

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 SRST deployed for voice services. The Cisco Integrated Services Router (Cisco ISR) running the Cisco Unified SRST deployed for voice unified at the edge of the network. Cisco UBE plays an important role in serving multiple functions when connecting to other networks.

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

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

Network Topology

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

Enterprise Headquarter

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

Enterprise Branch

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

Service Provider

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



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

Prerequisites

Prerequisites are grouped into the following sections:

- Components Used, page 4
- Cisco IOS Software Releases, page 6
- Conventions, page 6

Components Used

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

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

Cisco Unified Communications Manager

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

Cisco Unified Border Element

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

SCCP Analog Voice Gateway

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

Voice Mail at the Enterprise Headquarter Site

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

Cisco Adaptive Security Appliance Firewall Appliance

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

Note

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

Cisco Survivable Remote Site Telephony

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

Cisco IOS Software Releases

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

Conventions

Refer to Cisco Technical Tips Conventions for information on document conventions.

Solution Description

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

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

The focus of services using this solution are:

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

The following topics describe the solution:

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

Feature Summary

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

Enterprise Headquarter Site Features

• Cisco Unified Communications Manager call control

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

Enterprise Branch Offices Features

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

Service Provider Features

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

Basic Phone Features Served in the Topology

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

SIP Trunking Design Considerations

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

- DTMF Transport, page 8
- SIP Delayed Offer and Early Offer, page 8
- Early Media Cut Through, page 9
- SIP Trunk Transport Protocols, page 9
- Monitoring SIP Trunk State, page 9

DTMF Transport

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

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

OOB SIP DTMF signaling methods include:

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

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

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

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

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

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

SIP Delayed Offer and Early Offer

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

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

Note

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

Early Media Cut Through

The terms Early Offer and Early Media are often confused.

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

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

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

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

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

SIP Trunk Transport Protocols

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

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

Monitoring SIP Trunk State

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

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

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

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

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

SIP Trunk Redundancy and Load Balancing

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

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

Dial-peer preferences and Dial-peer Hunting

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

```
dial-peer voice 3670000 voip
 description "first hunting for 3670000 to ent2-hq-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

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Use the setup example shown in Figure 2 into achieve redundancy based on DNS SRV.

Figure 2



Redundancy and Cooling in CID networks

SIP Network Redundancy and Scaling Based on DNS SRV

GK load balancing for H.323 Networks

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

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

Redundancy and Scaling in H.323 Networks



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Route List & Route Group option from CCM

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

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

Figure 4 Configuring Route Groups

Route Group	Configurat	ion	Back to Find/List Ro	Route Group oute Groups	5
Route Group Members	Route Group: load	balance-ipipgw60-rg			
15.3.30.60	Status: Ready				
	Update Delete				
	Route Group Inform	mation			
	Route Group Name*	loadbalance			
	Distribution Algorithm	m* Top Down	•		
	Route Group Memb	per Information			
	Find Devices to Ad	ld to Route Group			
	Device Name contai	ns		Find	
	Available Devices (select device, then select port below)	15.3.30.70 pinamojito-ipipgw1-15.5.15.80			
	Port(s)	All Add to Route Group			
	Current Route Gro	up Members			
		Reverse Order of Select	ted Devices		
	Selected Devices* (ordered by highest priority)	15.3.30.60 (All Ports)		¢	
		•	<u>۸</u>		
	Removed Devices (to be removed from Route Group when you click Update)				273873

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273874

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

Figure 5	Configuring A	A Route List for Route Groups	5
Find a	and List Route	Groups	Add a New Route Group
2 m	atching record(s) for F	Route Group Name begins with	י ""
	Route Groups where Route G show 20 items per page To list all item	roup Name begins with 💌	Find
Matchir	g record(s) 1 to 2 of 2		
	Route Group Name	1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 - 1997 -	
	loadbalance-ipipgw60-rg		
E 🚟	loadbalance-ipipgw70-rg		
Delete	Selected	First Previous Next Last	Page 1 of 1

3. Configure Route List under Route Pattern Gateway or Route List (see Figure 6_.

Figure 6 Add a new Route List Back to Find/List Route Lists **Route List Configuration** Dependency Records Route List Details Route List: loadbalance-ipipgw-rl loadbalance-ipipgw60-rg Status: Ready Ioadbalance-ipipgw70-rg Copy Update Delete Reset **Route List Information** Route List Name* loadbalance-ipipgw-rl loadbalancebetween60-70 Description Cisco CallManager PUB • Group* WARNING! The selected Cisco CallManager Group has only one Cisco CallManager configured. For the control process to have redundancy protection, please select a Cisco CallManager Group with more than one Cisco CallManager. Enable this Route List (change effective on Update; no reset required) **Route List Member Information** Add Route Group Selected Groups* (ordered by highest loadbalance-ipipgw60-rg[non-QSIG] loadbalance-ipipgw70-rg[non-QSIG] priority) V **V** Removed Groups (to be removed from Route List when you click Update) * indicates required item 273875

Configuring A Route List Under Route Pattern Gateway or Route List

Figure 7

Configuring Max-Con

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

0	C C C C C C C C C C C C C C C C C C C
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 Patter Copy Update Delete Pattern Definition	m automatically resets the associated gateway or Route List
Route Pattern*	6xxx
Partition	< None >
Description	via 15.5.15.60
Numbering Plan*	North American Numbering Plan
Route Filter	< None >
MLPP Precedence	Default
Gateway or Route List*	loadbalance-ipipgw-rl 💌 (Edit)
Route Option	Route this pattern
	C Block this pattern - Not Selected -
Call Classification*	OffNet Commentation Allow Device Override
Provide Outside Dial Tone	Allow Overlap Sending Urgent Priority
Require Forced Authorizati	on 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 S
Connected Party Transforma	Default

IP Connectivity

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

The sample configuration in the "Configurations" section on page 21 shows a Gigabit Ethernet interface.

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

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

Table 1 Cisco CPE Router Network Connectivity Options

Quality of Service

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

- Congestion Management, page 16
- Packet Marking, page 17
- Call Admission Control, page 17
- Delay, page 17
- Echo, page 18

Congestion Management

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

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

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

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

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

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

Packet Marking

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

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

Call Admission Control

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

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

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

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

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

Note

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

Delay

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

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

Echo

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

Voice Mail

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

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

Dial Plan

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

Security

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

- Authentication, page 18
- Encryption of Media and Signaling, page 18
- Firewall, page 19

Authentication

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

Encryption of Media and Signaling

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

Firewall

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

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

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

Failover and Redundancy

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

Fax and Modem

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

Billing and Management

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

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

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

Best Practices for SIP Trunk implementation Using Cisco UBE

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

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

- description *** dial-peer to Service Provider ***
- description *** dial-peer to Publisher Cisco Unified CM ***
- description *** dial-peer to Subscriber Cisco Unified CM ***
- Always use a keepalive mechanism, such as Out of Dialog OPTIONS-ping, over the SIP trunk to detect upstream entity failure before routing calls to the service provider.
- Configure the Cisco UBE for media inactivity based on RTP, or RTCP, or both to accelerate the detection of *hung* calls.
- Because it is the most widely deployed and most interoperable DTMF mechanism for SIP trunks, use RFC 2833 to configure DTMF.
- If Cisco UBE is configured to do Delayed Offer to Early Offer conversions, ensure that PRACK is enabled on Cisco Unified CM, for call flows involving early media cut through (18x w/SDP) to work seamlessly.
- Fine tune the failover timers, especially when using clustered/DNS-SRV addressing.

To ensure minimum Post Dial Delay during failover situations, fine tune the **sip-ua retry** *xxx* **parameters**, where *xxx* is the request name and response code. We recommend the value for INVITEs as *retry invite 2*.

• Do not configure Cisco HSRP on the router that runs Cisco UBE functionality.

The Layer 3 and Layer 7 embedded SIP addresses can be unpredictable when Cisco HSRP is enabled. Refer to the caveats section for exact Bug-ID's.

• Use SIP profiles to insert or remove elements in the SIP headers.

SIP Profiles is a very powerful SIP message normalization and protocol repair tool that can quickly fix or create a workaround to minor interoperability issues when two SIP implementations communicate with each other. This feature is available in Cisco IOS 12.4(15)XZ and Cisco IOS 12.4(20)T and later.

- If SIP trunk capacity requires a stack of Cisco UBEs to scale capacity, consider using the Cisco Unified SIP Proxy and Cisco UBE scaling architecture at the HQ location.
- Pay close attention to DTMF interoperability and call flows.

Adjust the payload types for DTMF as needed when the default Cisco values are in conflict (for example, PT 96 is used for RFC 2833, which is by default reserved for cisco fax-relay).

- Adjust SIP incoming and outgoing ports as required to accommodate send and listen devices on non-standard SIP ports.
- Always test call flows with supplementary services as they present the most likely interoperability issues.
- Configure ACLs on Cisco UBE to allow traffic only from valid call agents and endpoints to avoid toll-fraud.

You can configure CLI commands such as allow term.

- Configure fax traffic on TDM PSTN access if at all possible
- Mark all the outbound voice traffic with the appropriate DSCP values so that it gets the right priority in the service provider network. All other traffic should be appropriately marked.
- Provision backup FXO trunks on the Cisco CPE router to provide emergency PSTN access if the SIP trunk is down.

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• The service provider should ensure appropriate call routing for emergency (911) calls using the shared hop-off PSTN gateway.

Caveats

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

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

Configurations

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

Note

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

Configuration Verification

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

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

The firewall configuration can be verified with the following commands:

- show ip inspect sessions
- show ip inspect statistics

Troubleshooting



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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 3800 Series Integrated Services Routers
- Cisco Cable High-Speed WAN Interface Cards
- Cisco High Density Analog and Digital Extension Module for Voice and Fax
- Cisco IAD243X Business Class Integrated Access Device
- Cisco Systems Support
- *"Configuring Conferencing"* chapter of the Cisco Unified Communications Manager Express System Administrator Guide
- Configuring Frame Relay and Frame Relay Traffic Shaping
- Configuring SIP Support for Hookflash
- Echo Analysis for Voice over IP
- Enterprise QoS Solution Reference Network Design Guide
- Equivalent MIB Objects for VoIP show Commands
- IP Communications Voice/Fax Network Module
- Quality of Service for Voice Over IP

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- RADIUS VSA Voice Implementation Guide
- Service Provider Quality-of-Service Overview
- Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)

Obtaining Documentation and Submitting a Service Request

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

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

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

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

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

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

Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration, page 125

Overview of Test Configurations

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

Enterprise 1 HQ Components

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

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

Enterprise 1 and Branch 1 Components

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

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

Caveats

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

Global Caveats

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

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

Cisco Unified CM 6.1.0.9901-372 Caveats

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

High-Level Operation

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

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

CAll Flow Within Enterprise 1

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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 l 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



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

Test Topology

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

Figure 8 Test Topology



Example Configuration Details

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

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

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

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

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

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

Enterprise 1 HQ Cisco UBE Example Configuration

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

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

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```
voice service voip
 address-hiding
 allow-connections sip to sip
 fax protocol pass-through g711ulaw
modem passthrough nse codec g711ulaw
 sip
 bind control source-interface Loopback0
 bind media source-interface Loopback0
 min-se 2000
 header-passing error-passthru
  options-ping 1200
  listen-port non-secure 5090
 midcall-signaling passthru
1
voice translation-rule 1
rule 1 /^61/ /1/
rule 2 /^71/ /1/
voice translation-profile OUTGOING-SIP-TRK-DIGIT-STRIP
translate called 1
I.
Т
interface Loopback0
ip address 10.10.11.151 255.255.255.255
I.
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
1
ip rtcp report interval 9000
1
sccp local GigabitEthernet0/0
sccp ccm 10.40.97.2 identifier 5 priority 1 version 6.0
sccp
1
sccp ccm group 10
associate ccm 5 priority 1
associate profile 10 register MTP111222333
associate profile 12 register CON111222333
associate profile 11 register XCODE111222333
1
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
1
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
1
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
1
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
1
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
1
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
```

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```
Enterprise 1 HQ Cisco Unified CM Example Configuration
```

```
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
1
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 180
1
sip-ua
keepalive target ipv4:10.3.33.22
authentication username yyyy password 7 xxxxxxxxx
no remote-party-id
retry invite 2
retry bye 2
retry cancel 2
 timers keepalive active 600
reason-header override
g729-annexb override
1
Ent1_HQ_CUBE1#
```

Enterprise 1 HQ Cisco Unified CM Example Configuration

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

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

Configuring the Cisco Unified CM System Parameters

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

- System: Server Parameters, page 33
- System: Region Parameters, page 34
- System: Device Pool Parameters, page 47
- System: Location Parameters, page 56

System: Server Parameters

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

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

cisco		nified CM Ad		ation			Navigation <mark>Cisco</mark>	Unified CM . admin	Administrat	ion 💌 🤇
System 👻	Call Routing 👻	Media Resources 👻	Voice Mail 👻	Device 👻	Application \bullet	User Management 👻	Bulk Administration 👻	Help 👻		
Server Co	onfiguration						Related Link	s: Back To	Find/List	•
🔜 Save	🗙 Delete 🗖	🔓 Add New								
- Server II	Replication e/IP Address* ess	Publisher 10.40.97.2 Ent1-HQ-CUCM								
Save	Delete Ad	dd New								
(i) *- inc	dicates require	d item.								

System: Region Parameters

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

Figure 10 System Region Cisco Unified CM Administration Window

CIECO	co Unified CM Administration	Navigation Cisco Unified CM Administration 💌 admin About Log
	uting → Media Resources → Voice Mail → Device → Application → User Managem	
nd and List Re	gions	
Add New	Select All 🔛 Clear All 💥 Delete Selected	
tatus 12 records fo	bund	
egions (1 - i	12 of 12)	Rows per Page 50 💌
nd Regions whe	ere Name begins with 💌 🛛 🖓 📼]
	Name 📥	
	Default	
	Region Br1 Phones Analog	
	Region Br1 DSPfarm	
	Region Br1 DSPfarm Conference	
	Region Br1 DSPfarm Transcoder	
	Region Br1 Phones IP	
	Region HQ_DSPfarm	
	Region HQ DSPfarm Conference	
	Region HQ DSPfarm Transcoder	
	Region HQ Phones Analog	
	Region HQ Phones IP	
	Region Wan	
Add New S	Select All Clear All Delete Selected	

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CISCO Unified CM Admin For Cisco Unified Communications Solu			admin About Log	
iystem 👻 Call Routing 👻 Media Resources 👻 Voice t	Mail 👻 Device 👻 Application 👻	User Management 👻 🛛 Bulk Administratio	on ▾ Help ▾	
egion Configuration		Related	Links: Back To Find/List	
🔒 Save 🗙 Delete				
Region Information Name* Default				
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	¥ideo Call Bandwidth	Link Loss Type	
Default	Keep Current Setting	Keep Current Setting	Keep Current Setting 💌	
Region Br1 Phones Analog Region_Br1_DSPfarm		🔿 Use System Default		
Region_Br1_DSPfarm_Conference Region_Br1_DSPfarm_Transcoder		O None O kbps		
Save Delete Reset Add New		· · ·		

Figure 11 System Region Default Cisco Unified CM Administration Window

(i) *- indicates required item.

(i)

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**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.

Enterprise 1 HQ Cisco Unified CM Example Configuration

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

tem 👻 Call Routing 👻 Media Resources 👻 🕚	/oice Mail 🔻	Device 👻 Application 👻 U	ser Management 🔻	Bulk Administration	▼ Help ▼
gion Configuration				Related Li	nks: Back To Find/List
Save 🗙 Delete 👇 Reset 🕂 Add New	,		_		
🖌 save 👗 Delete 🍸 Reset 🥁 Aud New					
legion Information					
ame* Region Br1 Phones Analog					
egion Relationships					
Region 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
IOTE: Regions(s) not displayed	Use Sy	stem Default	Use System Defa	ult	Use System Default
lodify Relationship to other Regions					
Regions		Audio Codec	Video	Call Bandwidth	Link Loss Type
Default Design But Blowner Amelan		Keep Current Setting 💌	🛛 💿 Keep C	Current Setting	Keep Current Setting 💌
Region Br1 Phones Analog Region Br1 DSPfarm			· · · · · · · · · · · · · · · · · · ·	stem Default	
Region_Br1_DSPfarm_Conference			C None	_	
Region_Br1_DSPfarm_Transcoder			o [kbps	
Save Delete Reset Add New -					

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

Cisco Unified CM Ad For Cisco Unified Communicatio		n	Naviga	ation Cisco Unified CM Administration 💌 admin About Log
ystem 👻 Call Routing 👻 Media Resources 👻	Voice Mail 👻 Devic	e 👻 Application 👻 I	Jser Management 👻 🛛 Bulk Admi	inistration 👻 Help 👻
egion Configuration			Re	elated Links: Back To Find/List 💌
🚽 Save 🗶 Delete Paset 🛟 Add N	ew			
Region Information Name [*] Region_Br1_DSPfarm				
Region Relationships				
Region Region Br1 DSPfarm	Au	idio Codec G.729	Video Call Bandwidth 384	h Link Loss Type 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
IOTE: Regions(s) not displayed	Use System	n Default	Use System Default	Use System Default
lodify Relationship to other Regions ——				
Regions Default Region Br1 Phones Analog Region_Br1_DSPfarm Region_Br1_DSPfarm_Conference Region_Br1_DSPfarm_Transcoder	► ►	Audio Codec	Video Call Bandw Keep Current Set C Use System Defa C None C kbps	tting 🛛 🛛 Keep Current Setting 💌
Save Delete Reset Add New -				

**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.

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Enterprise 1 HQ Cisco Unified CM Example Configuration

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

	lutions		admin About I
stem ▼ Call Routing ▼ Media Resources ▼ Voic	e Mail 👻 Device 👻 Application 👻	User Management 👻 Bulk Administratio	on ▼ Help ▼
gion Configuration		Related	Links: Back To Find/List
]] Save 💢 Delete			
Region Information			
lame* Region_Br1_DSPfarm_Conference			
Region Relationships			
Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_Phones_IP	G.711	384	Use System Default
Region_HQ_Phones_IP	G.729	384	Use System Default
Region_Wan	G.729	384	Use System Default
NOTE: Regions(s) not displayed	Use System Default	Use System Default	Use System Default
Modify Relationship to other Regions			
Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	Keep Current Setting	 Keep Current Setting 	Keep Current Setting 💌
Region Br1 Phones Analog		C Use System Default	
Region_Br1_DSPfarm_Conference	1	C None	
Region_Br1_DSPfarm_Transcoder	1	C kbps	
Save Delete Reset Add New			
i) *- indicates required item.			

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

CISCO Unified CM Admin For Cisco Unified Communications So			isco Unified CM Administration
rstem 👻 Call Routing 👻 Media Resources 👻 Voic	e Mail 👻 Device 👻 Application 👻 U	ser Management 👻 🛛 Bulk Administratio	n ▼ Help ▼
egion Configuration		Related	Links: Back To Find/List
]] Save 🗙 Delete 省 Reset 🕂 Add New 🛛			
Region Information			
ame* Region_Br1_DSPfarm_Transcoder			
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
Default	Keep Current Setting 💌	Keep Current Setting	Keep Current Setting 💌
Region Br1 Phones Analog Region Br1 DSPfarm		O Use System Default	
Region_Br1_DSPfarm_Conference		O None	
Region_Br1_DSPfarm_Transcoder 📃 💌		O kbps	
Save Delete Reset Add New			

**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.

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Enterprise 1 HQ Cisco Unified CM Example Configuration

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

alialia cisco		Navigation Cisco Unified CM Administration 💌 🤇
	For Cisco Unified Communications Solutions	admin About Logou
System 👻	Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management	 ■ Bulk Administration ■ Help
Region C	onfiguration	Related Links: Back To Find/List 💽 🤇
🔜 Save	🗙 Delete 🏻 🎦 Reset 🖧 Add New	
-	nformation egion_Br1_Phones_IP	

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
rE: 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 💌	 Keep Current Setting Use System Default None kbps 	Keep Current Setting 💌

Save Delete Reset Add New



(i) *- 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.

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

Cisco Unified CM Admin For Cisco Unified Communications So		Navigation C	isco Unified CM Administration
ystem 👻 Call Routing 👻 Media Resources 👻 Voice	e Mail 👻 Device 👻 Application 👻 I	Jser Management 👻 🛛 Bulk Administratio	
egion Configuration		Related	Links: Back To Find/List
🚽 Save 🗙 Delete			
Region Information			
Jame* Region_HQ_DSPfarm			
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
Default 🔺	Keep Current Setting 🛉	 Keep Current Setting 	Keep Current Setting 💌
Region Br1 Phones Analog Region_Br1_DSPfarm		🔿 Use System Default	
Region_Br1_DSPfarm_Conference		O None	
Region_Br1_DSPfarm_Transcoder		C kbps	
Save Delete Reset Add New			
i) *- 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.

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Enterprise 1 HQ Cisco Unified CM Example Configuration

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

Cisco Unified CM Admin For Cisco Unified Communications Solution		Navigation C	isco Unified CM Administration 💌 🕻 admin About Logou
System 👻 Call Routing 👻 Media Resources 👻 Voice M	Mail ▼ Device ▼ Application ▼ 0	Jser Management 👻 Bulk Administration	n ▾ Help ▾
Region Configuration		Related	Links: Back To Find/List 💽 🕻
🔚 Save 🗙 Delete Peset 🕂 Add New 👘			
-Region Information Name* Region_HQ_DSPfarm_Conference			
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 NOTE: Regions(s) not displayed	G.729 Use System Default	384 Use System Default	Use System Default Use System Default
-Modify Relationship to other Regions			
Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
Default	Keep Current Setting	 Keep Current Setting 	Keep Current Setting 💌
Region Br1 Phones Analog		C Use System Default	
Region Br1 DSPfarm Conference		O None	
Region_Br1_DSPfarm_Transcoder		C kbps	
Save Delete Reset Add New			
(i) *- indicates required item.			
(i) **The Audio Codec selection determines bandw regions and can be used interchangeably.	vidth only. The G.711 and G.722 o	odecs both result in a maximum ban	dwidth of 64 Kbps between

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

rstem 👻 Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help 👻	Cisco Unified CM Admin For Cisco Unified Communications Solo		Naviyation <mark>ju</mark>	isco Unified CM Administration
Save	stem 👻 Call Routing 👻 Media Resources 👻 Voice	Mail 👻 Device 👻 Application 👻 U	ser Management 👻 🛛 Bulk Administratio	
Region Information arme * Region_HQ_DSPfarm_Transcoder Region_HQ_DSPfarm_Transcoder G.711 384 Use System Default Region_HQ_DSPfarm_Transcoder G.711 384 Use System Default 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 NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default Pofault Regions Audio Codec Video Call Bandwidth Link Loss Type Region_Br1_Phones_Analog Keep Current Setting © Keep Current Setting © Use System Default Region_Br1_DSPfarm_Conference Image: Setting © Image: Setting © Image: Setting © Image: Setting Region_Br1_DSPfarm_Transcoder Image: Setting Image: Setting Image: Setting Image: Setting	gion Configuration		Related	Links: Back To Find/List
Ame * Region_HQ_DSPfarm_Transcoder Region Relationships 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 Addify Relationship to other Regions Audio Codec Video Call Bandwidth Link Loss Type Region_Br1_Phones Analog Keep Current Setting © Keep Current Setting © Use System Default Region_Br1_DSPfarm_Conference Imaging Br1_DSPfarm_Transcoder © Liskps System Default	🕽 Save 🗙 Delete Paset 🕂 Add New 👘			
Region Relationships 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 NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default Modify Relationship to other Regions	2			
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 NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default NoTE: Regions(s) not displayed Use System Default Use System Default Use System Default NoTE: Regions(s) not displayed Use System Default Use System Default Use System Default NoTE: Regions(s) not displayed Use System Default Use System Default Link Loss Type Default Region_Br1_DSPfarm_Conference Keep Current Setting Keep Current Setting Keep Current Setting Region_Br1_DSPfarm_Transcoder Image: Conference Image: Conference Image: Conference Image: Conference Image: Conference Region_Br1_DSPfarm_Transcoder Image: Conference <	ame* Region_HQ_DSPfarm_Transcoder			
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 NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default NOTE: Regions(s) not displayed Use System Default Use System Default Use System Default NoTE: Regions(s) not displayed Use System Default Use System Default Use System Default Region Br1 Phones Analog Keep Current Setting © Keep Current Setting Keep Current Setting © Use System Default Region_Br1_DSPfarm_Conference Image: Conference © Mone © Mone © Mone Region_Br1_DSPfarm_Transcoder Image: Conference © Mone © Mone © Mone Region_Br1_DSPfarm_Transcoder Image: Conference Image: Conference © Mone © Mone	egion Relationships			
Region_HQ_Phones_IP G.711 384 Use System Default Region_Wan G.729 384 Use System Default IOTE: Regions(s) not displayed Use System Default Use System Default Use System Default IOTE: Regions(s) not displayed Use System Default Use System Default Use System Default IootE: Regions(s) not displayed Use System Default Use System Default Use System Default IootE: Regions Audio Codec Video Call Bandwidth Link Loss Type Default Keep Current Setting © Keep Current Setting © Current Setting Region_Br1_DSPfarm_Conference Ioone © None © Mone Region_Br1_DSPfarm_Transcoder Ioone © Mone Expose Ioone Ioone Ioone Ioone Ioone Region_Br1_DSPfarm_Transcoder Ioone Ioone Ioone Ioone	Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Wan G.729 384 Use System Default OTE: Regions(s) not displayed Use System Default Use System Default Use System Default odify Relationship to other Regions Default Regions Default Region_Br1 Phones Analog Region_Br1_DSPfarm Region_Br1_DSPfarm_Conference Region_Br1_DSPfarm_Transcoder	Region_HQ_DSPfarm_Transcoder	G.711	384	Use System Default
OTE: Regions(s) not displayed Use System Default Link Loss Type Call Bandwidth Link Loss Type Keepion_Br1_DSPfarm Conference Region_Br1_DSPfarm_Transcoder	Region_HQ_Phones_IP	G.711	384	Use System Default
Regions Audio Codec Video Call Bandwidth Link Loss Type Default Region Br1 Phones Analog Region_Br1_DSPfarm Region_Br1_DSPfarm_Transcoder Region_Br1_DSPfarm_Transcoder Kbps 	Region_Wan	G.729	384	Use System Default
Regions Audio Codec Video Call Bandwidth Link Loss Type Default Region Br1 Phones Analog Region_Br1_DSPfarm C Use System Default C None Region_Br1_DSPfarm_Transcoder C kbps 	OTE: Regions(s) not displayed	Use System Default	Use System Default	Use System Default
Default Region Br1 Phones Analog Region_Br1_DSPfarm Region_Br1_DSPfarm_Conference Region_Br1_DSPfarm_Transcoder	lodify Relationship to other Regions			
Region_Br1 Phones Analog C Use System Default Region_Br1_DSPfarm_Conference C None Region_Br1_DSPfarm_Transcoder C Ise System Default C None C Ise System Default C None C Ise System Default	Regions	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_Br1_DSPfarm_Conference C None C System Default C None Region_Br1_DSPfarm_Transcoder C None C kbps		Keep Current Setting 💌	🛛 💿 Keep Current Setting	Keep Current Setting 💌
Region_Br1_DSPfarm_Conference C None Region_Br1_DSPfarm_Transcoder C Market C kbps			O Use System Default	
	Region_Br1_DSPfarm_Conference,		O None	
Save Delete Reset Add New	Region_Br1_DSPfarm_Transcoder 📃		C kbps	
	Save Delete Reset Add New			

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

cisco Unified CM Adm For Cisco Unified Communications		Navigation	Cisco Unified CM Administration 💌 admin About Loga
rstem 👻 Call Routing 👻 Media Resources 👻 Vo	oice Mail 👻 Device 👻 Application	👻 User Management 👻 Bulk Administrat	tion 🔻 Help 👻
gion Configuration		Relate	d Links: Back To Find/List 💽
] Save 🗙 Delete 省 Reset 🕂 Add New			
Region Information			
ame* Region_HQ_Phones_Analog			
egion Relationships			
Region	Audio Codec	Video Call Bandwidth	Link Loss Type
Region_HQ_Phones_IP	G.711	384	Use System Default
Region_Wan	G.711	384	Use System Default
Region Br1 Phones Analog	G.711	384	Use System Default
NOTE: Regions(s) not displayed	Use System Default	Use System Default	Use System Default
1odify Relationship to other Regions			
Regions	Audio Code	ec Video Call Bandwidth	Link Loss Type
Default	 Keep Current Setti 	ng 💌 💿 Keep Current Setting	Keep Current Setting 💌
Region Br1 Phones Analog Region_Br1_DSPfarm		C Use System Default	
Region_Br1_DSPfarm_Conference	-1	O None	
Region_Br1_DSPfarm_Transcoder	•	C kbps	
Save Delete Reset Add New			
•- indicates required item.			
**The Audio Codec selection determines ba regions and can be used interchangeably.	andwidth only. The G.711 and G.7	22 codecs both result in a maximum ba	andwidth of 64 Kbps between

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

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions	Navigation Cisco Unified CM Administration 💌 C admin About Logou
System 👻	Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻	Application ▼ User Management ▼ Bulk Administration ▼ Help ▼
Region Co	onfiguration	Related Links: Back To Find/List 💽 🕒
🔚 Save	🗙 Delete 🍟 Reset 🖧 Add New	
-	nformation	

Name*	Region_HQ_Phones_IP
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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
E: Regions(s) not displayed	Use System Default	Use System Default	Use System Default

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 💌	 Keep Current Setting Use System Default None Mone 	Keep Current Setting 💌

– Save Delete Reset Add New

(i) *- indicates required item.

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**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.

Enterprise 1 HQ Cisco Unified CM Example Configuration

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

cisco	Cisco Unified CM Administration	Navigation Cisco Unified CM Administration 💌 🤇 admin About Logou								
System 👻	Call Routing ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Application ▼ U	iser Management → Bulk Administration → Help →								
Region C	onfiguration	Related Links: Back To Find/List 🔍 C								
📄 Save	🔚 Save 🗙 Delete 🍟 Reset 🕂 Add New									
	nformation egion_Wan									

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

—Modify Relationship to other Regions ——

-Region Relationships

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 💌		Keep Current Setting 💌		

- Save Delete Reset Add New

(i) *- 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.

System: Device Pool Parameters

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

Figure 23 System Device Pool Cisco Unified CM Administration Window

- 010		Cisco Unified CM Administration 💌								
CIS	• For Cisco Unified Communications S	olutions		admin About	Logou					
System	▼ Call Routing ▼ Media Resources ▼ Vo	ice Mail 👻 Device 👻 Application 👻	🗸 User Management 👻 Bulk Administrati	on 👻 Help 👻						
Find and List Device Pools										
异 Add New 🔠 Select All 🗮 Clear All 🙀 Delete Selected										
-										
-Statu										
18	records found									
Device Pool (1 - 8 of 8) Rows per Page 50 🔽										
Find D	evice Pool where Device Pool Name	💌 begins '	with 🗾 Find	Clear Filter 🛛 🕂 📼						
Find D	evice Pool where Device Pool Name Name ^	Cisco Unified CM Group	with Find Region	Clear Filter 🔂 📼 Date/Time Group	Сору					
		,			Сору					
	Name 📤	Cisco Unified CM Group	,	Date/Time Group						
	Name [▲] Default	Cisco Unified CM Group Default	Region	Date/Time Group						
	Name * Default DevicePool Br1 Analog Phones	Cisco Unified CM Group Default Default	Region Default Region Br1 Phones Analog	Date/Time Group CMLocal CMLocal	6					
	Name ▲ Default DevicePool Br1 Analog Phones DevicePool Br1 DSPfarm	Cisco Unified CM Group Default Default Default	Region Default Region Br1 Phones Analog Region Br1 DSPfarm	Date/Time Group CMLocal CMLocal CMLocal	6 6 6					
	Name ▲ Default DevicePool Br1 Analog Phones DevicePool Br1 DSPfarm DevicePool Br1 IP Phones	Cisco Unified CM Group Default Default Default Default	Region Default Region Br1 Phones Analog Region Br1 DSPfarm Region Br1 Phones IP	Date/Time Group CMLocal CMLocal CMLocal CMLocal	6 6 6					
	Name ▲ Default DevicePool Br1 Analog Phones DevicePool Br1 DSPfarm DevicePool Br1 IP Phones DevicePool HQ Analog Phones	Cisco Unified CM Group Default Default Default Default Default Default Default	Region Default Region Br1 Phones Analog Region Br1 DSPfarm Region Br1 Phones IP Region HQ Phones Analog	Date/Time Group CMLocal CMLocal CMLocal CMLocal CMLocal CMLocal						

Enterprise	1 HQ	Cisco	Unified	СМ	Example	Configuration	on

Figure 24 System Device Pool Default Cisco Unified CM Administration Window

	d CM Administrati	on		Navigation Cisco U	nified CM Ad	ministration	<u> </u>
CISCO For Cisco Unified Co	ommunications Solutions				admin	About L	logou
System 👻 Call Routing 👻 Media	Resources 👻 Voice Mail 👻 De	evice 👻 Application 👻	User Management 👻	Bulk Administration 👻 🕴	Help 🔻		
Device Pool Configuration				Related Links:	Back To F	ind/List	• 0
🔚 Save 🗙 Delete 🗋 Copy	🛉 🎦 Reset 🕂 Add New						
— Status (i) Status: Ready							
— Device Pool Information — Device Pool: Default (3 memb	ers**)						
- Device Pool Settings Device Pool Name* Cisco Unified Communications M Calling Search Space for Auto-re Reverted Call Focus Priority			•				
-Roaming Sensitive Settings							
Region*	CMLocal	<u> </u>					
Media Resource Group List	Default < None >	<u> </u>					
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 Inform	nation****						
Device Mobility Calling Search Sp	ace < None >		•				
AAR Calling Search Space	< None >		•				
AAR Group	< None >		•				
– Save Delete Copy R	eset Add New						
(i) *- indicates required item.							
•	have to be reset when this dev	vice pool is updated. T	'o see a detailed list	of these devices and ot	her depende	ncies, click o	n
(i) ***leave blank to use defa	ult.						
(i) ****These three paramete	ers will overwrite device level s	ettings when device i	s roaming and in the	same device mobility gr	oup.		

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System Device Pool-DevicePool Branch 1 Analog Phones Cisco Unified CM Administration Window Figure 25 Navigation Cisco Unified CM Administration 💌 🖸 **Cisco Unified CM Administration** ahaha cisco For Cisco Unified Communications Solutions admin Logo Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 System -Bulk Administration 👻 Help -**Device Pool Configuration** Related Links: Back To Find/List • 🔚 Save 🗶 Delete 🕞 Copy 嗋 Reset 🕂 Add New -Status (i) 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 -AAR Calling Search Space < None > • AAR Group • |< None > – Save Delete Copy Reset Add New (i) *- 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 **i**) Dependency Records. ***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.

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	Enterprise	1 HQ	Cisco	Unified	СМ	Example	Configuration
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Figure 26 System Device Pool-DevicePool Branch 1 DSP Farm Cisco Unified CM Administration Window

01500								dministrati	
System - Call Routing - Medi			Device 🔻	Application 👻	User Management 🔻	- Bulk Administration 👻		About	Logou
Device Pool Configuration						Related Link		Find/List	•
🔚 Save 🗙 Delete 🗋 Cop	oy 🎦 Reset I	🕂 Add New							
- Status i Status: Ready									
— Device Pool Information Device Pool: DevicePool_Br1_	_DSPfarm(3 me	embers**)							
— Device Pool Settings Device Pool Name*		DevicePool	_Br1_DSP	farm					
Cisco Unified Communications I	Manager Group'	* Default			•				
Calling Search Space for Auto-r	registration	< None >			•				
Reverted Call Focus Priority		Default			•				
Date/Time Group*	CMLocal			•					
Region*	Region_Br1_	DSPfarm		•					
Media Resource Group List	Br1 HW MRG	L		•					
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 Info									
Device Mobility Calling Search 9		,			•				
AAR Calling Search Space	< None >								
AAR Group	< None >				-				
– Save Delete Copy I	Reset Add N	lew							
(i) *- indicates required item									
**Number of devices that		et when this	device no	ol is updated. 1	To see a detailed lis	t of these devices and r	other denen	dencies, cli	ck on
Dependency Records.									
(i) ***leave blank to use def	fault.								
(i) ****These three paramet	ters will overwri	te device lev	el settings	s when device i	s roaming and in th	e same device mobility (aroup.		

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Figure 27 System Device Pool-DevicePool Branch 1 IP Phones Cisco Unified CM Administration Window Navigation Cisco Unified CM Administration 💌 🖸 **Cisco Unified CM Administration** ahaha cisco For Cisco Unified Communications Solutions admin Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 System -Bulk Administration 👻 Help -**Device Pool Configuration** Related Links: Back To Find/List • 🔚 Save 🗶 Delete 🕞 Copy 嗋 Reset 🕂 Add New -Status (i) 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 -AAR Calling Search Space < None > • AAR Group < None > • – Save Delete Copy Reset Add New (i) *- 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 **i**) Dependency Records. 273769 (\mathbf{i}) ***leave blank to use default. **These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

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

cisco	Cisco Ul For Cisco Un				ation			Navigation Cisc	o Unified CM A		
System 👻 🤇	Call Routing 👻	Media F	esources 👻	Voice Mail 👻	Device 🔻	Application 👻	User Management 🔹	Bulk Administration 🖣		About	Logoc
Device Poo	l Configura	tion						Related Lin	iks: Back To I	Find/List	• •
🔒 Save	🗙 Delete [🗋 Сору	eset 省	🕂 Add Nev	v					_	
-Status i Status:	: Ready										
	ol Informatio : DevicePoo		alog_Phon	es (3 membe	rs**)						
Calling Sear		Auto-reg	_			og_Phones	V V				
— Roaming S Date/Time (ensitive Set Group*	tings —	CMLocal			•					
Region*				PhonesAn	alog						
Media Reso	urce Group Li	ist	HQ HW MR	GL		•					
Location			Hub_HQ			•					
Network Lo	cale		< None >			•					
SRST Refere	ence*		SRST_Ent1	_Br1		•					
Connection	Monitor Dura	ition***									
Single Butto	on Barge*		Default			•					
Join Across	Lines*		Default			•					
Physical Loo	cation		< None >			•					
Device Mobi	ility Group		< None >			•					
-Device Mo	bility Relate ility Calling Se						_				
	Search Space										
AAR Group		-	< None				•				
– Save [Delete Co	py Re	set Add	New							
(i) *- india	cates require	d item.									
	nber of device dency Record		ave to be re	set when this	s device po	ol is updated. [.]	To see a detailed lis	t of these devices and	l other depend	lencies, clic	:k on
(i) ***lea	ave blank to u	ıse defau	ılt.								20
(i) ****T	hese three pa	arameter	s will overw	rite device le	vel settings	when device	is roaming and in th	e same device mobilit	y group.		273770

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System Device Pool-DevicePool HQ DSP Farm Cisco Unified CM Administration Window Figure 29 Navigation Cisco Unified CM Administration 💌 🤇 **Cisco Unified CM Administration** ahaha cisco For Cisco Unified Communications Solutions admin Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 System -Bulk Administration 👻 Help 🔻 **Device Pool Configuration** Related Links: Back To Find/List • 🔚 Save 🗶 Delete 🕞 Copy 嗋 Reset 🕂 Add New -Status (i) 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 -AAR Calling Search Space < None > • AAR Group -< None > - Save Delete Copy Reset Add New (i) *- 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 **i**) Dependency Records. (\mathbf{i}) ***leave blank to use default. 273771 **These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

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

cisco	Cisco Unifie			tion				Navigation	Cisco U		dministrat	
System 👻	Call Routing 👻 Media			Device 🔻	Application 👻	User Manageme	ent 👻	Bulk Administratio	on 👻 I		About	Logou
Device Po	ool Configuration							Related	Links	Back To	Find/List	•
📄 Save	🗙 Delete 📄 Copy	省 Reset	🕂 Add New									
— Status — (i) Statu	ıs: Ready											
	Pool Information pol: DevicePool_HQ_I	P_Phones (12	2 members**)								
Device Po	Pool Settings ool Name* fied Communications M	anager Group	DevicePool	_HQ_IP_P	hones	V						
Calling Se	arch Space for Auto-re	gistration	< None >			•						
Reverted	Call Focus Priority		Default			•						
Date/Time Region* Media Res Location Network L SRST Refe Connectio Single But Join Acros Physical L	source Group List Locale erence* on Monitor Duration*** tton Barge* ss Lines*	CMLocal Region_HQ_ HQ HW MRG Hub_HQ < None > SRST_Ent1_ Default Default < None > < None >	L		• • • • • • • • •							
Device Mo	1obility Related Inforr obility Calling Search Sp ng Search Space p		>									
(i) *- ini	Delete Copy R dicates required item. umber of devices that h			device poo	ol is updated. 1	To see a detaile	d list o	f these devices	and ot	her depend	dencies. cli	ck on
	endency Records. eave blank to use defa *These three paramete	ult.									,	·
\smile												

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Navigation Cisco Unified CM Administration 💌 🖸 **Cisco Unified CM Administration** ahaha cisco For Cisco Unified Communications Solutions admin Logo Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 System -Bulk Administration 👻 Help 🔻 **Device Pool Configuration** Related Links: Back To Find/List • 🔚 Save 🗶 Delete 🕞 Copy 嗋 Reset 🕂 Add New -Status (i) 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 -AAR Calling Search Space < None > • AAR Group -< None > - Save Delete Copy Reset Add New (i) *- 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 **i**) Dependency Records. 273773 (\mathbf{i}) ***leave blank to use default. **These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.

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

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Enterprise 1 HQ Cisco Unified CM Example Configuration

System: Location Parameters

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

Figure 32 System Location Cisco Unified CM Administration Window

	Cisco Unified CM	Administration		Navigation Cisco	Unified CM Administration 💌 🔤
cisco	For Cisco Unified Communica	tions Solutions			admin About Logo
System 👻 🛛 C	all Routing 👻 Media Resources	▼ Voice Mail ▼ Device ▼	Application 👻 User Manag	jement 👻 Bulk Administration 👻	Help 👻
Find and Li	st Locations				
Add New	/ Select All Clear All				
— Status —					
(i) 5 recor	ds found				
Locations	(1 - 5 of 5)				Rows per Page 50 💌
Find Locatio	ns where Location	begins with 💌	Find Clea	ar Filter 🛛 🕂 📼	
	Location *	Audio	Bandwidth	Video Bandwidth	Сору
	Hub Br1	85		NONE	ß
	<u>Hub HQ</u>	110		NONE	ß
	<u>Hub None</u>	UNLIMITED		UNLIMITED	ß
	<u>Trunk Br1</u>	85		NONE	ū.
	<u>Trunk HQ</u>	110		NONE	ß
Add New	Select All Clear All	Delete Selected			

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cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions		Navigation Cisco Unified CM A	dministration 💌 🕻
System 👻	Call Routing Media Resources Voice Mail Device Application	on 👻 User Management 👻	Bulk Administration 👻 Help 👻	
Location	Configuration		Related Links: Back To	Find/List 🔽 🤆
🔚 Save	🗙 Delete 📋 Copy 🕂 Add New 👌 Resync Bandwidth			
— Status — (i) Statu	s: Ready			
— Location Name* Hu	Information ub_Br1			
Audio Ban If the aud Video Ca	IIs Information dwidth* C Unlimited © 85kbps io quality is poor or choppy, lower the bandwidth setting. For ISDN, IIs Information dwidth* © None C Unlimited Ckbps	use multiples of 56 kbps o	ır 64 kbps.	
-Location	RSVP Settings			
	Location		RSVP Setting	
NOTE: Lo	ocation(s) not displayed	Use System Default		
-Modify S	etting(s) to Other Locations			
, iouily o	Location		RSVP Setting	
Hub_Br1 Hub_HQ Hub_No Trunk Br Trunk H0	ne 1	Use System Default	Y	
- Save	Delete Copy Add New Resync Bandwidth			
	dicates required item.			273775

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

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	Enterprise 1	I HQ Cisco	Unified CN	/I Example	Configuration
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Figure 34 System Location Hub HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration	Navigation Cisco Unified CM Administration 💌 🕻 admin About Logou
System ▼ Call Routing ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Applicat	
Location Configuration	Related Links: Back To Find/List 👤 🤄
🔚 Save 🗙 Delete 🗋 Copy 🕂 Add New 📎 Resync Bandwidth	
- Status Status: Ready	
-Location Information Name* Hub_HQ	
- Audio Calls Information Audio Bandwidth* O Unlimited ⓒ 110kbps 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 –	9
(i) *- indicates required item.	273776

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cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions		Navigation Cisco Unified CM Administration 💌 🕻 admin About Logou
System 👻	Call Routing • Media Resources • Voice Mail • Device • Applic	ation 👻 User Management 👻	Bulk Administration 👻 Help 👻
Location	Configuration		Related Links: Back To Find/List 💽 🤇
🔚 Save	🗋 Copy 🕂 Add New 👌 Resync Bandwidth		
— Status — (i) Statu	is: Ready		
— Location Name*	Information		
Audio Ban If the aud	Ills Information Idwidth [*] O Unlimited O kbps Iio quality is poor or choppy, lower the bandwidth setting. For ISDN Ills Information Idwidth [*] O None O Unlimited O kbps	I, use multiples of 56 kbps or	64 kbps.
Location	Location		RSVP Setting
NOTE: Lo	ocation(s) not displayed	Use System Default	
—Modify S	etting(s) to Other Locations		
	Location		RSVP Setting
Hub_Br: Hub_HQ Hub_No Trunk Br Trunk H) ne r1	Use System Default	
Save	Copy Add New Resync Bandwidth		
(i) *- ind	dicates required item.		273771

Figure 35 System Location Hub None Cisco Unified CM Administration Window

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	Enterprise 1	I HQ Cisco	Unified CN	/I Example	Configuration
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Figure 36 System Location-Location Trunk Branch 1 Cisco Unified CM Administration Window

Cisco Unified CM Administration	Navigation Cisco Unified CM Administration 🗾 🤇
System Call Routing Media Resources Voice Mail Device App	admin About Logou plication ▼ User Management ▼ Bulk Administration ▼ Help ▼
Location Configuration	Related Links: Back To Find/List
🔚 Save 🗙 Delete 📋 Copy 🕂 Add New 👌 Resync Bandwidth 👘	
- Status i Status: Ready	
-Location Information Name* Trunk Br1	
Audio Calls Information Audio Bandwidth* O Unlimited © 85kbps If the audio quality is poor or choppy, lower the bandwidth setting. For IS	DN, use multiples of 56 kbps or 64 kbps.
-Video Calls Information Video Bandwidth [*] ⊙ None ○ Unlimited ○kbps	
-Location RSVP Settings	
Location	RSVP Setting
NOTE: Location(s) not displayed	Use System Default
—Modify Setting(s) to Other Locations————————————————————————————————————	
Location	RSVP Setting
Hub_Br1 Hub_HQ Hub_None Trunk Br1 Trunk HQ	Use System Default
– Save Delete Copy Add New Resync Bandwidth –	
i *- indicates required item.	273778

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cisco	Cisco Unified CM Administration Navigation Cisco Unified CM Administration 🔽 C For Cisco Unified Communications Solutions admin About Logou
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- i Statu	us: Ready
— Location Name* T	a Information
Audio Bar If the aud — Video Ca Video Bar	alls Information ndwidth* O Unlimited © 110 kbps dio quality is poor or choppy, lower the bandwidth setting. For ISDN, use multiples of 56 kbps or 64 kbps. alls Information ndwidth* © None O Unlimited O kbps
-Location	n RSVP Settings
NOTE: L	ocation(s) not displayed Use System Default
—Modify S	Setting(s) to Other Locations
	Location RSVP Setting
Hub_Br Hub_H(Hub_No Trunk B Trunk H	Q
- Save	Delete Copy Add New Resync Bandwidth
	ndicates required item.

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

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System: SRST Parameters

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

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

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions	Navigation Cisco Unified CM Administration 💌 🕻 admin About Logou
System 👻	Call Routing ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Application	
SRST Refe	erence Configuration	Related Links: Back To Find/List 💽 🤇
🔜 Save	🗙 Delete [ြ Copy 省 Reset 🕂 Add New	
— Status — (i) Statu	s: Ready	
	ference Status rence: SRST_Ent1_Br1 (used by 13 devices)	
-SRST Ref	ference Information	
Name*	SRST_Ent1_Br1	
Port*	2000	
IP Addres	s* 10.40.103.1	
SIP Netwo	ork/IP Address	
SIP Port*	5060	
SRST Cert	ificate Provider Port* 2445	
🗆 Is SRS	T Secure?	
- Save	Delete Copy Reset Add New	C C C C C C C C C C C C C C C C C C C
U *- ind	dicates required item.	08767

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Configuring the Cisco Unified CM Call Routing Parameters

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

- Call Routing: Route/Hunt Parameters, page 63
- Call Routing: Class of Control Parameters, page 68

Call Routing: Route/Hunt Parameters

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

Figure 39 Call Routing RouteHunt Route Pattern Cisco Unified CM Administration Window

aha		ified CM Administration	Navi	gation Cisco l	Unified CM Administration	n 🔽 🤇		
CISC	• For Cisco Unif	ied Communications Solutions			admin About	Logou		
System	▼ Call Routing ▼	Media Resources 👻 Voice Mail 👻 Device 👻 🗚	Application 👻 User Management 👻 Bulk Adn	ninistration 👻	Help 🔻			
Find a	nd List Route Patt	erns						
Add New 🔛 Select All 🔛 Clear All 💥 Delete Selected								
-	- Status i 4 records found							
Route Patterns (1 - 4 of 4) Rows per Page 50								
Route	e Patterns – (1 - 4	of 4)			Rows per Page 5	0 💌		
	e Patterns (1 - 4) pute Patterns where		Find Clear Filter	ф <u>–</u>	Rows per Page 50	0 🔽		
	•		Find Clear Filter	Route Filter	Rows per Page 50	Cop [.]		
Find Ro	oute Patterns where	Pattern 💌 begins with 💌						
Find Ro	pute Patterns where Pattern ^	Pattern Description	Partition		Associated Device	Cop		
Find Ro	pute Patterns where Pattern ^ 9.1XXXXXXXXXX	Pattern begins with Description RP Ent1-HQ IP Phone LongDistance	Partition Partition IP		Associated Device 10.10.11.151	Cop		
Find Ro	Pattern Pattern Pattern Pattern Pattern Pattern Pattern Pattern	Pattern begins with Description RP Ent1-HQ IP Phone LongDistance RP Ent1-HQ Analog Phone LongDistance	Partition Partition-HQ Phones IP Partition-HQ Phones Analog		Associated Device 10.10.11.151 10.10.11.151	Cop [.]		

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

ahaha Cis	co Unified CM Adminis	tration		Navigation Cisco) Unified CM A	dministration 💌 🕻
CISCO	Cisco Unified Communications Solutio	ns			admin	About Logou
System 👻 Call R	outing 👻 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 i Status: Rea	dy					
—Pattern Definit	ion					
Route Pattern*	9.1xxxxxxxx					
Route Partition	Partition-HQ_Phones_IP	<u> </u>				
Description	RP Ent1-HQ IP Phone LongDistan	ce				
Numbering Plan	Not Selected	~				
Route Filter	< None >	~				
MLPP Precedenc	Thoradic	▼	,			
	List* 10.10.11.151	▼ (<u>Edit</u>)			
Route Option	Route this pattern					
	C Block this pattern No Error					
Call Classificatio	Tourise	_	_			
	Override 🗹 Provide Outside Dial Tone	Allow Overlap Sending	Urgent Priority			
Authorization Le	ed Authorization Code					
_	0					
🗆 Require Clien	t Matter Code					
—Calling Party T	ransformations					
	arty\'s External Phone Number Mask					
Calling Party Tra	insform Mask					
Prefix Digits (Ou	tgoing Calls)					
Calling Line ID P	resentation* Default	•				
Calling Name Pre	esentation* Default					
	ty Transformations		_			
	ID Presentation* Default		▼			
Connecteu Nam	e Presentation* Default		▼			
—Called Party Tr	ansformations					
Discard Digits	PreDot		•			
Called Party Tra	nsform Mask					
Prefix Digits (Ou	tgoing Calls) 6					
— ISDN Network	-Specific Facilities Information Eleme	nt				
	Protocol Not Selected	▼				
Carrier Identifica						
Network Service	Servi	e Parameter Name	Se	ervice Parameter Value		
Not Selected		t Exist >				
SaveDelet	e Copy Add New					
(i) *- indicates	s required item.					07870

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Figure 41	1 Call R	outing Route	Hunt Route Patte	ern RP Ent1	HQ Analog Ph	none LongDistand	e Administr	ation Window
cisco			ministration			Navigation Ci	sco Unified CM A	dministration 💌 🕻
	For Cisco Unified			- Application	- Licer Management	t 👻 Bulk Administration		About Logou
	-	_	Voice Mail + Device +	Application	• Oser Managemen	_		
Route Pati	tern Configuratio	on				Relate	ed Links: Back	To Find/List 🔽 🤆
🔚 Save	🗙 Delete 🗋 C	opy 🕂 Add Nev	/					
— Status —								
(i) Status	s: Ready							
—Pattern D								
Route Patt	12:4000	****						
Route Part	tition Partitio	on-HQ_Phones_A	nalog	•				
Descriptior	n RP Ent	1-HQ Analog Pho	ne LongDistance					
Numbering		Selected		-				
Route Filte	er < Non	e >		$\overline{\mathbf{v}}$				
MLPP Prec	[Dordan			•				
Gateway/F	Route List* 10.10.	11.151		(Edit)				
Route Opt	ion 💿 Rou	ute this pattern						
	O Blo	ck this pattern N	o Error	•				
Call Classif	fication*	OffNet		•				
🗆 Allow D	evice Override 🗹	Provide Outside D	ial Tone 🗖 Allow Ove	erlap Sending	🗆 Urgent Priority			
🗆 Require	e Forced Authorizat	tion Code						
Authorizat	tion Level*	0						
🗆 Require	e Client Matter Cod	le						
-	arty Transformatio							
	lling Party\'s Extern		r Mask					
	rty Transform Mask	1						
	ts (Outgoing Calls)	1						
Calling Lin	e ID Presentation*	Default		•				
Calling Nar	me Presentation*	Default						
—Connecte	d Party Transform	nations ———						
Connected	d Line ID Presentat	ion* Default			-			
Connected	d Name Presentatio	on* Default			-			
—Called Pa	irty Transformatio	ins						
Discard Dig	gits	PreDot			•			
Called Part	ty Transform Mask							
Prefix Digit	ts (Outgoing Calls)	7						
— ISDN Net	twork-Specific Fa	cilities Informatio	on Element					
		- Not Selected		•				
Carrier Ide	entification Code							
Network S	ervice		Service Parameter	r Name		Service Parameter Val	ue	
Not Sel	ected		Not Exist >					
- Save	Delete Copy	Add New -						
								205
(i) *- ind	licates required ite	m.						273705

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

ahaha Cisc	o Unified CM Administration	Navigation Cisco Unified CM Administration 💌 🤇
CISCO For Cis	sco Unified Communications Solutions	admin About Logou
System 👻 🛛 Call Rou	uting ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Application ▼ User Manage	ement 👻 Bulk Administration 👻 Help 👻
Route Pattern Co	onfiguration	Related Links: Back To Find/List 💌 🤇
🔚 Save 🗙 Del	lete [Copy 🕂 Add New	
-Status		
(i) Status: Ready	/	
–Pattern Definitio	n	
Route Pattern*	9.1XXXXXXXXX	
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 Lis	st* 10.80.80.82 💽 (Edit)	
Route Option	Route this pattern	
	O Block this pattern No Error	
Call Classification*	* OffNet	
Allow Device Ov	verride 🗹 Provide Outside Dial Tone 🗔 Allow Overlap Sending 🗔 Urgent Prior	itv
	Authorization Code	,
Authorization Leve		
🗆 Require Client M	-	
-Calling Party Tra	nsformations —	
-	ty\'s External Phone Number Mask	
Calling Party Trans	sform Mask 415555XXXX	
Prefix Digits (Outg	oing Calls)	
Calling Line ID Pre	sentation* Default	
Calling Name Pres	entation* Default	
- Connected Dautu	Turneformations	
-Connected Party Connected Line ID) Presentation* Default	
	Presentation* Default	
-Called Party Trar	nsformations	
Discard Digits	PreDot 💌	
Called Party Trans	form Mask	
Prefix Digits (Outg	oing Calls) 7	
	pecific Facilities Information Element	
Network Service Pi		
Carrier Identificatio		
	Service Parameter Name	Service Parameter Value
Network Service		
- Save Delete	Copy Add New	
(i) *- indicates re	equired item.	
\smile		

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Figure 43	3 Call Routing RouteHunt Route Pattern RP Ent1 Br1 IP Ph	one LongDistance Administration Window
cisco	Cisco Unified CM Administration	Navigation Cisco Unified CM Administration 💌 🕻
	For Cisco Unified Communications Solutions Call Routing Media Resources Voice Mail Device Application User Manage	admin About Logou jement ▼ BulkAdministration ▼ Help ▼
	tern Configuration	Related Links: Back To Find/List 💌 G
📄 Save	🗙 Delete 📋 Copy 🕂 Add New	
— Status —		
i Status	s: Ready	
—Pattern D		
Route Patt	tern* 9.1XXXXXXXXX	
Route Part	tition Partition-Br1_Phones_IP 💌	
Descriptior	n RP Ent1-Br1 IP Phone LongDistance	
Numbering	g Plan Not Selected	
Route Filte	er < None >	
MLPP Prec	edence* Default	
Gateway/R	Route List* 10.80.80.82 💽 (Edit)	
Route Opt	ion 💿 Route this pattern	
	O Block this pattern No Error	
Call Classif	ification* OffNet	
🗆 Allow D	Device Override 🗹 Provide Outside Dial Tone 🗖 Allow Overlap Sending 🗖 Urgent Pric	prity
🗆 Require	e Forced Authorization Code	
Authorizat		
🗆 Require	e Client Matter Code	
i Koquire		
—Calling Pa	arty Transformations	
🗆 Use Ca	illing Party\'s External Phone Number Mask	
Calling Par	rty Transform Mask 415555XXXX	
Prefix Digit	ts (Outgoing Calls)	
Calling Lin	e ID Presentation*	
Calling Nar	me Presentation* Default	
—Connecte	d Party Transformations	
Connected	Line ID Presentation* Default	
Connected	d Name Presentation* Default	
	arty Transformations	
Discard Dig	gits PreDot 🗾	
Called Pari	ty Transform Mask	
Prefix Digit	ts (Outgoing Calls) 6	
	twork-Specific Facilities Information Element	
	Service Protocol 🛛 Not Selected	
Carrier Ide	entification Code	
Network S	Service Service Parameter Name	Service Parameter Value
Not Sel	ected <	
Save	Delete Copy Add New	~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
(i) * := ·	liastac required item	273707
• · · ind	dicates required item.	27

Call Routing: Class of Control Parameters

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

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

cisco	Cisco Unified CM Administration	Navigation Cisco Unified CM Administration 💌 C	
System 👻	Call Routing ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Application ▼ User Management ▼ B	ulk Administration 👻 Help 👻	
Find and L	st Partitions		
🕂 Add New 🔠 Select All 🔛 Clear All 💥 Delete Selected			
- Status			
Partition	(1 - 4 of 4)	Rows per Page 50 💌	
Find Partiti	n where Name 💽 begins with 💌 🛛 🗛 📼	1	
	Partition Name 🗕	Description	
	Partition-Br1 Phones Analog	Analog Phones	
	Partition-Br1 Phones IP	IP Phones	
	Partition-HQ_Phones_Analog	Analog Phones	
	Partition-HQ Phones IP	IP Phones	
Add Nev	Select All Clear All Delete Selected		

cisco	Cisco Unified CM Administratio For Cisco Unified Communications Solutions	n		Navigation Cisco Unified CM Administration 💌 🕻 admin About Logou	
System 👻	Call Routing 👻 Media Resources 👻 Voice Mail 👻 Devic	e 👻 Application 👻	User Management 👻	Bulk Administration 👻 Help 👻	
Partition	Configuration			Related Links: Back To Find/List 🗾 🤇	
📄 Save	🗙 Delete				
— Status — (i) Statu	ıs: Ready				
—Partition	Information				
Name*	Partition-Br1_Phones_Analog]			
Descriptio	on Analog Phones]			
Time Sche	edule < None >				
Time Zon	e 💿 Originating Device				
	C Specific Time Zone Greenwich Standard Time		V		
Save	Delete Reset Add New				
(i) *- in	dicates required item.				
					273709

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

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Enterprise 1 HQ Cisco Unified CM Example Configuration

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

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions			Navigation Cisco Unified CM Administration 💌	
System 👻	Call Routing • Media Resources • Voice Mail • Device •	Application 👻	User Management 👻		
Partition (Configuration			Related Links: Back To Find/List 🗾	G
🔜 Save	🗙 Delete				
-Status	s: Ready				-
Name* Description	dule < None >		Y		
- <u>Save</u>	Delete Reset Add New				273710
					0

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Figure 47	Call Routing Class of Control Partition-Partition HQ Phones Anal	log Administration Window
CIECO	Cisco Unified CM Administration For Cisco Unified Communications Solutions	Navigation Cisco Unified CM Administration 💌 C admin About Logou
System 👻 🤇	all Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻	Bulk Administration 👻 Help 👻
Partition C	onfiguration	Related Links: Back To Find/List 💽 G
🔚 Save	🗶 Delete 🏻 🖕 Reset 🖧 Add New	
Time Zone		

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Enterprise 1 HQ Cisco Unified CM Example Configuration

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

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions	Navigation Cisco Unified CM Administration 💌 🕻 admin About Logou
System 👻	Call Routing ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Application ▼ User Manager	
Partition	Configuration	Related Links: Back To Find/List 💽 G
🔜 Save	🗙 Delete 🍟 Reset 🖧 Add New	
— Status — (i) Statu	is: Ready	
Name* Descriptio	edule < None >	
(i) *- ind	dicates required item.	273712

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Navigation Cisco Unified CM Administration 💌 🖸 **Cisco Unified CM Administration** ahaha cisco For Cisco Unified Communications Solutions admin About Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help 💌 System 👻 Find and List Calling Search Spaces 🕂 Add New 🗰 Select All 📅 Clear All 💥 Delete Selected -Status (i) 5 records found Rows per Page 50 💌 Calling Search Space (1 - 5 of 5) Find Calling Search Space where CSS Name 💌 begins with 💌 Find Clear Filter ÷ ____ Description Сору CSS Name * CSS-Br1_Phones_Analog ß CSS-Br1 Phones Analog CSS-Br1_Phones_IP ß CSS-Br1 Phones IP CSS-HQ Phones Analog CSS-HQ_Phones_Analog ß ß CSS-HQ_Phones_IP CSS-HQ Phones IP Select All Clear All Delete Selected Add New

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

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

cisco				Administ ations Solution				Navigation Cisco		Administrati About	
System 👻	Call Routing	🔹 Media	a Resource:	s 👻 Voice Mail	- Device -	Application 👻	User Management 👻	Bulk Administration 👻			
Calling Se	earch Spac	e Config	uration					Related Lin	ks: Back To) Find/List	•
Rave	X Delete	Cor	y 🛟 Ad	d New							
— Status — i Statu	ıs: Ready										
	earch Spac										
Name*	CSS-Br1_										
Descriptio	on CSS-Br1_	Phones_/	Analog								
	artitions for Partitions**		ng Search	Space							
Selected F	Partitions	Partition Partition	I-Br1_Phoi	nes_Analog			*				
- Save	Delete	Сору	Add New]							
(i) *- ind	dicates requ	ired item.									4
(i) **Se	elected Parti	tions are	ordered by	/ highest priorit	У						273714

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

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions	Navigation Cisco Unified CM Administration 💌 🕻 admin About Logou
System 👻	Call Routing - Media Resources - Voice Mail - Device - Application - U	Jser Management 👻 Bulk Administration 👻 Help 👻
Calling Se	arch Space Configuration	Related Links: Back To Find/List 💽 C
📄 Save	🗙 Delete 📋 Copy 🕂 Add New	
— Status — i Statu:	s: Ready	
	earch Space Information	
Name*	CSS-Br1_Phones_IP	
Descriptio	ⁿ CSS-Br1_Phones_IP	
	Partitions for this Calling Search Space	*
- Save	Delete Copy Add New	<u>م</u>
	lected Partitions are ordered by highest priority	273715

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

cisco		Unified CM Ad		ation				Navigation Cisc	o Unified CM . admin	Administration 💌 🖸
System 👻		✓ Media Resources ✓		Device 🔻	Application 👻	User Manag	ement 👻	Bulk Administration 👻		About Logou
Calling Se	earch Spac	e Configuration						Related Lin	ks: Back To	Find/List 💽 🤆
🔚 Save	X Delete	Copy 🕂 Add Ne	W							
— Status — (i) Statu	ıs: Ready									
Name*	CSS-HQ_F	e Information Phones_Analog Phones_Analog								
	artitions for t Partitions**	this Calling Search Spa	ace ———							
Selected F	Partitions	♥ Partition-HQ_Phones_ Partition-Br1_Phones_ Partition-HQ_Phones_	Analog IP			×				
- Save	Delete	Copy Add New								
×	dicates requi elected Partit	ired item. ions are ordered by hig	hest priority							273716

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

diala cisco				dministr Ins Solutions	ation			Navigation Cisc	OUnified CM	Administrati	
System 👻	Call Routing	👻 Media F	Resources 👻	Voice Mail 🔻	Device 👻	Application 👻	User Management 👻	Bulk Administration 👻	Help 🔻		
Calling Se	earch Spac	e Configu	ation					Related Lin	ks: Back To	Find/List	•
📄 Save	X Delete	Copy	🕂 Add Ne	BW							
— Status — (i) Status	ıs: Ready										
	earch Spac	e Informati	on ———								
Name*	,	Phones_IP									
Descriptio	on CSS-HQ_	Phones_IP									
	Partitions for Partitions** Partitions	Partition-F Partition-E Partition-E		IP _Analog _IP			*				
- Save	Delete	Copy Ac	dd New 🛛 –								
ž	dicates requ elected Parti		dered by hi	ghest priority							273717
0											N

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Configuring the Cisco Unified CM Media Resources Parameters

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

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

Media Resources: Annunciator Parameters

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

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

cisco		nified CM Ad		ation			Navigation Cisco Unified CM		z Igou
System 👻	Call Routing 👻	Media Resources 👻	Voice Mail 👻	Device 🔻	Application 👻	User Management 👻	Bulk Administration 👻 Help 👻		
Annuncia	tor Configura	tion					Related Links: <mark>Back T</mark>	o Find/List 💌	•
🔜 Save	省 Reset								
— Status — (i) Statu:	s: Ready								
IP Address Server* Name* Descriptio Device Poo Location*	on Registered 5 10.40.97.2 10.40.97.2 ANN_2 n ANN_2_Ent 01* DevicePool	:1-HQ-CUCM _HQ_IP_Phones	ommunication	s Manager	40.40.97.2				

Media Resources: Conference Bridge Parameters

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

Figure 55 Media Resources Conference Bridges Cisco Unified CM Administration Window

cisco			n	Navigation Cisco Unifi	ed CM Administratic admin About	in 🔽 🕻 Logou		
System 👻	🕤 Call Routing 👻 Media Resources	▼ Voice Mail ▼ Device	e 👻 Application 👻 User Manag	ement 👻 Bulk Administration 👻 Helj	o 🕶			
Find and	d List Conference Bridges							
🕂 Add	🕂 Add New 🏢 Select All 🔛 Clear All 💥 Delete Selected 🏻 🎦 Reset Selected							
	3 records found							
Confer	rence Bridges (1 - 3 of 3)				Rows per Page	.0 💌		
Find Con	nference Bridges where Name	💌 begins with 💌	Find	Clear Filter 🔂 😑				
	Conference Bridge Name 🕇	Description	Device Pool	Status	IP Address	Copy		
<u>C</u>	FB 2	CFB_2-Ent1-HQ	Default	Registered with 10.40.97.2	10.40.97.2	ß		
	ON001AA29DF631	CFB-Ent1-Br1	DevicePool Br1 DSPfarm	Registered with 10.40.97.2	10.40.103.1	ß		
	ON111222333	CFB-Ent1-HQ	DevicePool HQ DSPfarm	Registered with 10.40.97.2	10.40.97.1	ß		
Add N	lew Select All Clear All	Delete Selected	Reset Selected					

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

Cisco Unified			ation			Navigation Cisco	Unified CM A admin	Administrati About	ion 💌 🤇
System 👻 Call Routing 👻 Me	dia Resources 👻	Voice Mail 👻	Device 🔻	Application 👻	User Management 👻	Bulk Administration 👻	Help 🔻		
Conference Bridge Configu	Iration			Related L	inks: Back To Find/	'List			•
🔚 Save 🗙 Delete 📔 C	opy 🎦 Reset	🕂 Add New	,						
Status									
i) Status: Ready									
Conference Bridge Informa Conference Bridge : CON001/		Coti Dri)							
-	ed with Cisco Ur	,	nications M	anager 10.40.	97.2				
IP Address 10.40.10				-					
• IOS Conference Bridge Info Conference Bridge Type*	Cisco IOS Enh	anced Confer	ence Brida						
Conference Bridge Name*			shee bhag	,					
Description	CFB-Ent1-Br1	51001							
Device Pool*									
Common Device Configuration	DevicePool_B	r1_USPtarm							
Location*				•					
	Hub_Br1			-					
Device Security Mode*	Non Secure C	onference Bri	dge	•					
Save Delete Copy	Reset Add	New							
 *- indicates required iter 	n.								

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

ababa	Cisco Unifi	ed CM Admin	istration			Navigation Cisco) Unified CM A	Administration	
cisco	For Cisco Unified	Communications Solu	itions				admin	About	Logou
System 👻	Call Routing 👻 Med	ia Resources 👻 Voice I	Mail 👻 Device 🖣	- Application 👻	User Management 👻	Bulk Administration 👻	Help 👻		
Conferen	ce Bridge Configu	ration		Related L	inks: Back To Find/	List			- C
📄 Save	🗙 Delete 🗋 Co	py 🎦 Reset 🕂 A	dd New						
— Status — i Statu	ıs: Ready								
	ion Registere	22333 (CFB-Ent1-HQ) d with Cisco Unified Co	ommunications I	Manager 10.40.	97.2				
Conference Conference Descriptio Device Po Common I Location*	ol* Device Configuration	Cisco IOS Enhanced C CON111222333 CFB-Ent1-HQ DevicePool_HQ_DSP	farm						
– <u>Save</u> j	Delete Copy	Reset Add New							273737

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Media Resources: Media Termination Point Parameters

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

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

	Cisco Unified CM Administration Navigation Cisco Unified CM Administration Cisco Unified CM Ad								
System	• 0	Call Routing 👻 Media Reso	urces 👻 Voice Mail 👻	Device - Application - Use	r Management 👻 Bulk Administration 👻	Help 🔻			
Find an	ıd Li	st Media Termination I	Points						
🕂 Ada	🗜 Add New 🌐 Select All 🔛 Clear All 💥 Delete Selected 🏻 🍟 Reset Selected								
	Status i 3 records found Media Termination Point (1 - 3 of 3) Rows per Page 50								
Find Me	edia ⁻	Termination Point where	Name 💌 begin	s with 👤	Find Clear Filter 🔂 🛥				
		Name 🕈	Description	Device Pool	Status	IP Address	Сору		
	MTP IDV2	MTP001AA29DF631	MTP-Ent1-Br1	DevicePool Br1 DSPfarm	Registered with 10.40.97.2	10.40.103.1	6		
	MTP IDV2	MTP111222333	MTP-Ent1-HQ	DevicePool HQ DSPfarm	Registered with 10.40.97.2	10.40.97.1	6		
	мтр	MTP_2	MTP_2-Ent1-HQ	<u>Default</u>	Registered with 10.40.97.2	10.40.97.2	Not Allowed		
Add	Add New Select All Clear All Delete Selected Reset Selected								

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

cisco	Cisco Unified CM Administration For Cisco Unified Communications Solutions	Navigation Cisco Unified CM Administration 💌 C
System 👻	Call Routing Media Resources Voice Mail Device Application User Management	
Media Ter	rmination Point Configuration	Related Links: Back To Find/List 💽 G
🔚 Save	🗙 Delete 📔 Copy 🎦 Reset 🕂 Add New	
— Status — () Statu	is: Ready	
Registrati IP Addres Media Ter	s 10.40.103.1 mination Point Type* Cisco IOS Enhanced Software Media Termination Point mination Point Name* MTP001AA29DF631 n MTP-Ent1-Br1	
- Save	Delete Copy Reset Add New dicates required item.	

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

cisco	Cisco Unifie For Cisco Unified C					Navigation Cisco	Unified CM / admin	Administrati About	on 💌 🤇
System 👻	Call Routing 👻 Media	Resources 👻 Voice	e Mail 👻 Device 👻	Application \bullet	User Management 👻	Bulk Administration $ \star $	Help 🔻		
Media Tel	rmination Point Cor	nfiguration				Related Link	s: Back To	Find/List	•
🔚 Save	🗙 Delete 📋 Cop	y 🎦 Reset 🕂	Add New						
— Status — (i) Statu	ıs: Ready								
	ermination Point Info								
Registrati IP Addres		-	isco Unified Commu	nications Mana	ager 10.40.97.2				
	» mination Point Type*	10.40.97.1 Cisco IOS Enhance	od Software Media "	Cormination Do	aint				
	mination Point Name*		eu Joitware Meula		Jine				
		MIP111222333							
Descriptic	on	MTP-Ent1-HQ							
Device Po	ool*	DevicePool_HQ_D	DSPfarm	•					
- Save	Delete Copy R	Reset Add New							

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Media Resources: Music on Hold Server Parameters

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

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

01000	d CM Administration	Navigation Cisco Unified CM Administration 💌 🖸 admin About Logou
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 💽 G
🔚 Save		
Status Ready		
—Device Information ———		
Registration IP Address	Registered with Cisco Unified Communications Manager 10.40.97.2 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 Inform	ation —	
🗆 Enable Multicast Audio Sourc	es on this MOH Server	
Base Multicast IP Address*	0.0.0	
Base Multicast Port Number* 0	(Even numbers only)	
Increment Multicast on*	Port Number O IP Address	
Calendary Multiplet Audio Com		
 Selected Multicast Audio Sour There are no Music On Hold Aud Audio Sources. 	ces io Sources selected for Multicasting. Click Configure Audio Sources in the	top right corner of the page to select Multicast



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(i) *- indicates required item.

Media Resources: Transcoder Parameters

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

Figure 62 Media Resources Transcoder Cisco Unified CM Administration Window

cisco	fied CM Administratio	on 💌 🤇 Logou								
System 👻 Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help 👻										
Find and List Transcoders										
Add New ESelect All Clear All Clear All Clear All Clear All Clear All Clear All Reset Selected Status 2 records found Transcoder (1 - 2 of 2) Rows per Page 50 T										
Find Transco	oder where Name	begins with 💌	FindClear	Filter 🕂 😑						
	Name 🕇	Description	Device Pool	Status	IP Address	Сору				
	XCD001AA29DF631	XCODE-Ent1-Br1	DevicePool Br1 DSPfarm	Registered with 10.40.97.2	10.40.103.1	ß				
	XCODE111222333	XCODE-Ent1-HQ	DevicePool HQ DSPfarm	Registered with 10.40.97.2	10.40.97.1	ß				
Add New	Select All Clear All	Delete Selected	Reset Selected							

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Figure 63 Media Resources Transcoder XC	CODE Enterprise 1 Branch 1 Cisco Unified CM Administration Window	/
Cisco Unified CM Administrations		J.ogou
System ▼ Call Routing ▼ Media Resources ▼ Voice Mail ▼ De	vice 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help 👻	
Transcoder Configuration	Related Links: Back To Find/List	•
🔚 Save 🗙 Delete 🗈 Copy 省 Reset 🕂 Add New		
Transcoder Information Transcoder: XCD001AA29DF631 (XCODE-Ent1-Br1) Registration Registered with Cisco Unified Communications N IP Address 10.40.103.1 IOS Transcoder Info Transcoder Type* Cisco IOS Enhanced Media Term		
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	
- Save Delete Copy Reset Add New		

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Enterprise 1 HQ Cisco Unified CM Ex	xample Configurat	ion

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

abola Cisco Unified CM Administration	Navigation _Cisco Unified CM Administration 💌 🕻
CISCO For Cisco Unified Communications Solutions	admin About Logou
System ▼ Call Routing ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Application	✓ User Management ✓ Bulk Administration ✓ Help ✓
Transcoder Configuration	Related Links: Back To Find/List 💽 G
🔚 Save 🗶 Delete 🗋 Copy 嗋 Reset 🕂 Add New	
Transcoder Information Transcoder: XCODE111222333 (XCODE-Ent1-HQ) Registration Registered with Cisco Unified Communications Manager 10.40.97 IP Address 10.40.97.1	2
— 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	

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Media Resources: Media Resource Group Parameters

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

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

CISCO	Cisco Unified CM Ad				Navigation Cisco	Unified CM Adminis	stration 💌 🖸 out Logou			
System 👻 Ca	all Routing 👻 Media Resources 👻	Voice Mail 👻 Device 👻	Application \star	User Management 👻	Bulk Administration $ \star $	Help 👻				
Find and Lis	t Media Resource Groups									
Add New	Add New 🔠 Select All 🔛 Clear All 💥 Delete Selected									
	Status i 2 records found Media Resource Group (1 - 2 of 2) Rows per Page 50									
Find Media R	esource Group where Name	💌 begins with 💌		Find Clear I	Filter 🕂 😑					
	Name	•	Des	cription	Multicast		Сору			
	Br1 HW MRG	E	ent 1 Br1		false	ß				
	HQ HW MRG	E	int 1 HQ		false	ß				
Add New	Select All Clear All I	Delete Selected								

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

cisco		d CM Administration 💌 🕻 dmin About Logou
System 👻	Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help	•
Media Res	ource Group Configuration Related Links: Ba	ck To Find/List 💽 🤇
ave 🔚	🗙 Delete 🗋 Copy 🌯 Reset 🕂 Add New	
— Status — i Statu:	: Ready	
	source Group Status ource Group: Br1_HW_MRG (used by 11 devices)	
—Media Re Name*	source Group Information	
	Br1_HW_MRG	
Descriptio	Ent 1 Br1	
- Dauisas f	or this Group	
	Iedia Resources** ANN_2 CFB_2 CON111222333 MTP_111222333 MTP_2 ▼	
Selected N	edia Resources* CON001AA29DF631 (CFB) MOH-Ent1 (MOH) MTP001AA29DF631 (MTP) XCD001AA29DF631 (XCODE)	
🗆 Use Mu	lticast for MOH Audio (If at least one multicast MOH resource is available)	
- Save	Delete Copy Reset Add New	
(i) *- inc	icates required item.	746
(i) **Ind	ludes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and T	ranscoders (XCODE) 24 24 24 24 24 24 24 24 24 24 24 24 24 2

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cisco	Cisco Unif			tion			Navigation Cisco	Unified CM Administration 💌 admin About Loc
System 👻	Call Routing 👻 Me	dia Resources 🔻	Voice Mail 👻	Device 🔻	Application 👻	User Management 👻	Bulk Administration 👻	
Media Re	source Group Col	nfiguration					Related Lini	<s: back="" find="" list<="" td="" to=""></s:>
ave 🔚	X Delete 🗋 C	Copy 🎦 Reset	🕂 Add New					
— Status — i Statu	us: Ready							
	esource Group Stat source Group: HQ_H		y 19 devices)					
— Media Re Name*	esource Group Info HQ_HW_MRG	ormation ———						
Descriptio	on Ent 1 HQ							
	for this Group							
Available	Media Resources**	ANN_2 CFB_2 CON001AA29D MTP001AA29DF MTP_2				• •		
Selected I	Media Resources*	CON11122233 MOH-Ent1 (MO MTP111222333 XCODE111222	3 (CFB) H) 8 (MTP)			*		
🗆 Use Mi	ulticast for MOH Aud	dio (If at least on	e multicast M	OH resour	ce is available)			
- Save	Delete Copy	Reset Add	New					
(i) *- in	dicates required ite	m.						
(i) **In	cludes Annunciator:	s (ANN), Confere	nce Bridges ((CFB), Media	a Termination A	Points (MTP), Music O	n Hold Servers (MOH)	and Transcoders (XCODE)

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

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Media Resources: Media Resource Group List Parameters

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

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

cisco		nified CM Ad		n		Navigation Cisc		ministration 💌 🤇
System 👻		Media Resources 👻		ce 👻 Application 🔻	· User Management 👻	Bulk Administration 👻	admin Help 🔻	About Logou
Find and	List Media Re	source Group Lists						
Add N	lew Select	All 🔲 Clear All 🙀	Delete Selected					
-Status-	ords found							
Media R	lesource Group	List (1 - 2 of 2)					Rows p	er Page 50 💌
Find Medi	ia Resource Gro	up List where Name 🛛	pegins with 💌		Find Clear Filter	÷ + -		
	Γ			Name 🕇			Сор	ру
		Br1 HW MRGL				ß		
		HO HW MRGL				5		
Add Ne	ew Select Al	l Clear All De	elete Selected					

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Figure 6	9 Media Resources-Media Resource Group List Branch 1 HW MRGL	Cisco Unified CM Administration Window
cisco	Cisco Unified CM Administration	Navigation Cisco Unified CM Administration 💌 C
	For Cisco Unified Communications Solutions	admin About Logou
System 👻	Call Routing ▼ Media Resources ▼ Voice Mail ▼ Device ▼ Application ▼ User Management ▼	Bulk Administration 👻 Help 👻
Media Re:	source Group List Configuration	Related Links: Back To Find/List 👤 🤆
🔚 Save	🗙 Delete 📔 Copy 🎦 Reset 🕂 Add New	
—Status —		
(1) Statu	s: Ready	
Media Res	source Group List Status source Group List: Br1 HW MRGL (used by 11 devices)	
Name* Bi	1 HW MRGL	
—Media Re	source Groups for this List	
	Media Resource Groups HQ_HW_MRG	
Selected I	Media Resource Groups Br1_HW_MRG	
Save	Delete Copy Reset Add New	4 0
(i) *- inc	dicates required item.	273749

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

cisco	Cisco Unified CM Administration	Navigation Cisco Unified CM Administration 💌 🕻
cisco	For Cisco Unified Communications Solutions	admin About Logou
System \star	Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management	t 👻 Bulk Administration 👻 Help 👻
Media Re	source Group List Configuration	Related Links: Back To Find/List 👤 🤆
🔚 Save	🗙 Delete 📋 Copy 🎦 Reset 🕂 Add New	
-Status - i Statu	ıs: Ready	
	esource Group List Status source Group List: HQ HW MRGL (used by 19 devices)	
	esource Group List Information Q HW MRGL	
	esource Groups for this List Media Resource Groups Br1_HW_MRG	
Selected	Media Resource Groups HQ_HW_MRG	×
- Save	Delete Copy Reset Add New	20
(i) *- ir	dicates required item.	273750

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Configuring the Cisco Unified CM Voice Mail Parameters

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

- Voice Mail: Cisco Voice Mail Port Parameters, page 95
- Voice Mail: Message Waiting Parameters, page 97
- Voice Mail: Voice Mail Pilot Parameters, page 100
- Voice Mail: Voice Mail Profile Parameters, page 101

Voice Mail: Cisco Voice Mail Port Parameters

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

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

Cisco Unified CM Administration Navigation Cisco Unified CM Administration Cisco Unified CM Ad										
System 👻 Call Routing 👻 Media Resources 👻 Voice Mail 👻 Device 👻 Application 👻 User Management 👻 Bulk Administration 👻 Help 👻										
Find and List Voice Mail Ports										
Stat			Clear All	P Reset S	elected					
Voice Mail Port (1 - 5 of 5) Rows per Page 50										
Find Voice Mail Port where Device Name 💌 begins with 💌 Find Find Clear Filter 🔂 📼 Select item or enter search text 💌										
	Device Name ≜	Description	Device Pool	Device Security Mode	Calling Search Space	Ext.	Partition	Status	IP Address	Сору
	<u>CiscoUM1-</u> <u>VI1</u>	Voicemail for Enterprise1	DevicePool HQ IP Phones	Non Secure Voice Mail Port	<u>CSS-</u> HQ Phones IP	<u>1090</u>	<u>Partition-</u> HQ Phones IP	Registered with 10.40.97.2	10 40.97.253	ß
	<u>CiscoUM1-</u> <u>VI2</u>	Voicemail for Enterprise1	DevicePool HQ IP Phones	Non Secure Voice Mail Port	<u>CSS-</u> HQ Phones IP	<u>1091</u>	<u>Partition-</u> HQ Phones IP	Registered with 10.40.97.2	10.40.97.253	ß
	<u>CiscoUM1-</u> <u>VI3</u>	Voicemail for Enterprise1	DevicePool HQ IP Phones	Non Secure Voice Mail Port	<u>CSS-</u> HQ Phones IP	<u>1092</u>	<u>Partition-</u> HQ Phones IP	Registered with 10.40.97.2	10.40.97.253	ß
	<u>CiscoUM1-</u> VI4	Voicemail for Enterprise1	DevicePool HQ IP Phones	Non Secure Voice Mail Port	<u>CSS-</u> HQ Phones IP	<u>1093</u>	<u>Partition-</u> HQ Phones IP	Registered with 10.40.97.2	10.40.97.253	ß
	<u>CiscoUM1-</u> <u>VI5</u>	Voicemail for Enterprise1	DevicePool HQ IP Phones	Non Secure Voice Mail Port	<u>CSS-</u> HQ Phones IP	<u>1094</u>	<u>Partition-</u> HQ Phones IP	Registered with 10.40.97.2	10.40.97.253	ß
Ac	d New S	elect All Cle	ar All 📗 Delete Selected	Reset	Selected					

	Enterprise 1	I HQ Cisco	Unified CN	/I Example	Configuration
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Figure 72 Voice Mail-Voice Mail Port CiscoUM1 VI1 Cisco Unified CM Administration Window

cisco	Cisco Unified			ation				Na	vigation Ci		ied CM . admin		tion 🔽 🕻
System 👻	Call Routing 👻 Medi	ia Resource:	s 👻 Voice Mail 👻	Device 🔻	Application 👻	User M	anagement 👻	Bulk A	dministration		p 🕶		
Voice Mai	il Port Configuratic	on							Related L	inks: 🖪	Back To) Find/List	▼ 6
📄 Save	🗙 Delete ြ Co	ıpy 🎦 Re	eset 🛟 Add New	/									
— Status — (i) Statu	is: Ready												
— Device I Registrati IP Addres Port Name	s	Registered 10.40.97.2 CiscoUM1-		ed Commur	nications Mana	ger 10.4	0.97.2						
Descriptio	n	Voicemail	for Enterprise1										
Device Po	ol*	DevicePor	ol_HQ_IP_Phones	5	•								
Common I	Device Configuration	< None >			•								
Calling Se	arch Space	CSS-HQ_F	Phones_IP		•								
AAR Callin	ig Search Space	< None >			•								
Location*		Hub_HQ			•								
Device Se	curity Mode*	Non Secu	re Voice Mail Port		▼								
	y Number Informatio												
Directory	Number*		1090										
Partition			Partition-HQ_Pho			_							
-	earch Space		CSS-HQ_Phones	_IP		•							
AAR Group			< None >			•							
	Caller ID Display		VoiceMail										
Internal C	Caller ID Display (ASC	:II format) 🗗	VoiceMail										
External N	Number Mask	ſ	415555XXXX										
- Save	Delete Copy		Add New										273782

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Voice Mail: Message Waiting Parameters

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

Figure 73 Voice Mail Message Waiting Cisco Unified CM Administration Window

cisco	Cisco Unified CM Ad For Cisco Unified Communication			Navigation Cis	co Unified CM Administrat admin About	ion 💌 🤇
System 👻	Call Routing 👻 Media Resources 👻	Voice Mail 👻 Device 👻	Application 👻 User Mana	gement 👻 Bulk Administration	✓ Help ✓	
Find and	List Message Waiting Numbers					
Add N	lew 🔛 Select All 🔛 Clear All 💥	Delete Selected				
	e Waiting Numbers (1 - 2 of 2)				Rows per Page	50 💌
-	sage Waiting where Directory Numb	per 💌 begins with	•	and where Message Waiting Indicator is Both 💌		÷ =
	Directory Number *	Description	Partition	Call	ling Search Space	Сору
	1080	MWI-On	Partition-HQ Phones IP	CSS-HQ_P	hones IP	6
	<u>1081</u>	MWI-Off	Partition-HQ Phones IP	<u>CSS-HQ_P</u>	hones IP	ß
Add Ne	ew Select All Clear All D	elete Selected				

	Enterprise	1 HQ	Cisco	Unified	СМ	Example	Configu	ration

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

սիսիս	Cisco Uni	fied CM Adm	ninistra	tion			Navigation Cisco	Unified CM A	dministrati	on 🔽 🤇
cisco	For Cisco Unifie	ed Communications	Solutions					admin	About	Logou
System 👻	Call Routing 👻 M	ledia Resources 👻 🗸 V	oice Mail 👻	Device 👻	Application \bullet	User Management 👻	Bulk Administration $ ullet $	Help 👻		
Message	Waiting Configu	ıration					Related I	_inks: Back	To Find/Li	st 🔻 🤆
🔚 Save	🗙 Delete [🗋	Copy 🕂 Add New								
— Status — (i) Statu	ıs: Ready									
—Message	Waiting Informa	tion								
Message	Waiting Number*	1080								
Partition		Partition-HQ_Phon	ies_IP		•					
Descriptio	on	MWI-On								
Message	Waiting Indicator*	'⊙on Coff								
Calling Se	arch Space	CSS-HQ_Phones_I	IP.		•					
- Save	Delete Copy	Add New								

1

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

ahaha	Cisco Uni	fied CM Adr	ninistra	ation			Navigation Cisco	Unified CM 4	\dministrati	on 💌 🤇
cisco	For Cisco Unific	ed Communications	Solutions					admin	About	Logou
System 👻	Call Routing 👻 🛛 🕅	ledia Resources 👻 🕚	voice Mail 👻	Device 👻	Application \bullet	User Management 👻	Bulk Administration 👻	Help 🔻		
Message	Waiting Configu	ıration					Related	Links: Back	To Find/Li	ist 🔻 🤇
🔚 Save	🗙 Delete 🗋	Copy 🕂 Add New								
— Status — (i) Statu	s: Ready									
—Message	Waiting Informa	tion								
Message	Waiting Number*	1081								
Partition		Partition-HQ_Pho	nes_IP		•					
Descriptio	n	MWI-Off								
Message ^v	Waiting Indicator'	[*] ⊂ On ☉ Off								
Calling Se	arch Space	CSS-HQ_Phones_	IP		•					
- Save	Delete Copy	Add New								

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Voice Mail: Voice Mail Pilot Parameters

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

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

cisco		nified CM Ad		ation			Navigation Cisco	Unified CM admin	Administrat	ion 🔽 🤇
System 👻	Call Routing 👻	Media Resources 👻	Voice Mail 👻	Device 👻	Application 👻	User Management 👻	Bulk Administration 👻	Help 👻		
Voice Mai	il Pilot Config	uration					Related Link	s: Back To) Find/List	- 6
🔚 Save	X Delete	Add New								
<u> </u>	s: Ready									
	i l Pilot Inform Pilot Number									
		CSS-HQ_Phones_IP			•					
Descriptio	n F	Voicemail Pilot								
🗹 Make t	his the default	Voice Mail Pilot for th	ne system							
– <u>Save</u>	Delete Ac	dd New								

273786

Voice Mail: Voice Mail Profile Parameters

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

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

ahaha	Cisco U	nified CM Ad	ministra	ation				Navigation Cise	o Unified CM	1 Administrat	ion 🔽 🤇
cisco	For Cisco Un	ified Communication	ns Solutions						admin	About	Logou
System 👻	Call Routing 👻	Media Resources 👻	Voice Mail 👻	Device 👻	Application •	 User Management 	•	Bulk Administration	• Help •		
Voice Mai	il Profile Conf	iguration						Related Lir	nks: Back T	o Find/List	•
📄 Save	X Delete	🗋 Copy 🎦 Reset	🕂 Add New								
— Status — (i) Statu	ıs: Ready										
Voice Mail Voice Mail Descriptio Voice Mail Voice Mail	Profile Name* n Pilot** Box Mask	mation VM-Profile-Ent1-HQ (VM-Profile-Ent1-HQ (Default voice messa 1099/CSS-HQ_Phon Voice Mail Profile for	ging profile nes_IP	levices)	V						
\smile	Delete Cop dicates required	d item.			ar and ithic a	avecage dias Calling			e Veice Mail I		
Callir	ng Search Spac	Pilot is comprised of the pilot is comprised of the pilot is comprised of the pilot the pilot is a comprised of th	te voice Mall	Phot Nullio	er anu icys c	onesponding calling	, 580	загот зрасе мате (-	< voice mail i	Pilot Number	2/<

Configuring the Cisco Unified CM Device Parameters

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

- Device: Gateway Parameters, page 102
- Device: Phone Parameters, page 109
- Device: Trunk Parameters, page 114

Device: Gateway Parameters

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

Figure 78	Device Gateway	Cisco II	Inified CM	Administration	Window
rigule 70	Device Galeway	CISCO 0	inneu civi	Aunimistiation	vviilaovv

ciso				ation				Navigati	on Cis		CM Administra	
Quatam				Douise - A	nalisation -	Lloor Monor	vomont -	Dulle i desi	niatratia	adm		Logou
System	▼ Call Routing ▼ Mee	lia Resources 👻 🕚	voice Mail 🔻	Device 🔻 A	pplication 🔻	User Mana <u>o</u>	gement 👻	Bulk Admi	nistratio	on 🔻 Help	•	
Find a	nd List Gateway											
🕂 Ad	ld New 🔛 Select All	Clear All 🙀	Delete Selec	ted 🎦 Resi	et Selected							
— Statu	s											
i 2	records found											
Gate	ways (1 - 2 of 2)									R	ows per Page	e 50 💌
Find G	ateways where Name	•	begins wit	:h 💌		Hide 💌	endpoint	ts Find	Clear	Filter	₽ —	
				Select	item or ente	r search te	xt 💌					
	Device Name 🌥	Description	Device Pool	Calling Search Space	Extension	Partition	Route Group	Priority	Port	Device Type	Status	IP Addres
	Ent1 Br1.Ent1.com	<u>n</u> Ent1_Br1								Cisco 3845	<u>See</u> Endpoints	
	SKIGW0C863972F	5 Ent1-HQ- VG224								VG224	<u>See</u> Endpoints	

alada cisco	Cisco Unified			tion			Navigation Cisco	Unified CM A	dministration 💌 🕻 About 📔 Logou
System 👻	Call Routing 👻 Medi	ia Resources 👻	Voice Mail 👻	Device 👻	Application 👻	User Management 👻	Bulk Administration 👻	Help 👻	
Gateway	Configuration						Related L	inks: Back	To Find/List 🖵 🤇
🔚 Save	🗙 Delete	set 🕂 Add Ne	9W						
— Status — (i) Statu	is: Ready								
—Gateway	/ Details								
Product			Cisco 3845						
Gateway			Ent1_Br1.Er	it1.com					
Protocol			MGCP						
Domain N	ame*		Ent1_Br1.Er	nt1.com					
Descriptio	n		Ent1_Br1						
Cisco Unif	fied Communications	Manager Group	* Default			•			
Module in Module in Module in Module in	ed Slots, VICs and E Slot 0 < None > Slot 2 < None > Slot 3 < None > Slot 3 < None > Slot 4 NM-HDV2-2P0 Subunit 0 VIC Subunit 1 < N			gin Port 0 gin Port 0	., _,	4/0/ 1 👼			
-Product	Specific Configuration	on Layout ——							
ol 1 1					?				
	ON Switch Type	4ESS			_				
	ck Timing*	Graceful							
Switchba	ck uptime-delay (min)	10							
Switchba	ck schedule (hh:mm)	12:00							
	TMF Relay*	Current GW C	onfig		•				
- Save	Delete Reset	Add New -							273719

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

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

diala cisco		nified CM Ad		ation			Navigation Cisco	Unified CM admin		tion 💌 🤇
System 👻	Call Routing 🔻	Media Resources 👻	Voice Mail 👻	Device 👻	Application \bullet	User Management 👻	Bulk Administration 👻	Help 🔻		
Gateway	Configuration					F	Related Links: Bacl	< to MGCP	Configurat	ion 💌 🤆
🔚 Save	X Delete	👆 Reset 👍 Add N	BW							
— Status —										

D	Status:	Readv

Directory Number Information	Device Information					
•778 Line [1] - 1110 in Partition-	Product	Cisco MGCP FXS Port				
Br1 Phones Analog	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 Ent1_Br1_FXS DevicePool_Br1_Analog_Phones				
	Description					
	Device Pool*					
	Common Device Configuration					
	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 availab					
	MLPP Preemption Not availab	le 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						

i *- indicates required item.

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

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

CISCO For Cisco Unified Communications Solutions System • Call Routing • Media Resources • Voice Mail • Device •				admin About Log Application • User Management • Bulk Administration • Help •				
Directory Number Configuration			once man · Denne ·					
				Related Links:	Configure Device (AALN/S4/SU0/0@Ent1_Br	1.Ent1.com) 💌		
🛛 Save 🗙 Delete	Heset 1	Add New						
Status Status: Ready								
-								
Directory Number In Directory Number*	nformation 1110			1				
Route Partition	Partition-Br1_	Phones_Ana	log	1				
	1110			1				
	Ent1_Br1_111			3				
ASCII Alerting Name								
Associated Devices	AALN/S4/SUO	0@Ent1_Br1	.Ent1.com	Edit Device				
				Edit Line Appe	arance			
	~	^						
Dissociate Devices				1				
Directory Number S	ettings							
/oice Mail Profile		None >			<none> to use system default)</none>			
Calling Search Space Presence Group*		SS-Br1_Phor						
User Hold MOH Audio		tandard Pre: -SampleAudi						
Network Hold MOH Au								
				_				
AAR Settings	Voice M	ail	AAR D	estination Mask	AAR Group			
AAR 🗆 c	or .				< None >	×		
Retain this destin forwarding history	ation in the c	lla						
Call Forward and Ca		tings Voice Mail	D	estination	Calling Search Space			
Calling Search Space					Use System Default			
Forward All		🗆 or			< None >	•		
Secondary Calling Se	earch Space fo	r Forward A	1		< None >			
Forward Busy Intern	al	🗆 or			< None >	×		
Forward Busy Extern	al	🗆 or			< None >	×		
Forward No Answer	Internal	[] or			< None >			
Forward No Answer	External	- or			< None >	×		
Forward No Coverag	e Internal	□ or	[< None >			
Forward No Coverag	e External	□ or			< None >	*		
Forward on CTI Failu	ire	□ or			< None >	*		
Forward Unregistere	d Internal	□ or			< None >			
Forward Unregistere	d External	🗆 or			< None >			
No Answer Ring Durat	tion (seconds)							
Call Pickup Group		< None >						
MLPP Alternate Part	ty Settings—							
Target (Destination) MLPP Calling Search S								
MLPP Calling Search S MLPP No Answer Ring		< No	ne >	×				
Line Settings for All Hold Reversion Ring D					etting the Hold Reversion Ring Duration to zer			
(seconds)		feature	§	5	etting the Hold Reversion King Duration to zer	o will disable th		
Hold Reversion Notific (seconds)	ation Interval		the feature	S	etting the Hold Reversion Notification Interval	to zero will		
Line 1 on Device AA Display (Internal				Display text for a line	e appearance is intended for displaying text s	uch as a name		
Caller ID)	nstead of a di of the caller.	rectory num	per for internal calls. If	you specify a number,	the person receiving a call may not see the p	roper identity		
ASCII Display								
(Internal Caller ' ID)								
External Phone Number Mask	415371XXXX							
	alting Com	r on Davi-	AALN/S4/SUD/D@E	the Bel Cett				
Note:The range to sel Maximum Number of C	lect the Max N		ls is: 1-2					
Maximum Number or G Busy Trigger*	alls -		2			100		
ousy migger			1		(Less than or equal to Max. C	alls)		
	rmation Disp	ay on Devic	e AALN/S4/SU0/0@	Ent1_Br1.Ent1.com				
Caller Name								
Caller Number Redirected Number	,							
Redirected Number Dialed Number								
Users Associated wi	th Line	isers						
As	sociate End U							
Save Delete	Reset Add							

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

cisco	Cisco Unified CM Adm For Cisco Unified Communications				Naviç	ation Cisco (Jnified CM / admin	Administratio	
System 👻	Call Routing 👻 Media Resources 👻 🕔	Voice Mail 👻 Device 👻 Ap	oplication 👻	User Managemer	nt 👻 🛛 Bulk Ad	ministration 👻	Help 👻		
Gateway	Configuration					Related Li	inks: Back	: To Find/Lis	t 💌 🤆
🔚 Save	🗙 Delete Paset 🕂 Add New	1							
— Status — (i) Statu	ıs: Ready								
—Gateway	y Details								
Product		VG224							
Gateway Protocol		SKIGWOC863972F5 SCCP							
	ess (Last 10 Characters)*	0C863972F5							
Descriptio		Ent1-HQ-VG224							
Cisco Unif	fied Communications Manager Group*	Default		•					
	red Slots, VICs and Endpoints								
	Subunit 0 24FXS-SCCP	2/0/ 0 🎬	2/0/1 🎬	2/0/ 2 🗳	2/0/ з 🗳	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		2/0/14 🗳 2/0/15 🗳	2/0/16 💕	_					
2/0/18		2/0/20 🗳 2/0/21 🗳	2/0/22 🗳	2/0/23 🗳					
- Save	Delete Reset Add New								
(i) *- in	dicates required item.								273722

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

abole Cisco Unified CM Administr	ation		Navigation Cisco U	Inified CM Admir	istration
CISCO For Cisco Unified Communications Solutions					bout l
stem ▼ Call Routing ▼ Media Resources ▼ Voice Mail ▼	Device - Application -	🔹 User Management 👻	Bulk Administration 👻	Help 🔻	
one Configuration		Related	Links: Back to Gat	eway	
] Save 🗙 Delete 📋 Copy 💁 Reset 🕂 Add New	,				
Status: Ready					
-					
Association Information Modify Button Items	Phone Type Product Type: An	alog Phone			
Line [1] - 1050 in Partition-HQ Phones Analog	Device Protocol: SC	-			
	Device Information Registration	Registered with Cisco	Unified Communicati	one Managor 1(1 40 07 2
ame Line [2] - Add a new DN	IP Address	10.40.97.254	onned communicad	ons Manager 10	7.40.57.2
	MAC Address*	0C863972F5400			
	Description	415555XXXX			
	Device Pool*	DevicePool_HQ_Ana	log_Phones	View D	<u>etails</u>
	Common Device Configuration	< None >		View D	<u>atails</u>
	Phone Button Template*	Standard Analog		•	
	Common Phone Profile*	Standard Common F	hone Profile	•	
		CSS-HQ_Phones_An	aloq	•	
	Media Resource	HQ HW MRGL			
	Group List				
	Location*	Hub_HQ		•	
	User Locale	< None >		•	
	Network Locale	< None >		•	
	Device Mobility Mode*	Default		View C	<u>urrent</u>
	Owner User ID	Device Mobility Settin	<u>qs</u>		
	_	< None >		•	
	Is Active				
		on Indicators (interna	calls only)		
	Allow Control of D				
	Logged Into Hunt	Group			
	Remote Device				
	Protocol Specific In				_
	Presence Group*		Presence group		•
	Device Security Profile	1	hone - Standard SCCI	P Non-Secure P	
		earch Space < None :	,		•
	Unattended Port				
	MLPP Information -				
		: None >		•	
)efault		•	
	MLPP Preemption*)efault		•	
Rave Delete Copy Boset Add New					
Save Delete Copy Reset Add New					
) *- 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.

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

	Configuration		Related Li	inks: Configure Device (ANOC863	972F5400) 💌
🚽 Save 🗙 Deleti	🕈 🍄 Reset 🛟 Add New				
Status) Status: Ready					
Directory Number	Information				
Directory Number*	1050				
Route Partition Description	Partition-HQ_Phones_Anal	00			
Alerting Name	1050				
ASCII Alerting Name					
R Allow Control of D					
Associated Devices	AN0C863972F5400		the Devilee		
			it Device Edit Line Appearance		
	~~				
Dissociate Devices	•••				
Directory Number	Settings				
Voice Mail Profile	< None >		Choose <none> to</none>	o use system default)	
Calling Search Space Presence Group*	and the second				
User Hold MOH Audio	Standard Pres		*		
	udio Source 1-SampleAudio				
AAR Settings					
AAR 🗖	Voice Mail	AAR Destinat		< None >	
Retain this dest					
forwarding history					
Call Forward and C	all Pickup Settings Voice Mail	Destinat		Calling Search Space	
Calling Search Space		C C P C P C P C P C P C P C P C P C P C		Use System Default	
Forward All	🗖 or			< None >	×
Secondary Calling 9	earch Space for Forward Al		1	< None >	•
Forward Busy Inter	nal 🗌 or		[< None >	۲
Forward Busy Exter	nal 🗆 or			< None >	
Forward No Answe	1. 01			< None >	
Forward No Answe	1.01			< None >	*
Forward No Covera	1.01			< None >	
Forward No Covera				< None >	
Forward on CTI Fail Forward Unregister		I		< None >	
Forward Unregister	1. 01	I		< None >	
No Answer Ring Duri				< NODE >	-
Call Pickup Group	< None >				
MLPP Alternate Pa	rty Settings				
Target (Destination)					
MLPP Calling Search	Space < Nor g Duration (seconds)	99 >			
Line Settings for A Hold Reversion Ring			Cattion the b	fold Reversion Ring Duration to zero	n will disable
(seconds)	the feat.	re			
Hold Reversion Notif (seconds)		he feature	Setting the F	iold Reversion Notification Interval	to zero will
Line 1 on Device A	NDC863972E5400				
Display (Internal Caller ID)				nce is intended for displaying text	
	identity of the caller.	y number for internal calls. If	you specify a number, the	e person receiving a call may not se	e the proper
ASCII Display (Internal Caller					
ID) External Phone					
Number Mask	415555XXXX				
Monitoring Calling Search Space	< None >				
Multiple Call/Call V	Vaiting Settings on Device	ANDC863972F5400			
Note: The range to s Maximum Number of	elect the Max Number of call Calls*	s is: 1-2		_	
Busy Trigger*		1		(Less than or equal to Max. C	alls)
Forwards 4 Coll 2-4	emation Director on D				
Forwarded Call Inf Caller Name	ormation Display on Devic	e ANOC863972F5400			
Caller Number					
Redirected Numb	er				
P Dialed Number					
Users Associated v					
4	ssociate End Users				
	and some t				
Save Delete	Reset Add New				
-					
Save Delete	uired item.				
i •- indicates req		ettings require restart.			

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Device: Phone Parameters

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

Figure 85 Device Phone 4155551000 Cisco Unified CM Administration Window

e Configuration Lee 🗶 Conto 🗋 Copy 🌚 Rosat 🚭 Addi No	iter	Related Links:	Neck To Find/Lat			
96 Status: Ready						
eciation Information Modify Button Items	Phone Type Product Type: Cince Device Protocal: SCO	7971				
Line F11 - 1000 in Partition HD Phones JP	- Device Information -					
The Color Add a new DN Reg Add a new SD	Registration IP Address MAC Address	Registered with 0 10.40.97.75 00387375C3FA	isco Unified Communications I	tanager 10.40.97.2		
Ag Add a new 50 Ag Add a new 50	Description	4155551000		-		
Per Add a new SD	Device Pool* Common Device	DevicePool_HQ_ < None >	P_Phones	Vew Cetals Vew Cetals		
Reg Add a new 50	Common Device Configuration Phone Button Template* Softkey Template	Standard 7971 1				
	Softkey Template Common Phone Profile*	Standard User	n Ohnea Brotha			
 Add a new SURL 	Calling Search Space AAR Calling Search	Standard User Standard Comm CSS-HQ_Phones in None 5	2			
Add a new BLF SD	Space			•		
Caliback Cali Park	User Hold MOH Audio	1-SampleAudioS	ource	-		
Cell Pickup	Auto Source	11-5-ampleAu/9075		*		
Conference List. Conference Do Not. Disturb	AAR Group	Hub_HQ <none></none>		*		
	User Locale Network Locale	< None > < None > Default		*		
Forward All Group Call Pickup Hold	Built In Bridge* Privacy*	Default Default		•		
Hurt Group Logicut	Device Mobility Mode*			Vex Current		
Malicious Call Identification	Owner User ID Phone Personalization*	Cerca Mobility Se < None > Code at				
Meet Me Conference Mobility	Phone Personalization* Phone Load Name	Default				
New Call Other Pickup	E in Artist					
Quality Reporting Tool Redial	Join Across Lines IF Ratty Video Call as A	Default				
Femore Last Participant Transfer	C Ignore Presentation R Allow Control of Devi	Indicators (intern	il calls only)			
Video Mode Privaty	V Logged Into Hunt Gr	oup				
None	E Remote Device					
	Protocol Specific Infor Packet Capture Mode* Packet Capture Duratio					
	Packet Capture Duration Presence Group* Device Security Profile* SUBSCRIBE Calling Sear	0 Standar	d Presence group	-		
	Device Security Profile* SUBSCRIBE Calling Sear	Cisco 7	11 - Standard SCCP Non-San	ine Profil		
	C Unattended Port					
	IT RFC2033 Disabled					
	Certification Authority Certificate Operation*	Presy Function (CAPE) Information	-		
	Authentication Mode* Authentication String	By Null Strie	9	2		
	Cenerate Dring	-				
	Operation Completes Ib:	r Ernin Itt	0 000 0000 MM 660440	2		
	Certificate Operation St. Note: Security Profile Co.	atus: None ontains Addition G	PT Settings.			
	Expansion Module Info Module 1					
	Module 1 Load Name					
	Module 2 Coad Name	None >	2			
			rave blank to use default)			
	External Data Lacations Information (Leave blank to use default) Information Directory					
	Messages Services					
	Authentication Server	-				
	Proxy Server					
	Ide Timer (seconds)			1		
	Extension Information	sbilty				
	Extrastes Information C Enable Extension Mo Log Out Profile - Use O Log out Time < None : Log out Time < None :	Current Device Set	ingi - 💌			
	MLPP Information MLPP Domain (c.M. MLPP Indication* (Def	010.3	2			
	MLPP Indication* [Def MLPP Preemption* [Def	nd nd	2			
	De Net Disturb					
	C Do Not Disturb	Ringer Off		9		
	DND lncoming Call Alert			•		
	Secure Shell Informat Secure Shell User					
	Secure Shell Password			2		
		guration Layout -		?		
	C Disable Speakerphor C Disable Speakerphor Forwarding Delay*	ne and Headoet				
	PC Port *		Disabled Enabled			
	Settings Access* Gratuitous ARP*		Erabled Disabled	2		
	PC Voice VLAN Access* Video Capabilities*		Enabled			
	Auto Line Select*		Disabled Disabled	2		
	Web Access* Days Display Not Active		Enabled Sources	*		
	Display On Time		Monday Tuesday 07:30	ž.		
	Display On Duration		10:30			
	Display Idle Timeout Span to PC Port*		01:00 Disabled	2		
	Logging Display*		PC Controlled	2		
	Recording Tone*		Disabled			
	Recording Tone Local Vo Recording Tone Remote	Volume*	100 50			
	Recording Tone Duration					
	Display On When Incom RTCP*		Disabled Disabled	-		
	"more" Soft Key Timer Auto Call Select*		5 Enabled			
	Log Server Advantise 6,722 Coder*					
	Wideband Headset ULC	Control*	Disabled Enabled			
	Wideband Handset UE C Wideband Headset*	Control*	Enabled Enabled	2		
	Wideband Handset*		Use Phone Default Disabled			
	Peer Firmware Sharing* Cisco Discovery Protoco Port*	(CDP) Switch	Enabled	-		
	Cisco Discovery Profaco	1 (COP): PC P647*	Enabled Enabled			
	Unit Layer Discovery Pro Endpoint Discover (ULD) Port* Unit Layer Discovery Pro Port*	PAREDS SWEEP	Enabled			
	Port* LLDP Asset ID	THE REPORT OF TH	Licrational			
	LLDP Power Priority*		Unknown			
	Con Perior Provid					

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Figure 86 Device Phone 1000 Cisco Unified CM Administration Window

cisco For Cisco Un	ified CM Administrations		admin About Logo ragement • Bulk Administration • Help •
rectory Number Con			telated Links: Configure Device (SEP00187373C3FA)
	Reset 🛁 Add New		
latus -			
Status: Ready			
Nectory Number Info	mation		
irectory Number* 100 loute Partition Part	10 tition HQ_Phones_IP	-	
escription 100			
Verting Name			
SCII Alerting Name	e trop (TI		
Associated Devices SE	00187371C3FA		
		Edit Device Edit Line Appea	srance
	**		
issociate Devices			
Directory Number Setti /oice Mail Profile			
alling Search Space	<none> CSS-HQ_Phones_IP</none>	Choose -	<none> to use system default)</none>
resence Group*	Standard Presence group	×	
iser Hold MOH Audio So Network Hold MOH Audio	Source 1-SampleAudioSource	×	
uto Answer*	Auto Answer Off	2	
AAR Settings			
AVR Ex	Yoice Mail	AAR Destination Mask	AAR Group
Retain this destination	on in the call		Leaves 2
forwarding history			
Cell Forward and Call F	Voice Mail	Destination	Celling Search Space
Calling Search Space Ac		Destination	Use System Default
Forward All	C or		< None >
Secondary Calling Sears Forward Busy Internal	h Space for Forward All		< None >
Forward Busy External	Ror		CSS-HQ_Phones_IP CSS-HQ_Phones_IP
Forward No Answer Inb			CSS-HQ_Phones_IP
Forward No Answer Ext			CSS-HQ_Phones_IP
Forward No Coverage 1			< None >
Forward No Coverage E Forward on CTI Failure	xternal Por		< None >
Forward Unregistered 1			CSS-HQ_Phones_IP
Forward Unregistered E			CSS-HQ_Phones_IP
Vo Answer Ring Duration	(seconds) 5		
Call Pickup Group	< None >	2	
MLPP Alternate Party 5	iettings		
arget (Destination) 4LPP Calling Search Spa	ce < None >	-	
ALPP No Answer Ring Du	ration (seconds)		
Line Settings for All De	vices		
told Reversion Ring Dura seconds)	the feature	Set	tting the Hold Reversion Ring Duration to zero will disable
fold Reversion Notification seconds)	on Interval disable the feature	Set	tting the Hold Reversion Notification Interval to zero will
Line 1 on Device SEPO Display (Internal Caller	0187371C3FA	Display text for	a line appearance is intended for displaying text such as a
10)	name instead of a directory number proper identity of the caller.	r for internal calls. If you specify	y a number, the person receiving a call may not see the
ASCII Display (Internal Caller ID)			
Line Text Label	[
ASCII Line Text Label			
External Phone Number Mask	4155551XXXX		
Visual Message Waiting Indicator Policy*	Use System Policy	2	
Audible Message	CH.		
Audible Message Waiting Indicator Policy*			
Ring Setting (Phone Idle)*	Use System Default	E	
Ring Setting (Phone Active)	Use System Default	Applies to this l	ine when any line on the phone has a call in progress.
Call Pickup Group Audio Alert Setting	Use System Default	2	
(Phone Idle) Call Rickup Group	Use System Default		
Audio Alert Setting (Phone Active)	Total Alian Contract	_	
Recording Option*	Call Recording Disabled		
Recording Profile Monitoring Calling	< None >	2	
Search Space			
Hultiple Call/Call Wait	ing Settings on Device SEP0018733 the Max Number of calls is: 1-200	IC3FA	
faximum Number of Call	s* (4		
usy Trigger*	2		(Less than or equal to Max. Calls)
Forwarded Call Inform	ation Display on Device SEP001073	71C3FA-	
Caller Name			
Caller Number Redirected Number			
P Dialed Number			
Users Associated with	Line		
	iate End Users		
Save Delete Res	Add New		
) *- indicates require			
) **- Changes to Line	or Directory Number settings require	restart.	

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Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 87 Device Phone 4155551170 Cisco Unified CM Administration Window

Configuration an 💥 Delete 🗋 Copy 🌚 Read: 🚽 A		Contra + Application + User Wanagement + Built Administration + Indo + Back To Principlest Back To Principlest						
an X Delate 📋 Copy 🍟 Reset 🖕 A In Latur: Ready								
ciation Information	Phone Type							
Modify Button Rame	Product Type: Cit Device Protocalt SC	Phane Type Product Type: Citca 7961 Service Protocuit: SCCP						
Line [2] - Add anew DN	Device Information	Repistered with C	ison Unified Communications Manager 1	0.40.97.2				
Reg Add a new SD Reg Add a new SD	Perior Information Reportation IP Address INAC Address	22.40.103.70 DODIE INARIE 78	ises Unified Communications Manager D					
Add a new SD	Cescrption Device Pool*	4155551170						
WgLAdd a new SD Unassigned Associated Items	Control Device	DevicePool_Br1_ < None >	D_Phones Vers.D	etals stals				
Rends a new SD	Phone Button Template Softkey Template	Standard 7961 S	CCP 📃					
SASLANEX SURL SASLANEX SUR SD	Common Phone Profil	standard User	n Phone Profile					
Add a new BLF Directed Call Park	Calling Search Space AAR Calling Search Space	CSS-Br1_Phones	<u>د</u> در					
Call Park Call Park Call Pickup	Media Resource Grou	D BY LINW MIRGL	-					
Conference List Conference	User Hold MOH Audio Source Network Hold MOH Audio Source	1-SampleAudioS						
Do Not Disturb End Call	Audio Source Location*	1-SampleAudioS Hu8_8rS						
Forward All Group Call Pickup	AAR Group	<none></none>	2					
Hold Hunt Group Logout	Network Locale Built in Bridge*	< None >	2					
Intercom Fill - Add a new Intercom	Privacy*	Default	-					
Malicious Call Identification Meet Me Conference	Cevice Mobility Mode	Default	Ven.0	ument				
Mobility New Call	Owner User 3D Phone Personalization	<none></none>	•					
Other Ridup Quality Reporting Tool Redial	Phone Load Name Single Button Barge							
Remove Last Safemant	E to Artist		9					
Transfer Video Mode	Join Across Lines IF Retry Video Call ar	Default Audio	2					
Privacy None	P Allow Control of D	in Indicators Onterna	i cals onis)					
	F Logged Into Hunt	Group						
	E Remote Device							
	Protocol Specific Im Packet Capture Mode Resist Capture Packet	* None	3	3				
	Packet Capture Mode Packet Capture Durat Presence Group* Device Security Profile	son p Standar	d Presence group 2 51 - Standard SCCP Non-Secure Profile					
	Cevce Security Profile SUBSCRIME Calling Se	Cisco H	11 - Standard SCCP Aon-Secure Profile	3				
	C Unattended Port							
	E RFC2033 Disabled							
	Certification Author	Ty Provy Function ((APF) Information	_				
	Certification Author Certificate Operation Authentication Mode	Ry Null Street	operation					
	Operation Completes	Centrate Store) Key San (Mo) Centrate Completes By Coort 11 [17 12 promote Completes						
	Certificate Operation Note: Security Profile	Certificate Operation Status: Rone Note: Security Profile Contains Addition CAPF Settings.						
	Module 1 Load Name	Expansion Module Information Module 1 Caune > Module 1 Load Name						
	Mobile 2							
		Coternal Data Locations Information (Leave blank to use default)						
	External Data Locat Solormation Cirectory	ions Information (L	lave blank to use default)					
	messages							
	Services Authentication Server							
	Proxy Server 35e							
	3de Timer (seconds)							
	Extension Informati	**		_				
	Enable Extension	Mobility a Current Device Set	nga 💌					
	Log in Time < Non Log out Time < Non	#>						
	MUPP Information -							
	MUPP Indication* [0	Mone > efault						
	MULLA Lossenstion."	elaut.	2					
	T Do Not Disturb							
	CND Option* CND throaming Call Air	Finger Off						
	-Secure Shell Inform							
	Secure Shell User Secure Shell Passwor							
	Product Specific Co							
				?				
	E Disable Speakerph E Disable Speakerph	one one and Headset						
	Fi Disable Speakerph Forwarding Delay* PC Port.*		Disabled Enabled	-				
	Settings Access*		Enabled Dnabled	-				
	Gratuitous ARP* PC Voice VLAN Access		Enabled	-				
	Video Capabilities* Auto Line Select*		Disabled Disabled					
	Web Access* Splan to PC Port*		Enabled Disabled	-				
	Logging Display*		PC Controlled	-				
	Recording Tone*		Dinabled					
	Recording Tone Local Recording Tone Remo		100					
	Recording Tone Dural		1					
	RTCP* "more" Soft Kes Time		Disabled B	•				
	Auto Call Select* Log Server		Enabled					
	Advertise 0.722 Code	·	Disabled	-				
	Wideband Headset U Wideband Handset U		Enabled Enabled	-				
	Wideband Headoet* Wideband Handoet*		Enabled Use Phone Default	•				
	Peer Firmware Sharin	g•	Disabled					
	Cisco Discovery Prota Port ⁴ Cisco Discovery Prota	col (CDP): PC Port*	Enabled Enabled	-				
	Linii Layer Discovery I Endpoint Discover (U. Port*	Protocol - Media DP-MEO3 Switch	Enabled	-				
	Link Layer Discovery I Port*	Protocol (LLDP): PC	Enabled					
	LLOP Asset 10		Utiknown					
	LLDP Power Priority*							

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Figure 88 Device Phone 1170 Cisco Unified CM Administration Window

rectory Number Conf		ce Mail + Device + /	and the second second second second	ernent • Bulk Administration • Help •	
			Rel	lated Links: Configure Device (SEP0019EBA88E7B)
] Save 🗙 Delete 🧣	Reset GP Add New				
Status: Ready					
Directory Number Infor	metico				_
Directory Number* 117	0				
Route Partition Par Description 117	tition-Br1_Phones_IP	×			
Verting Name	0				
SCII Alerting Name					
Allow Control of Devices	e from CTI				
associated bevices [SEP	20019E8A88E78		Edit Device		
			Edit Line Appeara	ince	
	**				
Dissociate Devices					
Directory Number Setti /oice Mail Profile	< None >		Choose ch	ione> to use system default)	_
Calling Search Space	CSS-Br1_Phone	IS_IP		to be a state of the state of t	
Presence Group*	Standard Prese		×		
Jser Hold MOH Audio Sou Vetwork Hold MOH Audio	Source 1-SampleAudio		×		
luto Answer*		ith Speakerphone			
AAR Settings					_
	Voice Mail	AAR Der	stination Mask	AAR Group	
AAR C or Retain this destination	and the state of the			< None >	*
forwarding history	on in the call				
Call Forward and Call P					_
Calling Search Space Ad	Voice Mail	Der	stination	Calling Search Space	-
Calling Search Space Ad Forward All	E or			Use System Default Kone >	*
Secondary Calling Search				< None >	1
Forward Busy Internal	□ or [< None >	٠
Forward Busy External	□ or [< None >	٠
Forward No Answer Inte				< None >	٠
Forward No Answer Exb				< None >	
Forward No Coverage In				< None >	
Forward No Coverage E Forward on CTI Failure	oternal ⊡or [< None >	
Forward Unregistered In				< None >	*
Forward Unregistered E				< None >	-
to Answer Ring Duration					_
Call Pickup Group	< None >				
MLPP Alternate Party S	settings				
Target (Destination)					
ALPP No Answer Ring Du		•>			
					_
Line Settings for All De- told Reversion Ring Dura	vices		Setti	ng the Hold Reversion Ring Duration to zero will disab	le
seconds)	the featur	0			
		e feature	Sette	ng the Hold Reversion Notification Interval to zero wil	
fold Reversion Notificatio seconds)	disable th				
	0.0000 07				
Hold Reversion Notification seconds)	019E8A88E78	stops pumber for inter		line appearance is intended for displaying text such a	
Hold Reversion Notification (seconds) Line 1 on Device SEPO0 Display (Internal Caller ID)	019E8A88E78	ctory number for inter caller.		line appearance is intended for displaying text such a number, the person receiving a call may not see the	
Hold Reversion Notificatio seconds) Line 1 on Device SEP00 Display (Internal Caller ID) ASCII Display (Internal Caller ID)	019E8AB8E78	ctory number for inter caller.			
told Reversion Notificatic seconds) Line 1 on Device SEPO0 Display (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label	019E8AB8E78	ictory number for inter aller.			
Idd Reversion Notificatio Seconds) Line 1 on Device SEP00 Display (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label ASCII Line Text Label	name instead of a dire proper identity of the i	ectory number for inter caller.			
Idd Breersion Notificatio seconds) Line 1 an Device SEP00 Display (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label ASCII Line Text Label External Phone Number Mask	ALL CONTRACTORS	ectory number for inter	nal calls. If you specify a		
Idd Reversion Notificatio seconds) Line 1 on Device SCP00 Display (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label ASCII Line Text Label Sutternal Phone Number Mask Visual Message	name instead of a dire proper identity of the i	ectory number for inter			
Iold Baversion Notificatio seconds) Line 1 on Device SEPOD Display (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label ACCII Line Text Label External Phone Number Mask Visual Messagor Policy Michael Caller	ALL CONTRACTORS	ectory number for inter	nal calls. If you specify a		
kold Reversion Notificatio seconds) Line 1 on Device SCP00 Deplay (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label ASCII Line Text Label ASCII Line Text Label Schemal Phone Number Mask Vaula Message Waterg Indicator Pelicy	otoriane instead of a dire name instead of a dire proper identity of the proper identity of the fillsssboocx [Use System Policy [Off	ictory number for inter	nal calls. If you specify a		
kold Reversion Notificatio seconds) Line 1 on Device SCP00 Deplay (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label ASCII Line Text Label ASCII Line Text Label Schemal Phone Number Mask Vaula Message Waterg Indicator Pelicy	019E0ABBE78 name instead of a dire proper identity of the r proper	ectory number for inter	nal calls. If you specify a		
Idel Berersion Notificatio seconds) Une 1 en Device SCPOO Display (Internal Caller ID) (Internal Caller ID) AGCII Display (Internal Caller ID) Une Text Label AGCII Line Text Label External Phone Number Nask Visitadi Maddel Message Watery Indicator Palcy Rog Entrol (Phone Res Schling (Phone Res Schling (Phone	otoriane instead of a dire name instead of a dire proper identity of the proper identity of the fillsssboocx [Use System Policy [Off	ectory number for inter	nal calis. If you specify a		
skol Reversion Notificatio seconds) Une 1 on Device SCPDO Display (Internal Cater ID) ASCII Display (Internal Cater ID) Im Text Label ASCII Line Text Label ASCII Inter Text Label ASCII In	Order 78 name instead of a dree rober identity of the r (1) (4) (4) (5) (5) (5) (5) (5) (5) (5) (5	ectory number for inter	nal calis. If you specify a	number, the person receiving a call may not see the	
skol Reversion Notificatio seconds) Une 1 on Device SEPDO Display (Internal Catter ID) ASCII Display (Internal Catter ID) ASCII Display (Internal Catter ID) ASCII Display (Internal ASCII Line Text Label ASCII Line Text L	Detector 25 0 October 25 anne instead of a dread proper identity of the or proper identity of the or (Use System Policy Otto System Default (Use System Default (Use System Default (Use System Default	ictory number for inter	nai cails. If you speedy a	number, the person receiving a call may not see the	
sido Reversion Notificatio seconds) Line 1 an Device SCPOO Dipplay (Internal Caller ID) ASCII Display (Internal Caller ID) Line Text Label ASCII Line Text Label ASCII Line Text Label Schemal Phone Number Nask Visial Message Watery Indicator Pacify Reg Setting (Interna Active) (Internation Reg Setting (Interna Active) (Internation Active) (Internation	Order Of a direct of the of a direct of the of the off off off off off off off off off of	ictory number for inter	nal calls. If you specify a	number, the person receiving a call may not see the	
skol Reversion Notificatio seconds) Une 1 on Device SEPDO Display (Internal Catter ID) ASCII Display (Internal Catter ID) ASCII Display (Internal Catter ID) ASCII Display (Internal ASCII Line Text Label ASCII Line Text L	Order 20	adler.	nai cails. If you speedy a	number, the person receiving a call may not see the	
Idel Berersion Notificatio seconds) Une 1 en Device SCPDO Display (Internal Caller ID) Labor (Internal Caller ID) Labor (Internal Caller ID) Labor Text Label ACCII Display (Internal ACCII Display (Internal ACCII Display (Internal Number Nask Valad Message Watery Indicator Number Indicator Nadels Message Watery Indicator Nadels Message Messa	Detector 25 0 October 25 anne instead of a dread proper identity of the or proper identity of the or (Use System Policy Otto System Default (Use System Default (Use System Default (Use System Default	adler.	nai cails. If you speedy a	number, the person receiving a call may not see the	
seconds) time 1 an Device SEPDG Display (Internal Caler ID) ASCII Display (Internal Caler ID) Inter Text Label ASCII Display (Internal Caler ID) External Ponce Number Text Label ASCII Display (Internal ASCII Display (Internal ASCII Display (Internal ASCII Display (Internal ASCII Display (Internal ASCII Display (Internal Ascie Nessoa Water Internal Ascie Nessoa Water Internal Resolution (Interna Ascie Nessoa Water Internal Resolution (Internal Ascie Nessoa (Internal Resolution (Internal Ascie Nessoa (Internal Ascie Nessoa (Internal	Order 20	adler.	nai cails. If you speedy a	number, the person receiving a call may not see the	
Idel Berersion Notificatio seconds) Une 1 en Device SEPDO Display (Unternal Catter ID) Catter ID) Line Text Label ASCII Line Text Label Schemal Phone Number Majk. Wathog Indicator Paley Sumg Setting (Phone Eds) Sing Setti	Concernent	ed .	nai cails. If you speedy a a a a a a a a a a a a a a	number, the person receiving a call may not see the	
seconds) time an Device SEPOG Display (Internal Caler ID) ASCII Display (Internal Caler ID) ID) ID (Internal Caler ID) ID) ID (Internal ASCII Display (Internal Caler ID) External Prone External Prone External Prone External Prone Active) Machine Ressage Waaroo Machine Ressage Waaroo Res Setting (Interne Active) Res Setting (Interne Active) Res Setting (Interne Active) Res Setting (Interne Active) Res Setting (Interne Active) Resorting Option Resorting Option Resorting Option Resorting Option Resorting Option Resorting Option Resorting Caling Setting Esting (Caling Setting Caling Setting Esting (Caling Setting Caling Setting Esting (Caling Setting Caling Setting Esting (Caling Setting Esting) Resorting Caling Setting Esting Resorting Caling Setting Esting Resorting Caling Setting Esting Resorting Caling Setting Esting Resorting Caling Setting Resorting Caling Resorting Cal	Constanting of a direct o	nd	nai cails. If you speedy a a a a a a a a a a a a a a	number, the person receiving a call may not see the	
slod Reversion Notificatio seconds) Units 1 an Device SCPOD Display (Internal Caller ID) ASCII Display (Internal Caller ID) Into Text Label ASCII Display (Internal ASCII DIsp	Constanting of a direct o	ed ed (POD19ERABSE78- 16: 1-20	nai cails. If you speedy a a a a a a a a a a a a a a	number, the person receiving a call may not see the a when any line on the phone has a call in progress.	
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Device: Trunk Parameters

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



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]		Name 🕈	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	Trunk Security Profile
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Figure 90 Device Trunk Enterprise 1 HQ CUBE1 Phones Analog Cisco Unified CM Administration Window

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

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	Communications Solutions
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tatus	
Status: Ready	
evice Information	
oduct:	SIP Trunk
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	Ent1-HQ-CUBE1
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ommon Device Configuration	
all Classification*	Use System Default
edia Resource Group List	HQ HW MRGL
	Trunk HQ
AR Group	< None >
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Media Termination Point Re	quired
Retry Video Call as Audio	
Transmit UTF-8 for Calling P	arty Name
Unattended Port	
	reemption (MLPP) Information
LPP Domain < None >	×
all Routing Information	
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Figure 92 Device Trunk Enterprise 1 Branch 1 CUBE1 Phones Analog Cisco Unified CM Administration Window

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

cisco Unifie			ation			Navigation Cisco	Unified CM Adminis	tration 💌
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nk Configuration						Related Links	Back To Find/L	ist 💌
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atus —								
Status: Ready								
evice Information								
oduct:	SIP Trunk							
evice Protocol: evice Name*	SIP 10.80.80.82							
scription	Ent1-Br1-CU							
vice Pool*	DevicePool			•				
mmon Device Configuration								
II Classification*	Use System	Default						
adia Resource Group List	Br1 HW MR0	3L		•				
cation*	Trunk Br1			•				
R Group	< None >			•				
cket Capture Mode*	None			•				
cket Capture Duration	0							
Media Termination Point Re	quired							
Retry Video Call as Audio Transmit UTF-8 for Calling F	Party Name							
Unattended Port	arcy Name							
fultilevel Precedence and P	reemption (MLPP) Informa						
LPP Domain < None >			•					
all Routing Information —								
Inbound Calls								
Significant Digits*	4				-			
Connected Line ID Presentat					•			
Connected Name Presentatio					-			
Calling Search Space		1_Phones_IP			-			
AAR Calling Search Space Prefix DN	< None	>			·			
	der Delivery	tehound						
Redirecting Diversion Head	der Dellvery -	Indound						
Outbound Calls								
Calling Party Selection*	Originator			•				
Calling Line ID Presentation* Calling Name Presentation*				•				
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Caller Name								
_	der Delivers	Outbourd						
Redirecting Diversion Hear	uer Delivery -	Jubound						
IP Information								
estination Address*	1	0.80.80.82						
Destination Address is an S	SRV							
estination Port*		060						
TP Preferred Originating Cod	lec*	11ulaw			*			
esence Group*	-	Standard Prese			V			
P Trunk Security Profile*	-	Non Secure SIP	Trunk Prof	file	•			
prouting Calling Search Space		< None >			-			
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P Profile*		< None >	ofilo		<u> </u>			
MF Signaling Method*	-	Standard SIP Pr	onie		 ▼ 			
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Save Delete Reset	Add New							
*- indicates required item	n.							
	quired for ch							

1

Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration

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

Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration

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

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

I

port 2/1
!
Ent1_HQ_VG224#

Enterprise 1 HQ Cisco ASA Firewall Example Configuration

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

```
Ent1-HQ-ASA#
interface Vlan65
nameif inside
security-level 100
ip address 10.40.99.1 255.255.255.0
1
interface Vlan70
nameif outside
security-level 0
ip address 10.40.98.2 255.255.255.0
Т
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
```

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

```
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
1
service-policy global_policy interface inside
service-policy outsidein interface outside
prompt hostname context
: end
Ent1-HQ-ASA#
```

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

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

```
Ent1_Br1#
```

I

```
!
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 media-renegotiate
```

```
fax protocol pass-through g711ulaw
modem passthrough nse codec g711ulaw
 sip
 min-se 90
 header-passing error-passthru
 midcall-signaling passthru
Т
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
1
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
1
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
I.
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
Т
voice-port 2/1/0
1
voice-port 2/1/1
1
voice-port 4/0/0
1
voice-port 4/0/1
!
voice-port 4/0:23
1
ccm-manager mgcp
1
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
1
mgcp profile default
1
sccp local GigabitEthernet0/1.1
sccp ccm 10.40.97.2 identifier 1 priority 1 version 6.0
sccp ip precedence 3
sccp
I.
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
I.
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
1
1
dial-peer voice 2000 voip
 description *** Voice: LAN to WAN - Incoming Dial-Peer ***
huntstop
```

I

```
codec g729r8
 session protocol sipv2
incoming called-number 6T
dtmf-relay rtp-nte digit-drop
no vad
1
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
1
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
1
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 415555111[0,1]
 dtmf-relay rtp-nte digit-drop
codec g711ulaw
no vad
1
dial-peer voice 3101 voip
 description *** Fax: WAN to LAN - Outgoing Dial-Peer ***
huntstop
destination-pattern 415555111[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
1
dial-peer voice 1 pots
service mgcpapp
port 4/0/0
I.
dial-peer voice 2 pots
service mgcpapp
port 4/0/1
Т
dial-peer hunt 3
sip-ua
authentication username yyyyy password 7 xxxxxxxxx
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
1
Ent1_Br1#
```

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

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To integrate the Branch 1 Cisco Unity Express with Cisco Unified CM configuration, see the *CallManager for Cisco Unity Express Configuration Example*.

Cisco Validated Design

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