new Route Group](#), [Back to Find/List Route Groups](#), and [Dependency Records](#). A vertical ID '279873' is visible on the right side of the page.

2. Configure one Route List to club all Route Groups (see [Figure 4](#)).

**Figure 4**      **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 criteria: "2 matching record(s) for Route Group Name begins with "" and a search form with a dropdown set to "begins with", an input field, and a "Find" button. Below the search form, it indicates "and show 20 items per page" and a note: "To list all items, click Find without entering any search text." The results section, titled "Matching record(s) 1 to 2 of 2", contains a table with two entries:

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

At the bottom of the interface, there is a "Delete Selected" button, navigation links "First Previous Next Last", and a page indicator "Page 1 of 1".

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- Configure Route List under Route Pattern Gateway or Route List (see [Figure 5](#)).

**Figure 5** *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](#)

<div style="background-color: #f0f0f0; padding: 5px; margin-bottom: 5px;"><b>Route List Details</b></div> <ul style="list-style-type: none"> <li style="margin-bottom: 5px;"> loadbalance-ippgw60-rg</li> <li style="margin-bottom: 5px;"> loadbalance-ippgw70-rg</li> </ul>	<div style="margin-bottom: 5px;"><b>Route List: loadbalance-ippgw-rl</b></div> <div style="margin-bottom: 5px;">Status: Ready</div> <div style="margin-bottom: 5px;"> <input type="button" value="Copy"/> <input type="button" value="Update"/> <input type="button" value="Delete"/> <input type="button" value="Reset"/> </div>
<div style="background-color: #f0f0f0; padding: 5px; margin-bottom: 5px;"><b>Route List Information</b></div> <div style="margin-bottom: 5px;">Route List Name* <input type="text" value="loadbalance-ippgw-rl"/></div> <div style="margin-bottom: 5px;">Description <input type="text" value="loadbalancebetween60-70"/></div> <div style="margin-bottom: 5px;">Cisco CallManager Group* <input type="text" value="PUB"/></div> <p style="font-size: small; margin: 0;"><b>WARNING!</b> 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> <div style="margin-bottom: 5px;"><input checked="" type="checkbox"/> Enable this Route List (change effective on Update; no reset required)</div>	
<div style="background-color: #f0f0f0; padding: 5px; margin-bottom: 5px;"><b>Route List Member Information</b></div> <div style="margin-bottom: 5px; text-align: center;"><input type="button" value="Add Route Group"/></div> <div style="margin-bottom: 5px;">                     Selected Groups*                      (ordered by highest priority)                 </div> <div style="border: 1px solid #ccc; padding: 5px; margin-bottom: 5px;">                     loadbalance-ippgw60-rg[non-QSIG]                      loadbalance-ippgw70-rg[non-QSIG]                 </div> <div style="text-align: center; margin-bottom: 5px;"> <input type="button" value="v"/> <input type="button" value="^"/> </div> <div style="margin-bottom: 5px;">                     Removed Groups                      (to be removed from Route List when you click Update)                 </div> <div style="border: 1px solid #ccc; padding: 5px; height: 40px; margin-bottom: 5px;"></div> <p style="font-size: x-small; margin: 0;">* 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 (see [Figure 6](#)).

**Figure 6** *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

**Pattern Definition**

Route Pattern\*   
 Partition   
 Description   
 Numbering Plan\*   
 Route Filter   
 MLPP Precedence   
 Gateway or 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**

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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 22](#) 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 15.1(1)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-SHDSL)	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 17](#)
- [Packet Marking, page 18](#)
- [Call Admission Control, page 18](#)
- [Delay, page 18](#)
- [Echo, page 19](#)

## 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 22](#). 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 20](#). 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 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 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 19](#)
- [Encryption of Media and Signaling, page 19](#)
- [Firewall, page 20](#)

## 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 22](#). The password is encrypted using Message Digest 5 (MD5).

## Encryption of Media and Signaling

Virtual Private Network (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 22, backup PSTN access was tested for alternate call routing when SIP trunk access was down.

**At the enterprise headquarter site, PSTN access was not explicitly tested on SIP trunk failure in this test configuration. PSTN access at the enterprise headquarter site can be deployed in a similar way because it was tested in other test configurations on SIP trunk failure.**

## 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 \*\*\*
- Configure valid descriptions for explicit incoming and outgoing VoIP dial-peers to and from the Service Provider/Enterprise to ease troubleshooting. For example:
  - dial-peer voice 100 voip  
description \*\*\* incoming calls from Service Provider \*\*\*  
incoming called number xxx
  - dial-peer voice 200 voip  
description \*\*\* outgoing calls to Enterprise \*\*\*  
destination-pattern xxx
- 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, RTCP, or both to accelerate the detection of *hung* calls.
- Use RFC 2833 to configure DTMF because it is the most widely deployed and most interoperable DTMF mechanism for SIP trunks.
- Enable PRACK on Cisco Unified CM if Cisco UBE is configured to do Delayed Offer to Early Offer conversions for call flows involving early media cut through (18x w/SDP).
- **If using G.729 over WAN, make sure the following CLI command is configured for RFC 3555 backward compliance: `sip-ua g729-annexb override`.**
- Tune the failover timers 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.
- 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.

- Use the Cisco Unified SIP Proxy and Cisco UBE scaling architecture at the HQ location, if SIP trunk capacity requires a stack of Cisco UBEs to scale capacity.
- Pay 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.
- Test call flows with supplementary services since they may present 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.
- Routing for emergency (911) calls using the shared hop-off PSTN gateway should be ensured by the service providers.

## Caveats

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

- Voice calls must use the same static codec. It can be any codec type, but the same codec must be maintained.
- Intra Enterprise calls were tested with G.711 codecs. SIP trunk calls were tested with G.729r8 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 25 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 *Command Lookup Tool* or the Cisco IOS master commands list at [http://www.cisco.com/en/US/docs/ios/mcl/allreleasemcl/all\\_book.html](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**

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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 2900 Series Integrated Services Routers (Cisco 2900 ISR-G2)
- *Cisco 3800 Series Integrated Services Routers*
- Cisco 3900 Series Integrated Services Routers (Cisco 3900 ISR-G2)
- *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 *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

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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 25](#)
- [Cisco ASA 8.0\(4\) Caveats High-Level Operation, page 27](#)
- [Test Topology, page 30](#)
- [Example Configuration Details, page 31](#)
- [Enterprise 1 HQ Cisco UBE Example Configuration, page 31](#)
- [Enterprise 1 HQ Cisco Unified CM Example Configuration, page 34](#)
- [Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration, page 122](#)
- [Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration, page 122](#)
- [Enterprise 1 HQ Cisco ASA Firewall Example Configuration, page 123](#)
- [Branch 1 Cisco UBE, TDM Gateway, and Cisco Unified SRST Example Configuration, page 124](#)
- [Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration, page 128](#)
- [Cisco Unified Border Element Performance Summary, page 128](#)

## 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 15.1(1)T) analog lines for Fax/Modem support
- Cisco UBE (Cisco IOS Release 15.1(1)T)

## 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.
- Intra Enterprise calls were tested with G.711 codecs. SIP trunk calls were tested with G.729r8 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

- 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.
- Cisco Unified CM does not support the midcall audio codec change (CSCsr03120).
- Enhance SIP Trunk display to minimize confusion (CSCsv80045).

### Cisco UBE Version 1.2 (IOS Release 15.1(1)T) Caveats

### Cisco Unity 5.0(1) Caveats

To view the caveats for Cisco Unity 5.0(1), see *Release Notes for Cisco Unity Release 5.0(1)*.

### Cisco Unity Express 3.2 Caveats

To view the caveats for Cisco Unity Express 3.2, see *Release Notes for Cisco Unity Express 3.2*.

## Cisco ASA 8.0(4) Caveats High-Level Operation

Users trying to configure the Voice Enterprise 1 topology should be 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

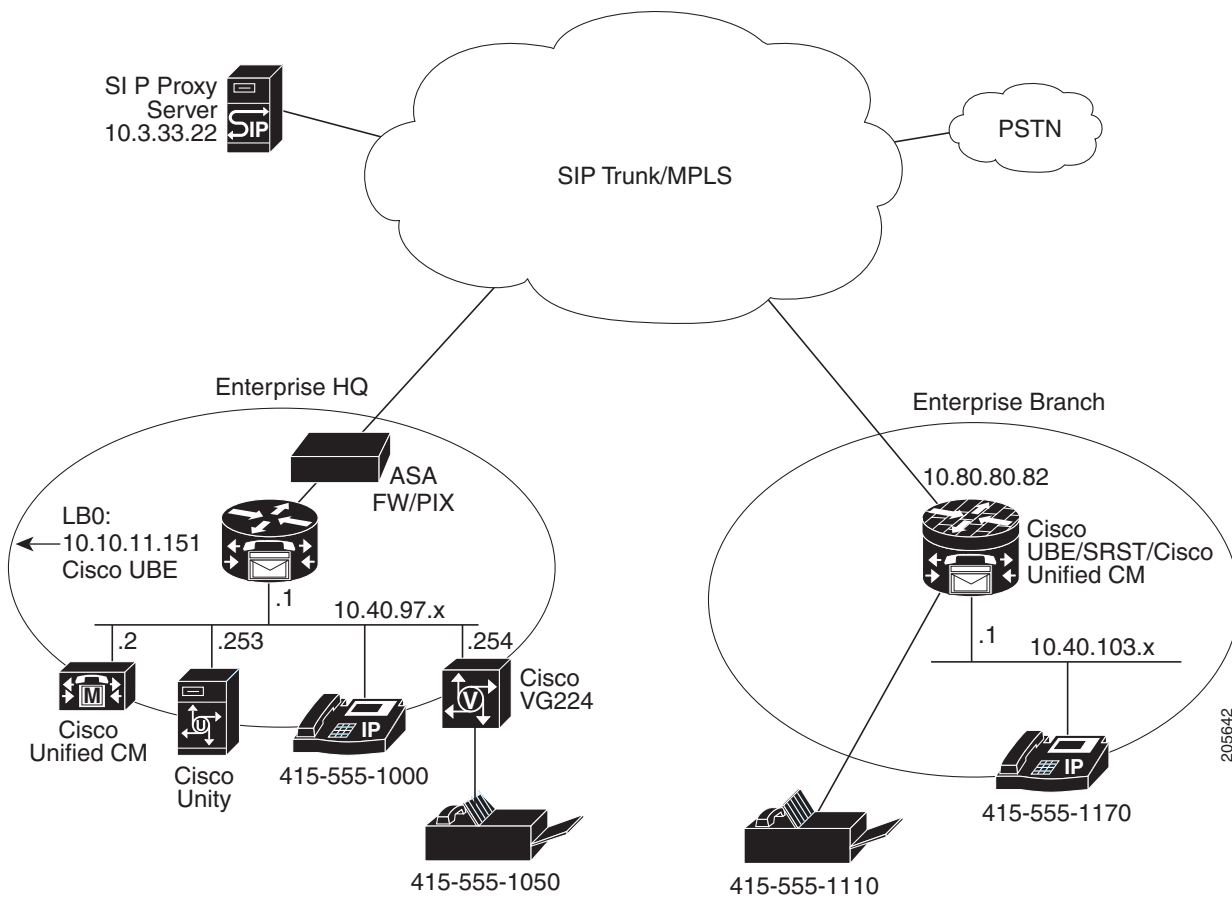
The call admission control (CAC) restrictions that are imposed by Cisco Unified CM for the whole enterprise are as follows:

- **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.”
- **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.
- **CPU%**—Cisco UBE at the enterprise HQ and the Branch restrict the maximum voice and fax/modem calls to configured CPU% threshold.
- **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 7 shows the setup test topology used in example configurations described in the following sections.

Figure 7 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:

- 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.
- 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.
- 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.
- 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 example shows a command-line interface (CLI) configuration example for the enterprise 1 HQ Cisco Unified Border Element for the test topology described in [Figure 7](#).

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

```

voice service voip
mode border-element
allow-connections h323 to h323
allow-connections h323 to sip
allow-connections sip to h323
allow-connections sip to sip
signaling forward unconditional
fax protocol pass-through g711ulaw
modem passthrough nse codec g711ulaw
h323
emptycapability
h245 passthru tcsnonstd-passthru
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 7](#). 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 35](#)
- [Configuring the Cisco Unified CM Call Routing Parameters, page 65](#)
- [Configuring the Cisco Unified CM Media Resources Parameters, page 81](#)

- [Configuring the Cisco Unified CM Voice Mail Parameters, page 98](#)
- [Configuring the Cisco Unified CM Device Parameters, page 105](#)

## 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 35](#)
- [System: Region Parameters, page 36](#)
- [System: Device Pool Parameters, page 49](#)
- [System: Location Parameters, page 58](#)

### System: Server Parameters

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

**Figure 8** System Server Enterprise 1 HQ Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration 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'. 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 'Server Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. Below this are three buttons: 'Save', 'Delete', and 'Add New'. A status message indicates 'Update successful'. The 'Server Information' section contains a form with the following fields: 'Database Replication' (Publisher), 'Host Name/IP Address\*' (10.40.97.2), 'MAC Address', and 'Description' (Ent1-HQ-CUCM). At the bottom, there are three buttons: 'Save', 'Delete', and 'Add New'. A note at the bottom left states '\* - indicates required item.'

273751

## System: Region Parameters

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

**Figure 9** System Region 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 ▾

**Find and List Regions**

+ Add New    Select All    Clear All    Delete Selected

— Status —  
12 records found

**Regions (1 - 12 of 12)**    Rows per Page: 50

Find Regions where Name begins with  Find Clear Filter + -

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

Add New    Select All    Clear All    Delete Selected

273752

Figure 10 System Region Default Cisco Unified CM Administration Window

**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 11 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 is a navigation bar with 'Cisco Unified CM Administration' and a user '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 'Region Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. Below this are action buttons: 'Save', 'Delete', 'Reset', and 'Add New'. The 'Region Information' section shows the 'Name\*' as 'Region Br1 Phones Analog'. The 'Region Relationships' section contains a table with columns for 'Region', 'Audio Codec', 'Video Call Bandwidth', and 'Link Loss Type'. Below the table is a 'NOTE' and a 'Modify Relationship to other Regions' section with a table for selecting other regions and their associated settings.

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

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

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

- \*- 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 12 System Region-Region Branch 1 DSP Farm Cisco Unified CM Administration Window

**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 13 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

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
<ul style="list-style-type: none"> <li>Default</li> <li>Region Br1 Phones Analog</li> <li>Region_Br1_DSPfarm</li> <li>Region_Br1_DSPfarm_Conference</li> <li>Region_Br1_DSPfarm_Transcoder</li> </ul>	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.

273756



Figure 14 System Region-Region Branch 1 DSP Farm Transcoder 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 "Region Configuration" for "Region\_Br1\_DSPfarm\_Transcoder".

**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"/>

Buttons: 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.

273757

Figure 15 System Region-Region 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

Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

**Region Information**

Name\* Region\_Br1\_Phones\_IP

**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
<ul style="list-style-type: none"> <li>Default</li> <li>Region Br1 Phones Analog</li> <li>Region_Br1_DSPfarm</li> <li>Region_Br1_DSPfarm_Conference</li> <li>Region_Br1_DSPfarm_Transcoder</li> </ul>	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.

Figure 16 System Region-Region HQ 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_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

**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.

273759

Figure 17 System Region-Region HQ DSP Farm Conference 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_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 18 System Region-Region HQ DSP Farm Transcoder 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 "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 19 System Region-Region HQ Phones Analog Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration 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". 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 "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\*" field set to "Region\_HQ\_Phones\_Analog". The "Region Relationships" section contains a table with columns for Region, Audio Codec, Video Call Bandwidth, and Link Loss Type. The table lists three relationships: Region\_HQ\_Phones\_IP, Region\_Wan, and Region Br1 Phones Analog, all with G.711 audio codec and 384 kbps video call bandwidth. Below the table is a note: "NOTE: Region(s) not displayed Use System Default Use System Default Use System Default". The "Modify Relationship to other Regions" section has 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 set to "Keep Current Setting". The "Video Call Bandwidth" column has radio buttons for "Keep Current Setting" (selected), "Use System Default", "None", and a text input field for "kbps". The "Link Loss Type" column has a dropdown set to "Keep Current Setting". At the bottom of this section are buttons for Save, Delete, Reset, and 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 20 System Region-Region HQ 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

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
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

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.

273763

Figure 21 System Region-Region WAN 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\_Wan

**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
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

**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** in the Cisco Unified CM Administration window.

**Figure 22** System Device Pool Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface for the 'Device Pool' configuration. The page title is 'Cisco Unified CM Administration' and the breadcrumb is 'System > Device Pool'. The page contains a search bar, a table of device pools, and navigation buttons.

**Find and List Device Pools**

Buttons: Add New, Select All, Clear All, Delete Selected

**Status**

8 records found

**Device Pool (1 - 8 of 8)** Rows per Page: 50

Find Device Pool where: Device Pool Name [dropdown] begins with [dropdown] Find Clear Filter + -

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

Buttons: Add New, Select All, Clear All, Delete Selected

273765

Figure 23 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 24 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

**i** \*- indicates required item.  
**i** \*\*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.  
**i** \*\*\*leave blank to use default.  
**i** \*\*\*\*These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

273767

Figure 25 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 26 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 27 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 28 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 29 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.

27372



Figure 30 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

**i** \*- indicates required item.

**i** \*\*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.

**i** \*\*\*leave blank to use default.

**i** \*\*\*\*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** in the Cisco Unified CM Administration window.

**Figure 31** 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' with the subtitle 'For Cisco Unified Communications Solutions'. 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 32 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 user is identified as "admin". A main menu contains various system management options like "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help".

The main content area is titled "Location Configuration" and includes a "Related Links" section with a "Back To Find/List" button. Below this is a toolbar with icons for "Save", "Delete", "Copy", "Add New", and "Resync Bandwidth".

The configuration is organized into several sections:

- Status:** Shows "Status: Ready" with an information icon.
- Location Information:** A text field for "Name\*" contains "Hub\_Br1".
- Audio Calls Information:** "Audio Bandwidth\*" is set to "85 kbps" with radio buttons for "Unlimited" and "None". A note below states: "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\*" is set to "None" with radio buttons for "Unlimited" and "None".
- Location RSVP Settings:** A table with two columns: "Location" and "RSVP Setting". The "Location" column is empty, and the "RSVP Setting" column contains "Use System Default". A note below the table says "NOTE: Location(s) not displayed".
- Modify Setting(s) to Other Locations:** A table with two columns: "Location" and "RSVP Setting". The "Location" column lists "Hub\_Br1", "Hub\_HQ", "Hub\_None", "Trunk Br1", and "Trunk HQ". The "RSVP Setting" column has a dropdown menu set to "Use System Default".

At the bottom, there is another toolbar with "Save", "Delete", "Copy", "Add New", and "Resync Bandwidth" buttons. A note with an information icon states: "\*- indicates required item."

273775

Figure 33 System Location Hub HQ Cisco Unified CM 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", and a navigation dropdown menu. Below the navigation bar 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 "Location Configuration" and includes a "Related Links" section with a "Back To Find/List" link. Below this are several sections for configuration:

- Status:** Shows "Status: Ready" with an information icon.
- Location Information:** A text field for "Name\*" containing "Hub\_HQ".
- Audio Calls Information:** "Audio Bandwidth\*" is set to "110" kbps. A note states: "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\*" is set to "None".
- Location RSVP Settings:** A table with columns "Location" and "RSVP Setting". The "Location" column is empty, and the "RSVP Setting" column contains "Use System Default". A note below the table says "NOTE: Location(s) not displayed".
- Modify Setting(s) to Other Locations:** A table with columns "Location" and "RSVP Setting". The "Location" column lists "Hub\_Br1", "Hub\_HQ", "Hub\_None", "Trunk Br1", and "Trunk HQ". The "RSVP Setting" column for "Hub\_HQ" is set to "Use System Default".

At the bottom of the configuration area are buttons for "Save", "Delete", "Copy", "Add New", and "Resync Bandwidth". A note at the bottom left states: "i \*- indicates required item."

273776

Figure 34 System Location Hub None 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 "Location Configuration" for a location named "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"> <li>Hub_Br1</li> <li>Hub_HQ</li> <li>Hub_None</li> <li>Trunk_Br1</li> <li>Trunk_HQ</li> </ul>	Use System Default

Buttons: Save, Copy, Add New, Resync Bandwidth

**i** \*- indicates required item.

273777

Figure 35 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"> <li>Hub_Br1</li> <li>Hub_HQ</li> <li>Hub_None</li> <li>Trunk Br1</li> <li>Trunk HQ</li> </ul>	Use System Default

Save | Delete | Copy | Add New | Resync Bandwidth

\*- indicates required item.

273778

Figure 36 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** in the Cisco Unified CM Administration window.

**Figure 37** 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 65](#)
- [Call Routing: Class of Control Parameters, page 71](#)

### Call Routing: Route/Hunt Parameters

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

**Figure 38** Call Routing RouteHunt 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 action buttons: "+ Add New", "Select All", "Clear All", and "Delete Selected". 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 search criteria are "Find Route Patterns where Pattern begins with". The table contains 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 39 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 40 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\* 9.1XXXXXXXXXX

Route Partition Partition-HQ\_Phones\_Analog

Description RP Ent1-HQ Analog 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) 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.

273705

Figure 41 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.

Figure 42 Call Routing RouteHunt Route Pattern RP Ent1 Br1 IP 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 ▾

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
-- Not Selected --	< Not Exist >	

---

\*- indicates required item.

273707

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

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## Call Routing: Class of Control Parameters

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

**Figure 43** 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, and the 'Find and List Partitions' section is active. 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 44 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 the subtitle 'For Cisco Unified Communications Solutions'. A navigation menu shows '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' section containing 'Back To Find/List'. Below the navigation is a toolbar with 'Save', 'Delete', 'Reset', and 'Add New' buttons. The main content area is divided into sections: 'Status' (Ready), 'Partition Information' (Name: Partition-Br1\_Phones\_Analog, Description: Analog Phones, Time Schedule: < None >, Time Zone: Originating Device), and a bottom section with 'Save', 'Delete', 'Reset', and 'Add New' buttons. A note at the bottom states '\*- 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\_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.

273709

Figure 45 Call Routing Class of Control Partition-Partition Br1 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 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' button. Action buttons for 'Save', 'Delete', 'Reset', and 'Add New' are visible. The configuration details are as follows:

- 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

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

273710



Figure 46 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 text 'Cisco Unified CM Administration For Cisco Unified Communications Solutions', and a user menu with 'admin', 'About', and 'Logout'. A secondary navigation bar lists menu items: 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. Below this 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-HQ\_Phones\_Analog
- Description: Analog Phones
- Time Schedule: < None >
- Time Zone:  Originating Device,  Specific Time Zone (Greenwich Standard Time)

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.'

273711

Figure 47 Call Routing Class of Control Partition-Partition HQ 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 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 48 Call Routing Class of Control CSS Cisco Unified CM Administration Window

The screenshot displays the Cisco Unified CM Administration interface. At the top, the navigation menu includes System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main heading is "Find and List Calling Search Spaces". Below this, there are controls for "Add New", "Select All", "Clear All", and "Delete Selected". A status bar indicates "5 records found". The main content area shows a search filter for "CSS Name" set to "begins with" and a "Find" button. Below the search bar is a table with the following data:

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

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

273713

Figure 49 Call Routing Class of Control CSS-CSS Branch 1 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

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\_Analog  
Description CSS-Br1\_Phones\_Analog

**Route Partitions for this Calling Search Space**  
Available Partitions\*\*

Selected Partitions  
Partition-Br1\_Phones\_Analog  
Partition-Br1\_Phones\_IP  
Partition-HQ\_Phones\_Analog  
Partition-HQ\_Phones\_IP

Save Delete Copy Add New

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

273714

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

The screenshot displays the Cisco Unified CM Administration interface. At the top, the navigation bar includes 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is 'Calling Search Space Configuration' for the entity 'CSS-Br1\_Phones\_IP'. The status is 'Ready'. Under 'Calling Search Space Information', the Name and Description are both 'CSS-Br1\_Phones\_IP'. The 'Route Partitions for this Calling Search Space' section shows an empty 'Available Partitions' list and a 'Selected Partitions' list containing: Partition-Br1\_Phones\_IP, Partition-Br1\_Phones\_Analog, Partition-HQ\_Phones\_Analog, and Partition-HQ\_Phones\_IP. At the bottom, there are buttons for 'Save', 'Delete', 'Copy', and 'Add New'. Informational icons indicate that an asterisk (\*) denotes a required item and double asterisks (\*\*) denote selected partitions ordered by highest priority.

273715

Figure 51 Call Routing Class of Control CSS-CSS HQ 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

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

Save Delete Copy Add New

**Status**  
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\*\*

Selected Partitions  
Partition-HQ\_Phones\_Analog  
Partition-Br1\_Phones\_Analog  
Partition-Br1\_Phones\_IP  
Partition-HQ\_Phones\_IP

Save Delete Copy Add New

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

273716

Figure 52 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 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The current page is 'Calling Search Space Configuration' for the entity 'CSS-HQ\_Phones\_IP'. The status is 'Ready'. The 'Calling Search Space Information' section shows the name and description as 'CSS-HQ\_Phones\_IP'. The 'Route Partitions for this Calling Search Space' section features two lists: 'Available Partitions\*\*' (currently empty) and 'Selected Partitions' (containing 'Partition-HQ\_Phones\_IP', 'Partition-Br1\_Phones\_Analog', 'Partition-Br1\_Phones\_IP', and 'Partition-HQ\_Phones\_Analog'). Below the lists are 'Save', 'Delete', 'Copy', and 'Add New' buttons. Informational icons indicate that an asterisk (\*) denotes a required item and double asterisks (\*\*) denote that 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 81](#)
- [Media Resources: Conference Bridge Parameters, page 82](#)
- [Media Resources: Media Termination Point Parameters, page 85](#)
- [Media Resources: Music on Hold Server Parameters, page 88](#)
- [Media Resources: Transcoder Parameters, page 89](#)
- [Media Resources: Media Resource Group Parameters, page 92](#)
- [Media Resources: Media Resource Group List Parameters, page 95](#)

### Media Resources: Annunciator Parameters

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

**Figure 53** Media Resources Annunciator ANN 2 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

**Annunciator Configuration** Related Links: Back To Find/List

Save Reset

**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

Save Reset

\*- 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** in the Cisco Unified CM Administration window.

**Figure 54** 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 main heading is 'Find and List Conference Bridges'. Below this, 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
<a href="#">CFB_2</a>	CFB_2-Ent1-HQ	<a href="#">Default</a>	Registered with 10.40.97.2	10.40.97.2	
<a href="#">CON001AA29DF631</a>	CFB-Ent1-Br1	<a href="#">DevicePool_Br1_DSPfarm</a>	Registered with 10.40.97.2	10.40.103.1	
<a href="#">CON111222333</a>	CFB-Ent1-HQ	<a href="#">DevicePool_HQ_DSPfarm</a>	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 55 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 56 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** in the Cisco Unified CM Administration window.

**Figure 57** Media Resources Media Termination Point 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 Media Termination Points". Below the navigation is a status bar indicating "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

273738

Figure 58 Media Resources Media Termination Point MTP Enterprise 1 Branch 1 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	10.40.103.1
Media Termination Point Type*	Cisco IOS Enhanced Software Media Termination Point
Media Termination Point Name*	MTP001AA29DF631
Description	MTP-Ent1-Br1
Device Pool*	DevicePool_Br1_DSPfarm

Save Delete Copy Reset Add New

\*- indicates required item.

273739

Figure 59 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** in the Cisco Unified CM Administration window.

**Figure 60** Media Resources Music on Hold Server MOH 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 ▾

Music On Hold (MOH) Server Configuration Related Links: Back To Find/List

Save Reset

---

**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.

---

Save Reset

\*- indicates required item.

273741

## Media Resources: Transcoder Parameters

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

**Figure 61** 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 a status of "2 records found".

Transcoder (1 - 2 of 2)							Rows per Page 50
Find Transcoder where							
Name		begins with		Find	Clear Filter	+ -	
<input type="checkbox"/>	Name ^	Description	Device Pool	Status	IP Address	Copy	
<input type="checkbox"/>	XCD001AA29DF631	XCODE-Ent1-Br1	<a href="#">DevicePool_Br1_DSPfarm</a>	Registered with 10.40.97.2	10.40.103.1		
<input type="checkbox"/>	XCODE111222333	XCODE-Ent1-HQ	<a href="#">DevicePool_HQ_DSPfarm</a>	Registered with 10.40.97.2	10.40.97.1		

Buttons at the bottom: Add New, Select All, Clear All, Delete Selected, Reset Selected.

273742



Figure 62 Media Resources Transcoder XCODE Enterprise 1 Branch 1 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 subtitle "For Cisco Unified Communications Solutions". The main navigation menu contains: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The current page is "Transcoder Configuration" for the transcoder "XCODE-Ent1-Br1".

**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 [View Details](#)  
 Common Device Configuration < None > [View Details](#)  
 Special Load Information  Leave blank to use default

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

**i** \*- indicates required item.

273743

Figure 63 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** in the Cisco Unified CM Administration window.

**Figure 64** 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"/>	<a href="#">Br1_HW_MRG</a>	Ent 1 Br1	false	
<input type="checkbox"/>	<a href="#">HQ_HW_MRG</a>	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 65 Media Resources-Media Resource Group Enterprise 1 Branch 1 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 "Br1\_HW\_MRG".

**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)

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)

273746

Figure 66 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** in the Cisco Unified CM Administration window.

**Figure 67** 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".

Below the navigation menu, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected". A status bar indicates "2 records found".

The main content area shows a search bar with the text "Find Media Resource Group List where Name begins with" and a search button. Below the search bar is a table with the following data:

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

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

273748

Figure 68 Media Resources-Media Resource Group List Branch 1 HW MRGL 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 title 'Cisco Unified CM Administration For Cisco Unified Communications Solutions', 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 'Media Resource Group List 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 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

At the bottom of the configuration area, there is another set of buttons: Save, Delete, Copy, Reset, and Add New. A note at the bottom left states: '\* - indicates required item.'

273749

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

The screenshot displays the Cisco Unified CM Administration interface. At the top, the navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the user "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 List 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:** 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

At the bottom, there is a toolbar with buttons for Save, Delete, Copy, Reset, and Add New. A note at the bottom left states: "\*- 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 98](#)
- [Voice Mail: Message Waiting Parameters, page 100](#)
- [Voice Mail: Voice Mail Pilot Parameters, page 103](#)
- [Voice Mail: Voice Mail Profile Parameters, page 104](#)

### 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** in the Cisco Unified CM Administration window.

Figure 70 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 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 '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 filter section for 'Voice Mail Port' with a dropdown for 'Device Name' and a search input field. The main table lists five records:

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	[Copy]
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	[Copy]
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	[Copy]
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	[Copy]
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	[Copy]

At the bottom of the table, there is another set of control buttons: 'Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected'.

273781

Figure 71 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

\*- 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** in the Cisco Unified CM Administration window.

**Figure 72** Voice Mail Message Waiting Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The main heading is "Cisco Unified CM Administration 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 Message Waiting Numbers".

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 search filter: "Message Waiting Numbers where Directory Number begins with and where Message Waiting Indicator is Both". There are "Find" and "Clear Filter" buttons.

<input type="checkbox"/>	Directory Number ^	Description	Partition	Calling Search Space	Copy
<input type="checkbox"/>	1080	MWI-On	Partition-HQ_Phones_IP	CSS-HQ_Phones_IP	
<input type="checkbox"/>	1081	MWI-Off	Partition-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".

273783

Figure 73 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\* 1080

Partition Partition-HQ\_Phones\_IP

Description MWI-On

Message Waiting Indicator\*  On  Off

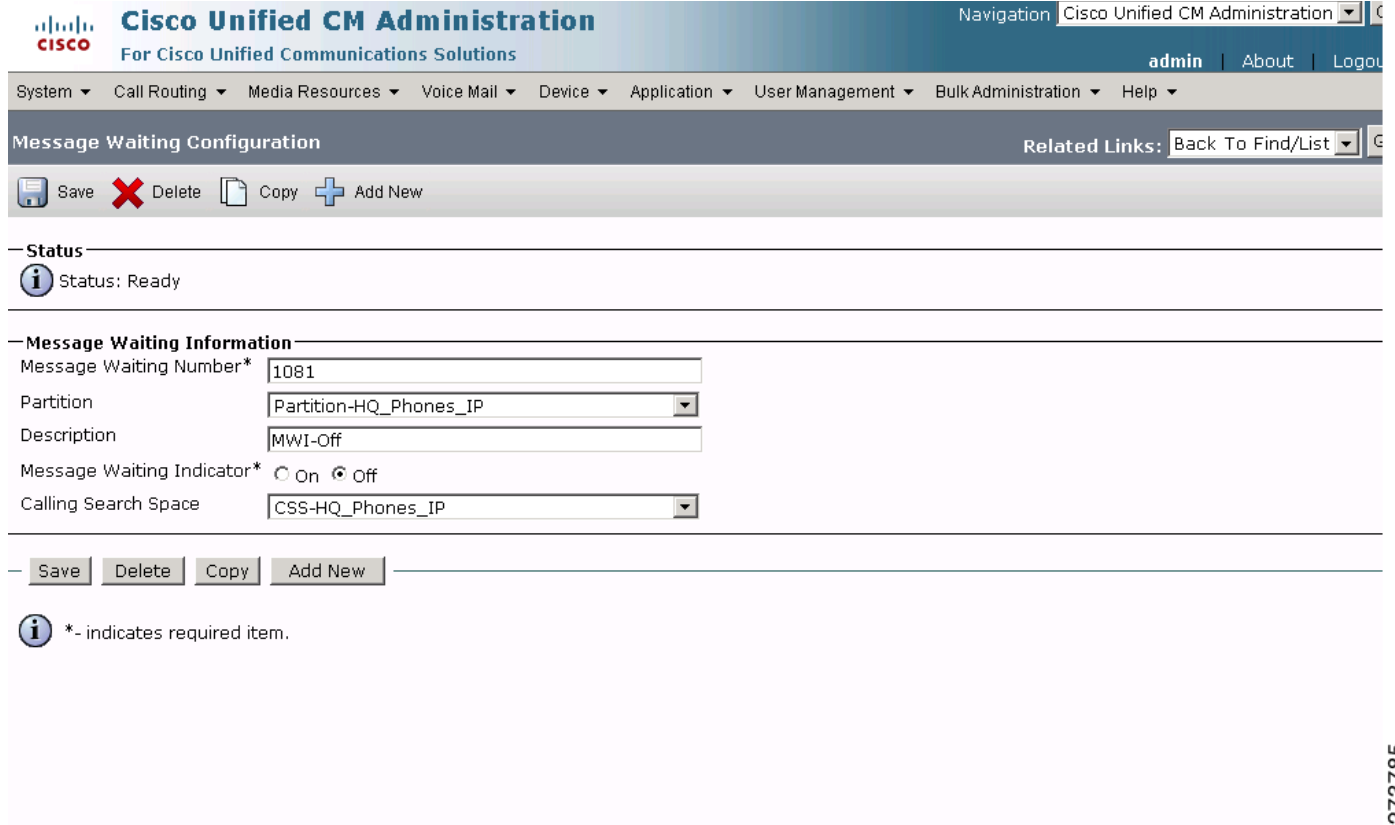
Calling Search Space CSS-HQ\_Phones\_IP

Save Delete Copy Add New

\*- indicates required item.

273784

Figure 74 Voice Mail Message Waiting MWI Off 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\* 1081

Partition Partition-HQ\_Phones\_IP

Description MWI-Off

Message Waiting Indicator\*  On  Off

Calling Search Space CSS-HQ\_Phones\_IP

Save Delete Copy Add New

\*- indicates required item.

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** in the Cisco Unified CM Administration window.

**Figure 75** Voice Mail-Voice Mail Pilot 1099 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 the subtitle "For Cisco Unified Communications Solutions". A navigation menu is visible with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The current page is titled "Voice Mail Pilot Configuration" and shows a "Status: Ready" indicator. Below this, the "Voice Mail Pilot Information" section contains the following fields:

- Voice Mail Pilot Number: 1099
- Calling Search Space: CSS-HQ\_Phones\_IP
- Description: Voicemail Pilot
- Make this the default Voice Mail Pilot for the system

At the bottom of the form, there are buttons for "Save", "Delete", and "Add New". A note at the bottom left states: "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** in the Cisco Unified CM Administration window.

**Figure 76** 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**  
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

\*- indicates required item.  
\*\*- 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 105](#)
- [Device: Phone Parameters, page 112](#)
- [Device: Trunk Parameters, page 117](#)

### Device: Gateway Parameters

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

**Figure 77** Device Gateway Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo and the text 'Cisco Unified CM Administration For Cisco Unified Communications Solutions'. The main navigation menu is expanded to show 'Device > Gateway'. Below the navigation menu, there is a 'Find and List Gateway' section with various action buttons: '+ Add New', 'Select All', 'Clear All', 'Delete Selected', and 'Reset Selected'. A status bar indicates '2 records found'. The main content area displays 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							3845	Cisco	See Endpoints	
SKIGWOC863972F5	Ent1-HQ-VG224							VG224		See Endpoints	

273718



Figure 78 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 79 Device Gateway Enterprise 1 Branch 1 Enterprise 1.com pots 1110 Cisco Unified CM Administration Window

**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 80 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/SUO/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/SUO/0@Ent1\_Br1.Ent1.com**

Display (Internal Caller ID) 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/SUO/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/SUO/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

Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 81 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 82 Device Gateway Enterprise 1 HQ VG224 ANA 1050 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 ▾

Phone Configuration Related Links: Back to Gateway

Save Delete Copy Reset Add New

---

**Status**  
Status: Ready

---

**Association Information**

Modify Button Items

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](#)  
[Device Mobility Settings](#)

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

---

Save Delete Copy Reset 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 83 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 (ANOC863972F5400)

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 ANOC863972F5400  
 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 <input type="checkbox"/>		< 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 <input type="checkbox"/>		< None >
Secondary Calling Search Space for Forward All		< None >
Forward Busy Internal <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward Busy External <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward No Answer Internal <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward No Answer External <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward No Coverage Internal <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward No Coverage External <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward on CTI Failure <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward Unregistered Internal <input type="checkbox"/> or <input type="checkbox"/>		< None >
Forward Unregistered External <input type="checkbox"/> or <input type="checkbox"/>		< 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 ANOC863972F5400**

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 ANOC863972F5400**

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 ANOC863972F5400**

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** in the Cisco Unified CM Administration window.

Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 84 Device Phone 415551000 Cisco Unified CM Administration Window

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

- Association Information:** Lists associated items like Line 1 (Line 1), Line 2 (Line 2), and Line 3 (Line 3).
- Device Information:** Shows registration details, IP address (10.40.87.2), device name (415551000), and various configuration options like Device Profile, Common Phone Profile, and Call Forwarding.
- Provision Specific Information:** Includes Packet Capture Mode, Presence Group, Device Security Profile, and Unattended Park settings.
- Configuration Authority Proxy Function (CAPF) Information:** Details CAPF operation, authentication mode, and key size.
- Extension Module Information:** Lists module 1 and module 2 details.
- External Data Location Information:** Provides fields for information, directory, messages, services, and authentication server.
- Extension Information:** Includes log out profile, log in time, and log out time.
- MFP Information:** Shows MFP format, MFP indication, and MFP interruption.
- Do Not Disturb:** Configures DND options and incoming call alert.
- Secure Shell Information:** Sets secure shell user and password.
- Product Specific Configuration Layout:** A large section with numerous checkboxes and dropdowns for features like Forwarding Delay, PC Port, Settings Access, PC Voice MUX Access, Video Capabilities, Auto Line Select, Web 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 85 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   
 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		< None >
Secondary Calling Search Space for Forward All		< None >
Forward Busy Internal		CSS-HQ_Phones_IP
Forward Busy External		CSS-HQ_Phones_IP
Forward No Answer Internal		CSS-HQ_Phones_IP
Forward No Answer External		CSS-HQ_Phones_IP
Forward No Coverage Internal		< None >
Forward No Coverage External		< None >
Forward on CTI Failure		< None >
Forward Unregistered Internal		CSS-HQ_Phones_IP
Forward Unregistered External		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)  
 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 SEP00187371C3FA**

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.

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  
 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

**Legend:**  
 i \* indicates required item.  
 \*\* Changes to Line or Directory Number settings require restart.

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Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 86 Device Phone 4155551170 Cisco Unified CM Administration Window

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

Figure 87 Device Phone 1170 Cisco Unified CM Administration Window

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration > admin > ABOUT > LOGO

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

Directory Number Configuration  
Related Links: Configure Device (SEP0019E8ABBE78)

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: SEP0019E8ABBE78  
[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 SEP0019E8ABBE78**

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: [ Use System Default ]  
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 SEP0019E8ABBE78**

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 SEP0019E8ABBE78**

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.

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## Device: Trunk Parameters

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

Figure 88 Device Trunk Cisco Unified CM Administration Window

The screenshot shows the 'Find and List Trunks' interface in the Cisco Unified CM Administration console. The page title is 'Cisco Unified CM Administration' with a navigation menu. Below the title is a breadcrumb trail: System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > Help. The main heading is 'Find and List Trunks' with action buttons: Add New, Select All, Clear All, Delete Selected, and Reset Selected. A status bar indicates '4 records found'. Below this is a search section for 'Trunks (1 - 4 of 4)' with a search filter set to 'Device Name' and 'begins with'. The search results are displayed in a table with the following columns: Name, Description, Calling Search Space, Device Pool, Route Pattern, Partition, Route Group, Priority, Trunk Type, and SIP Trunk Security Profile. The table contains four rows of trunk configurations. At the bottom of the table are buttons for Add New, Select All, Clear All, Delete Selected, and Reset Selected.

<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"/>	<a href="#">10.10.11.151</a>	Ent1-HQ-CUBE1	<a href="#">CSS-HQ_Phones_IP</a>	<a href="#">DevicePool_WAN</a>	<a href="#">9.1XXXXXXXXXX</a>	<a href="#">Partition-HQ_Phones_Analog</a>			SIP Trunk	<a href="#">Non Secure SIP Trunk Profile</a>
<input type="checkbox"/>	<a href="#">10.10.11.151</a>	Ent1-HQ-CUBE1	<a href="#">CSS-HQ_Phones_IP</a>	<a href="#">DevicePool_WAN</a>	<a href="#">9.1XXXXXXXXXX</a>	<a href="#">Partition-HQ_Phones_IP</a>			SIP Trunk	<a href="#">Non Secure SIP Trunk Profile</a>
<input type="checkbox"/>	<a href="#">10.80.80.82</a>	Ent1-Br1-CUBE1	<a href="#">CSS-Br1_Phones_IP</a>	<a href="#">DevicePool_WAN</a>	<a href="#">9.1XXXXXXXXXX</a>	<a href="#">Partition-Br1_Phones_IP</a>			SIP Trunk	<a href="#">Non Secure SIP Trunk Profile</a>
<input type="checkbox"/>	<a href="#">10.80.80.82</a>	Ent1-Br1-CUBE1	<a href="#">CSS-Br1_Phones_IP</a>	<a href="#">DevicePool_WAN</a>	<a href="#">9.1XXXXXXXXXX</a>	<a href="#">Partition-Br1_Phones_Analog</a>			SIP Trunk	<a href="#">Non Secure SIP Trunk Profile</a>

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Figure 89 Device Trunk Enterprise 1 HQ 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\*: 10.10.11.151  
 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

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## Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 90 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 91 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 92 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

**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 Unity and Cisco Unity Express Example Configuration

To integrate the Cisco Unity version 5.0 with Cisco Unified CM configuration, see [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 example shows a CLI configuration for the enterprise 1 HQ the Cisco VG224 Analog Phone Gateway for the test topology described in [Figure 7](#).

```
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 example shows a CLI configuration for the enterprise 1 HQ the Cisco ASA 8.0(4) 5500 Series Adaptive Security Appliances firewall for the test topology described in [Figure 7](#).

```

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 example shows a CLI configuration 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 7](#).

```

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 yyyyy 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 [CallManager for Cisco Unity Express Configuration Example](#).

## Cisco Unified Border Element Performance Summary

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## For a performance summary of the Cisco Unified Border Element for both the enterprise 1 HQ and Branch

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