



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

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Introduction

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

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

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

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

Network Topology

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

Enterprise Headquarter

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

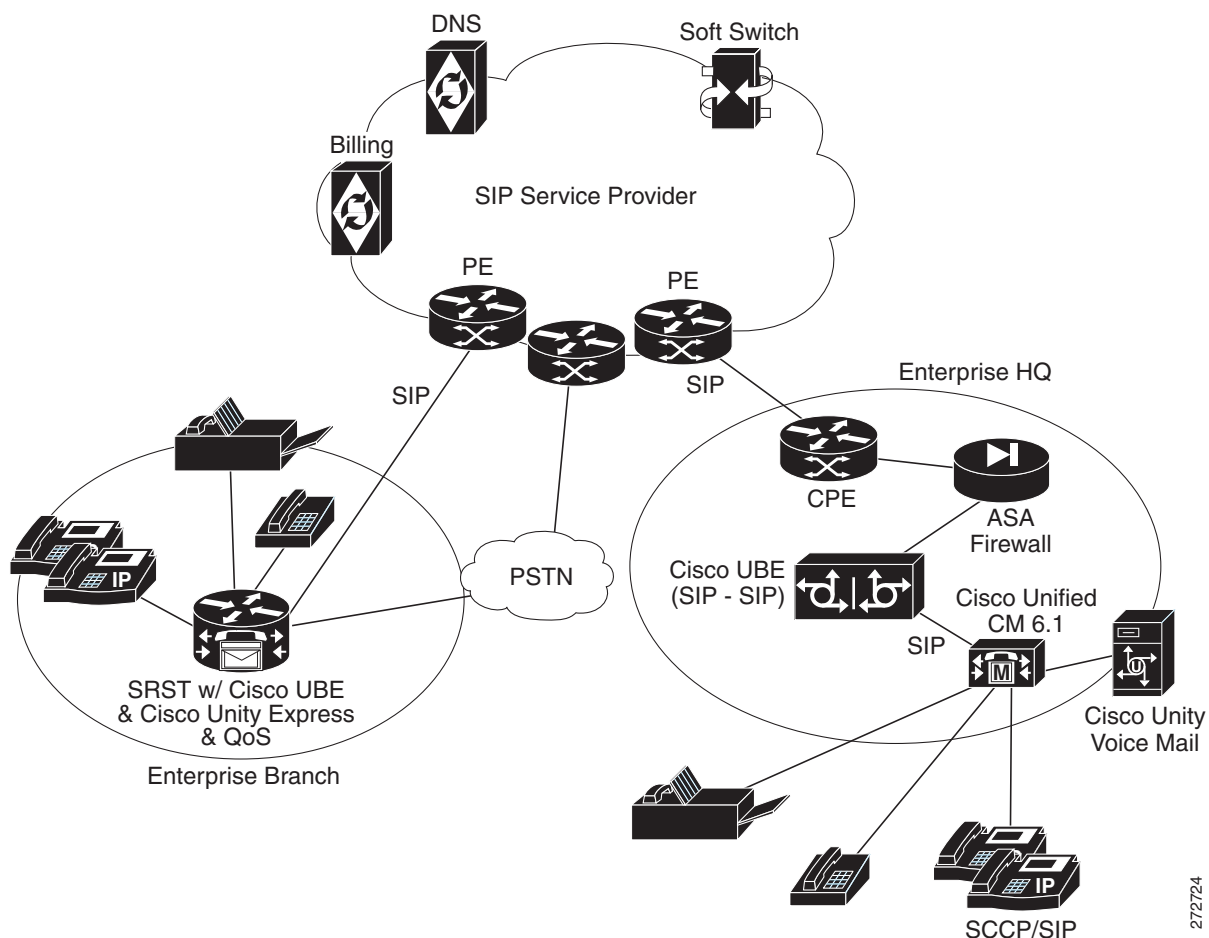
Enterprise Branch

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

Service Provider

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

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



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Prerequisites

Prerequisites are grouped into the following sections:

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

Components Used

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

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

Cisco Unified Communications Manager

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

Cisco Unified Border Element

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

SCCP Analog Voice Gateway

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

Voice Mail at the Enterprise Headquarter Site

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

Cisco Adaptive Security Appliance Firewall Appliance

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



Note

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

Cisco Survivable Remote Site Telephony

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

Cisco IOS Software Releases

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

Conventions

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

Solution Description

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

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

The focus of services using this solution are:

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

The following topics describe the solution:

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

Feature Summary

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

Enterprise Headquarter Site Features

- Cisco Unified Communications Manager call control

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

Enterprise Branch Offices Features

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

Service Provider Features

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

Basic Phone Features Served in the Topology

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

SIP Trunking Design Considerations

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

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

DTMF Transport

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

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

OOB SIP DTMF signaling methods include:

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

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

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

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

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

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

SIP Delayed Offer and Early Offer

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

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

**Note**

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

Early Media Cut Through

The terms Early Offer and Early Media are often confused.

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

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

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

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

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

SIP Trunk Transport Protocols

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

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

Monitoring SIP Trunk State

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

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

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

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

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

SIP Trunk Redundancy and Load Balancing

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

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

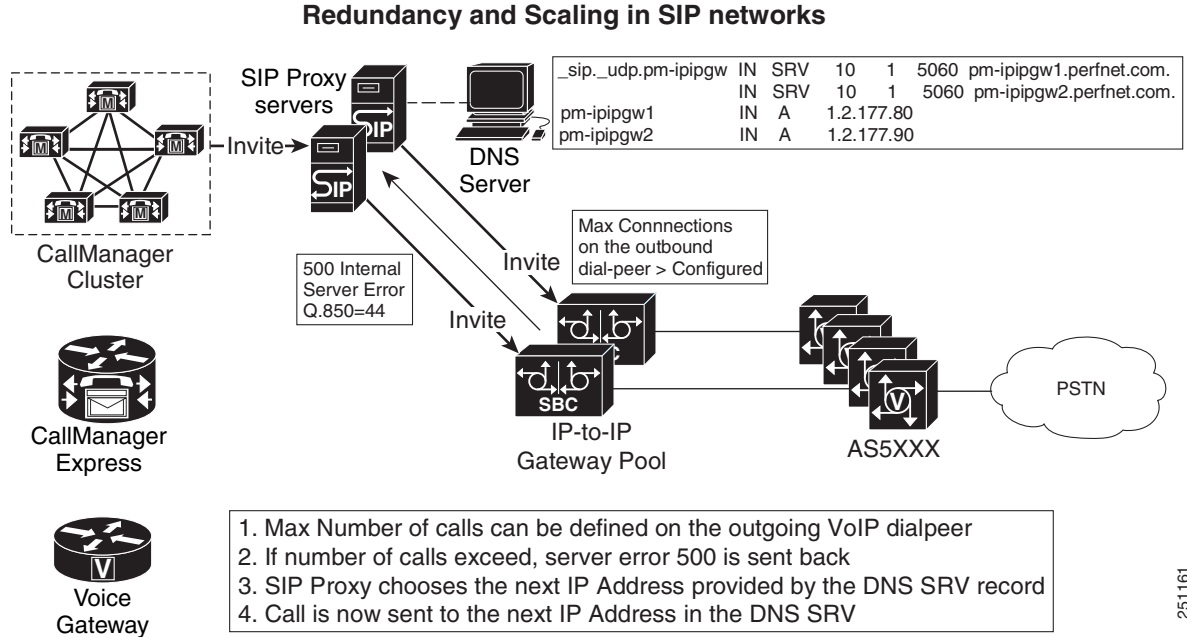
Dial-peer preferences and Dial-peer Hunting

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

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

DNS SRV

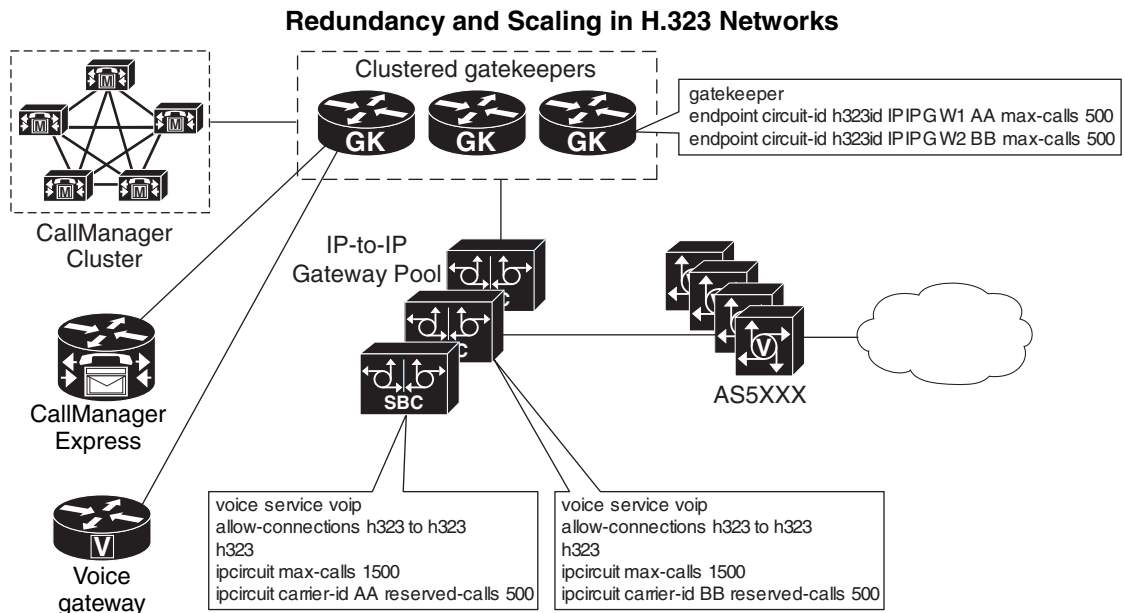
Use the setup example shown in [Figure 2](#) into achieve redundancy based on DNS SRV.

Figure 2 SIP Network Redundancy and Scaling Based on DNS SRV

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GK load balancing for H.323 Networks

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

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

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Route List & Route Group option from CCM

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

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

Figure 4 *Configuring Route Groups*

Route Group Configuration

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

Route Group Members

15.3.30.60
 H.323

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

Route Group Information

Route Group Name* loadbalance
 Distribution Algorithm* Top Down

Route Group Member Information

Find Devices to Add to Route Group

Device Name contains Find

Available Devices
 (select device, then
 select port below)

15.3.30.70
 pinamojito-ipipgw1-15.5.15.80

Port(s) All Add to Route Group

Current Route Group Members

Reverse Order of Selected Devices

Selected Devices*
 (ordered by highest
 priority)

15.3.30.60 (All Ports)

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

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

Figure 5 *Configuring A Route List for Route Groups*

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

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

Find Route Groups where Route Group Name

and show items per page

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

Matching record(s) 1 to 2 of 2

<input type="checkbox"/>	Route Group Name
<input type="checkbox"/>	loadbalance-ippgw60-rg
<input type="checkbox"/>	loadbalance-ippgw70-rg

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

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

Route List Configuration

Route List Details

loadbalance-ippgw60-rg

loadbalance-ippgw70-rg

Route List: loadbalance-ippgw-rl

Status: Ready

Copy

Update

Delete

Reset

Add a new Route List

Back to Find/List Route Lists

Dependency Records

Route List Information

Route List Name*

loadbalance-ippgw-rl

Description

loadbalancebetween60-70

Cisco CallManager Group*

PUB

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 priority)

loadbalance-ippgw60-rg[non-QSIG]
loadbalance-ippgw70-rg[non-QSIG]

Removed Groups
(to be removed from
Route List when you
click Update)

* indicates required item

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

Figure 7 *Configuring Max-Con*

Route Pattern Configuration

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

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

Copy Update Delete

Pattern Definition

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

Call Classification* OffNet ☐ Allow Device Override
☒ Provide Outside Dial Tone ☐ Allow Overlap Sending ☐ Urgent Priority
☐ Require Forced Authorization Code
 Authorization Level 0
☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask
 Calling Party Transform Mask
 Prefix Digits (Outgoing Calls)
 Calling Line ID Presentation Default
 Calling Name Presentation Default

Connected Party Transformations

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IP Connectivity

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

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

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

Table 1 *Cisco CPE Router Network Connectivity Options*

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

Quality of Service

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

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

Congestion Management

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

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

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

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

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

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

Packet Marking

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

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

Call Admission Control

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

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

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

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

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



Note

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

Delay

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

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

Echo

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

Voice Mail

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

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

Dial Plan

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

Security

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

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

Authentication

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

Encryption of Media and Signaling

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

Firewall

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

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

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

Failover and Redundancy

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

Fax and Modem

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

Billing and Management

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

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

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

Best Practices for SIP Trunk implementation Using Cisco UBE

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

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

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

Caveats

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

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

Configurations

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

**Note**

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

Configuration Verification

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

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

The firewall configuration can be verified with the following commands:

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

Troubleshooting

**Note**

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

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

- **debug ccsip messages**

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

- **debug ccsip all**

This command enables all SIP-related debugging including:

- **debug voip app**

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

- **debug voip ccapi inout**

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

- **debug ephone mtp**

This command enables Media Termination Point (MTP) debugging.

- **debug sccp events**

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

Related Information

The following information is referenced in this guide:

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

Obtaining Documentation and Submitting a Service Request

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

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

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

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

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

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

Overview of Test Configurations

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

Enterprise 1 HQ Components

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

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

Enterprise 1 and Branch 1 Components

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

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

Caveats

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

Global Caveats

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

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

Cisco Unified CM 6.1.0.9901-372 Caveats

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

High-Level Operation

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

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

Call Flow Within Enterprise 1

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

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

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

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

HQ Call Flow to Enterprise Offsite Remote Endpoint

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

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

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

Branch 1 Call Flow to Enterprise Offsite Remote Endpoint

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

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

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



Note

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

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

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

High-Level Configuration Summaries

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

Protocols

The following is a list of protocols used between components:

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

Codecs

The following is a list of codecs used between components:

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

**Note**

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

DSP Farms

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

Supplementary Services

The following is a list of supplementary services.

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

Call Admission Control

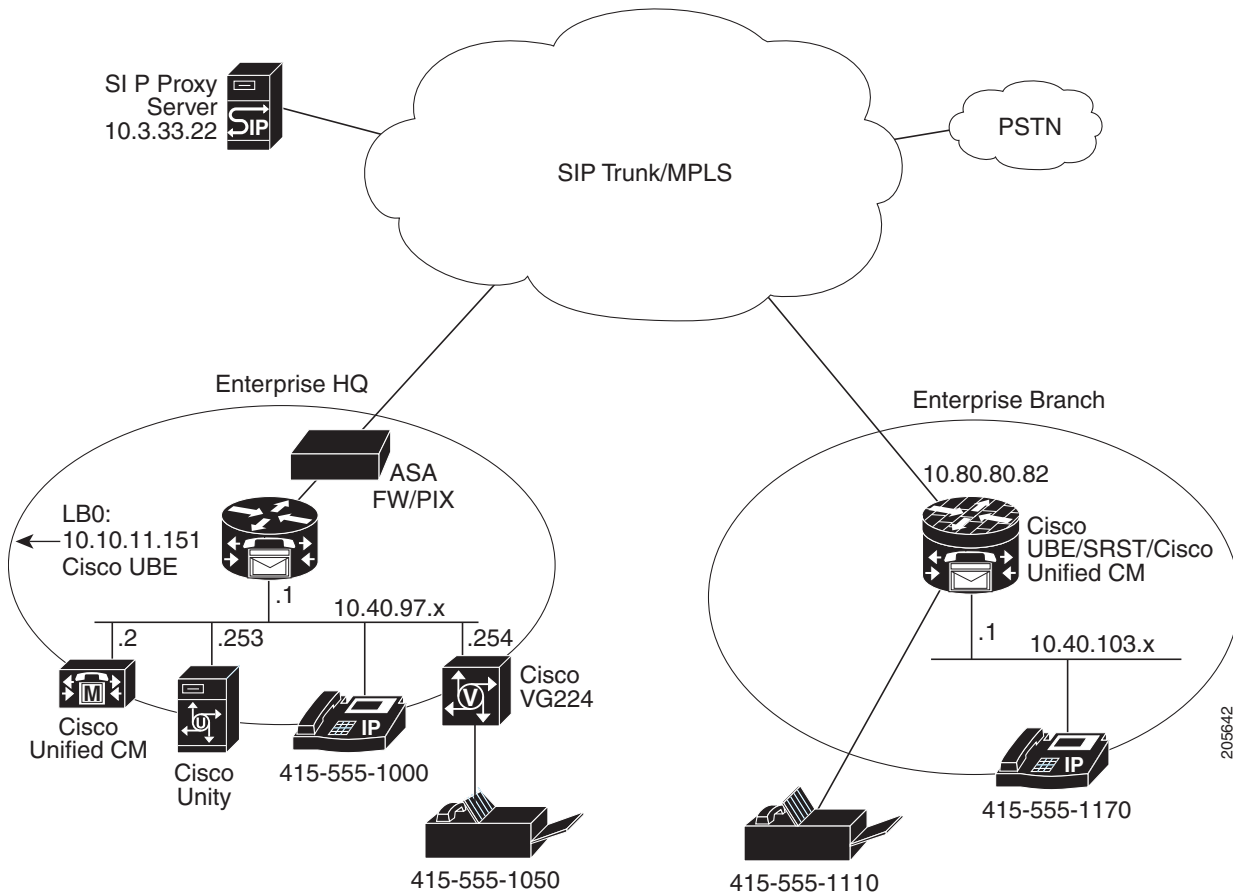
The call admission control (CAC) restrictions that are imposed by Cisco Unified CM for the whole enterprise are as follows:

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

Test Topology

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

Figure 8 **Test Topology**



Example Configuration Details

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

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

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

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

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

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

Enterprise 1 HQ Cisco UBE Example Configuration

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

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

```

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

```

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

```

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

```

Enterprise 1 HQ Cisco Unified CM Example Configuration

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

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

Configuring the Cisco Unified CM System Parameters

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

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

System: Server Parameters

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

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

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation menu with options like 'System', 'Call Routing', 'Media Resources', 'Voice Mail', 'Device', 'Application', 'User Management', 'Bulk Administration', and 'Help'. The main content area is titled 'Server Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. Below this, there are buttons for 'Save', 'Delete', and 'Add New'. A status message indicates 'Update successful'. The 'Server Information' section contains the following fields:

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

At the bottom, there are buttons for 'Save', 'Delete', and 'Add New'. A note indicates that '*' indicates a required 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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Find and List Regions

+ Add New Select All Clear All Delete Selected

— Status —
12 records found

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

Find Regions where Name begins with Find Clear Filter + -

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

Add New Select All Clear All Delete Selected

273752

Figure 11 System Region Default Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

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

Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

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

Save Delete Reset Add New

Footnote:

*- indicates required item.

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name*

Region Relationships

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

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Legend:

- *- 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.

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_Br1_DSPfarm

Region Relationships

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

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Legend:

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

273755

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_Br1_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: Region(s) not displayed Use System Default Use System Default Use System Default

Modify Relationship to other Regions

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

Save Delete Reset Add New

Legend:

- *- 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.

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin | About | Logout

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Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* 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 Region_Br1_Phones_Analog Region_Br1_DSPfarm Region_Br1_DSPfarm_Conference Region_Br1_DSPfarm_Transcoder	Keep Current Setting	<input checked="" type="radio"/> Keep Current Setting <input type="radio"/> Use System Default <input type="radio"/> None <input type="text"/> kbps	Keep Current Setting

Save Delete Reset Add New

Footnote:

*- indicates required item.

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

273757

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_Br1_Phones_IP

Region Relationships

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

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Legend:

- *- 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.

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

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Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* 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	<input checked="" type="radio"/> Keep Current Setting	Keep Current Setting
Region Br1 Phones Analog		<input type="radio"/> Use System Default	
Region_Br1_DSPfarm		<input type="radio"/> None	
Region_Br1_DSPfarm_Conference		<input type="radio"/> kbps	
Region_Br1_DSPfarm_Transcoder			

Save Delete Reset Add New

Footnote:

*- indicates required item.

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

273759

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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Region Configuration Related Links: Back To Find/List

Save Delete Reset 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	G.729	384	Use System Default

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Legend:

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

273760

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin | About | Logout

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Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_HQ_DSPfarm_Transcoder

Region Relationships

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

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Footnote:

*- indicates required item.

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

273761

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

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Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_HQ_Phones_Analog

Region Relationships

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

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Legend:

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

273762

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

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

Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_HQ_Phones_IP

Region Relationships

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

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Legend:

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

273763

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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Region Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Region Information

Name* Region_Wan

Region Relationships

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

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

Modify Relationship to other Regions

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

Save Delete Reset Add New

Information:

- *- 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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

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Find and List Device Pools

+ Add New Select All Clear All Delete Selected

Status
8 records found

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

Find Device Pool where Device Pool Name begins with Find Clear Filter

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

Add New Select All Clear All Delete Selected

273765

Figure 24 System Device Pool Default Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

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Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

— Status —
Status: Ready

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

— Device Pool Settings —

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

— Roaming Sensitive Settings —

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

— Device Mobility Related Information****

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

Save Delete Copy Reset Add New

* - indicates required item.

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

***leave blank to use default.

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

273766

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

—Status—
Status: Ready

—Device Pool Information—
Device Pool: DevicePool_Br1_Analog_Phones (2 members**)

—Device Pool Settings—

Device Pool Name*	DevicePool_Br1_Analog_Phones
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Reverted Call Focus Priority	Default

—Roaming Sensitive Settings—

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

—Device Mobility Related Information****

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

Save Delete Copy Reset Add New

Legend:

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

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Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

Device Pool Information
Device Pool: DevicePool_Br1_DSPfarm (3 members**)

Device Pool Settings

Device Pool Name*	DevicePool_Br1_DSPfarm
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Reverted Call Focus Priority	Default

Roaming Sensitive Settings

Date/Time Group*	CMLocal
Region*	Region_Br1_DSPfarm
Media Resource Group List	Br1 HW MRGL
Location	Hub_Br1
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default
Physical Location	< None >
Device Mobility Group	< None >

Device Mobility Related Information****

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

Save Delete Copy Reset Add New

Legend:

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

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Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

—Status—
Status: Ready

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

—Device Pool Settings—

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

—Roaming Sensitive Settings—

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

—Device Mobility Related Information****

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

Save Delete Copy Reset Add New

—

i *- indicates required item.

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

i ***leave blank to use default.

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

273769

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

— Status —
Status: Ready

— Device Pool Information —
Device Pool: DevicePool_HQ_Analog_Phones (3 members**)

— Device Pool Settings —

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

— Roaming Sensitive Settings —

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

— Device Mobility Related Information****

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

Save Delete Copy Reset Add New

*- indicates required item.

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

***leave blank to use default.

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

273770

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Device Pool Information

Device Pool: DevicePool_HQ_DSPfarm (3 members**)

Device Pool Settings

Device Pool Name* DevicePool_HQ_DSPfarm

Cisco Unified Communications Manager Group* Default

Calling Search Space for Auto-registration < None >

Reverted Call Focus Priority Default

Roaming Sensitive Settings

Date/Time Group* CMLocal

Region* Region_HQ_DSPfarm

Media Resource Group List HQ HW MRGL

Location Hub_HQ

Network Locale < None >

SRST Reference* Disable

Connection Monitor Duration***

Single Button Barge* Default

Join Across Lines* Default

Physical Location < None >

Device Mobility Group < None >

Device Mobility Related Information****

Device Mobility Calling Search Space < None >

AAR Calling Search Space < None >

AAR Group < None >

Save Delete Copy Reset Add New

*- indicates required item.

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

***leave blank to use default.

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

273771

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Device Pool Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status
Status: Ready

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

Device Pool Settings

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

Roaming Sensitive Settings

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

Device Mobility Related Information****

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

Save Delete Copy Reset Add New

*- indicates required item.

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

***leave blank to use default.

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

273772

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

Cisco Unified CM Administration

Navigation Cisco Unified CM Administration admin About Logout

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

Device Pool Configuration

Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Device Pool Information

Device Pool: DevicePool_WAN (2 members**)

Device Pool Settings

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

Roaming Sensitive Settings

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

Device Mobility Related Information****

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

Save Delete Copy Reset Add New

*- indicates required item.

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

***leave blank to use default.

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

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
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Find and List Locations

+ Add New Select All Clear All Delete Selected

— Status —
5 records found

Locations (1 - 5 of 5) Rows per Page: 50

Find Locations where Location ▾ begins with ▾ Find Clear Filter + -

<input type="checkbox"/>	Location ^	Audio Bandwidth	Video Bandwidth	Copy
<input type="checkbox"/>	Hub_Br1	85	NONE	
<input type="checkbox"/>	Hub_HQ	110	NONE	
<input type="checkbox"/>	Hub_None	UNLIMITED	UNLIMITED	
<input type="checkbox"/>	Trunk_Br1	85	NONE	
<input type="checkbox"/>	Trunk_HQ	110	NONE	

Add New Select All Clear All Delete Selected

273774

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Location Configuration Related Links: Back To Find/List

Save Delete Copy Add New Resync Bandwidth

— Status —
Status: Ready

— Location Information —
Name* Hub_Br1

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

— Video Calls Information —
Video Bandwidth* ☒ None ☐ Unlimited ☐ kbps

— Location RSVP Settings —

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

— Modify Setting(s) to Other Locations —

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

Save Delete Copy Add New Resync Bandwidth

i *- indicates required item.

273775

Figure 34 System Location Hub HQ Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Location Configuration Related Links: Back To Find/List

Save Delete Copy Add New Resync Bandwidth

Status

Status: Ready

Location Information

Name* Hub_HQ

Audio Calls Information

Audio Bandwidth* ☐ Unlimited ☒ 110 kbps

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

Video Calls Information

Video Bandwidth* ☒ None ☐ Unlimited ☐ kbps

Location RSVP Settings

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

Modify Setting(s) to Other Locations

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

Save Delete Copy Add New Resync Bandwidth

***** - Indicates required item.

Figure 35 System Location Hub None Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation dropdown menu. Below the navigation bar is a breadcrumb trail: System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > Help. The main content area is titled 'Location Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. The configuration form is divided into several sections: Status (Ready), Location Information (Name: Hub_None), Audio Calls Information (Audio Bandwidth: Unlimited), Video Calls Information (Video Bandwidth: Unlimited), Location RSVP Settings (NOTE: Location(s) not displayed, Use System Default), and Modify Setting(s) to Other Locations (Location list: Hub_Br1, Hub_HQ, Hub_None, Trunk Br1, Trunk HQ; RSVP Setting: Use System Default). At the bottom, there are buttons for Save, Copy, Add New, and Resync Bandwidth, and a note indicating that asterisks (*) denote required items.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Location Configuration Related Links: Back To Find/List

Save Copy Add New Resync Bandwidth

Status
Status: Ready

Location Information
Name* Hub_None

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

Video Calls Information
Video Bandwidth* ☐ None ☒ Unlimited ☐ kbps

Location RSVP Settings

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

Modify Setting(s) to Other Locations

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

Save Copy Add New Resync Bandwidth

*- indicates required item.

273777

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

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

Location Configuration Related Links: Back To Find/List

Save Delete Copy Add New Resync Bandwidth

Status
Status: Ready

Location Information
Name* Trunk Br1

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

Video Calls Information
Video Bandwidth* ☒ None ☐ Unlimited ☐ kbps

Location RSVP Settings

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

Modify Setting(s) to Other Locations

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

Save Delete Copy Add New Resync Bandwidth

*- indicates required item.

273778

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

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

Location Configuration Related Links: Back To Find/List

Save Delete Copy Add New Resync Bandwidth

Status
Status: Ready

Location Information
Name* Trunk HQ

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

Video Calls Information
Video Bandwidth* ☒ None ☐ Unlimited ☐ kbps

Location RSVP Settings

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

Modify Setting(s) to Other Locations

Location	RSVP Setting
Hub_Br1 Hub_HQ Hub_None Trunk_Br1 Trunk_HQ	Use System Default

Save Delete Copy Add New Resync Bandwidth

*- indicates required item.

273779

System: SRST Parameters

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

SRST Reference Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

SRST Reference Status

SRST Reference: SRST_Ent1_Br1 (used by 13 devices)

SRST Reference Information

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

Save Delete Copy Reset Add New

*- indicates required item.

273780

Configuring the Cisco Unified CM Call Routing Parameters


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

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

Call Routing: Route/Hunt Parameters

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

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




Cisco Unified CM Administration
For Cisco Unified Communications Solutions


Navigation Cisco Unified CM Administration


admin | About | Logout


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

Find and List Route Patterns


 Add New

 Select All



 Clear All





 Delete Selected

Status

 4 records found

Route Patterns (1 - 4 of 4) Rows per Page 50

Find Route Patterns where Pattern ▾ begins with ▾ Find Clear Filter  

<input type="checkbox"/>	Pattern ^	Description	Partition	Route Filter	Associated Device	Cop
<input type="checkbox"/>	9.1XXXXXXXXXX	RP Ent1-HQ IP Phone LongDistance	Partition-HQ_Phones_IP		10.10.11.151	
<input type="checkbox"/>	9.1XXXXXXXXXX	RP Ent1-HQ Analog Phone LongDistance	Partition-HQ_Phones_Analog		10.10.11.151	
<input type="checkbox"/>	9.1XXXXXXXXXX	RP Ent1-Br1 Analog Phone LongDistance	Partition-Br1_Phones_Analog		10.80.80.82	
<input type="checkbox"/>	9.1XXXXXXXXXX	RP Ent1-Br1 IP Phone LongDistance	Partition-Br1_Phones_IP		10.80.80.82	

Add New

Select All

Clear All

Delete Selected

273703

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

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

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status

Status: Ready

Pattern Definition

Route Pattern* 9.1XXXXXXXXXX

Route Partition Partition-HQ_Phones_IP

Description RP Ent1-HQ IP Phone LongDistance

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Gateway/Route List* 10.10.11.151 (Edit)

Route Option

☒ Route this pattern

☐ Block this pattern No Error

Call Classification* OffNet

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

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls) 6

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

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

Save Delete Copy Add New

i *- indicates required item.

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

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

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition

Route Pattern* 9.1XXXXXXXXXX

Route Partition Partition-HQ_Phones_Analog

Description RP Ent1-HQ Analog Phone LongDistance

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Gateway/Route List* 10.10.11.151 (Edit)

Route Option
☒ Route this pattern
☐ Block this pattern No Error

Call Classification* OffNet

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

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

Calling Party Transformations

☒ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls) 7

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

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

Save Delete Copy Add New

i *- indicates required item.

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

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

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition

Route Pattern* 9.1XXXXXXXXXX

Route Partition Partition-Br1_Phones_Analog

Description RP Ent1-Br1 Analog Phone LongDistance

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Gateway/Route List* 10.80.80.82 (Edit)

Route Option
☒ Route this pattern
☐ Block this pattern No Error

Call Classification* OffNet

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

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask 41555XXXX

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls) 7

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

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

Save Delete Copy Add New

i *- indicates required item.

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration admin About Logout

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

Route Pattern Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Status: Ready

Pattern Definition

Route Pattern* 9.1XXXXXXXXXX

Route Partition Partition-Br1_Phones_IP

Description RP Ent1-Br1 IP Phone LongDistance

Numbering Plan -- Not Selected --

Route Filter < None >

MLPP Precedence* Default

Gateway/Route List* 10.80.80.82 (Edit)

Route Option
☒ Route this pattern
☐ Block this pattern No Error

Call Classification* OffNet

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

☐ Require Forced Authorization Code

Authorization Level* 0

☐ Require Client Matter Code

Calling Party Transformations

☐ Use Calling Party's External Phone Number Mask

Calling Party Transform Mask 41555XXXX

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Default

Calling Name Presentation* Default

Connected Party Transformations

Connected Line ID Presentation* Default

Connected Name Presentation* Default

Called Party Transformations

Discard Digits PreDot

Called Party Transform Mask

Prefix Digits (Outgoing Calls) 6

ISDN Network-Specific Facilities Information Element

Network Service Protocol -- Not Selected --

Carrier Identification Code

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

Save Delete Copy Add New

i *- indicates required item.

Call Routing: Class of Control Parameters

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

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

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below the navigation bar is a menu bar with options: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Find and List Partitions". It includes a toolbar with buttons: Add New, Select All, Clear All, and Delete Selected. Below the toolbar, a status bar indicates "4 records found". The main table displays the following data:

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

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

273708

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

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the subtitle 'For Cisco Unified Communications Solutions'. A navigation dropdown menu is open, showing options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The 'Call Routing' menu is selected, and the 'Partition Configuration' window is active. The window title is 'Partition Configuration' and it includes a 'Related Links' section with a 'Back To Find/List' link. Below the title bar, there are icons for Save, Delete, Reset, and Add New. The main content area is divided into sections: 'Status' (Ready) and 'Partition Information'. The 'Partition Information' section contains fields for Name* (Partition-Br1_Phones_Analog), Description (Analog Phones), Time Schedule (< None >), and Time Zone (Originating Device selected, Specific Time Zone: Greenwich Standard Time). At the bottom, there are buttons for Save, Delete, Reset, and Add New, and a note indicating that '*' indicates a required item.

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Partition Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Partition Information

Name* Partition-Br1_Phones_Analog

Description Analog Phones

Time Schedule < None >

Time Zone
☒ Originating Device
☐ Specific Time Zone Greenwich Standard Time

Save Delete Reset Add New

*- indicates required item.

273709

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

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation dropdown menu. Below this is a secondary navigation bar with tabs for System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Partition Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. A toolbar at the top of the main area contains icons for Save, Delete, Reset, and Add New. The 'Status' section shows 'Status: Ready'. The 'Partition Information' section contains the following fields: Name* (Partition-Br1_Phones_IP), Description (IP Phones), Time Schedule (< None >), Time Zone (radio buttons for Originating Device and Specific Time Zone, with Greenwich Standard Time selected), and a bottom toolbar with Save, Delete, Reset, and Add New buttons. A note at the bottom states '*- indicates required item.'

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

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

Partition Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Partition Information

Name* Partition-Br1_Phones_IP

Description IP Phones

Time Schedule < None >

Time Zone
☒ Originating Device
☐ Specific Time Zone Greenwich Standard Time

Save Delete Reset Add New

*- indicates required item.

273710

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

The screenshot displays the Cisco Unified CM Administration web interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and the subtitle 'For Cisco Unified Communications Solutions'. A navigation menu on the right shows 'Cisco Unified CM Administration' selected. Below this, a secondary menu lists various administration areas: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled 'Partition Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. A toolbar at the top of the main area contains icons for Save, Delete, Reset, and Add New. The 'Status' section shows 'Status: Ready'. The 'Partition Information' section contains the following fields: 'Name*' (Partition-HQ_Phones_Analog), 'Description' (Analog Phones), 'Time Schedule' (< None >), and 'Time Zone' (Originating Device selected, Specific Time Zone: Greenwich Standard Time). At the bottom of the form, there are buttons for Save, Delete, Reset, and Add New. A note at the bottom left states: '*- indicates required item.'

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

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

Partition Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Partition Information

Name* Partition-HQ_Phones_Analog

Description Analog Phones

Time Schedule < None >

Time Zone ☒ Originating Device ☐ Specific Time Zone Greenwich Standard Time

Save Delete Reset Add New

*- indicates required item.

273711

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Partition Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Partition Information

Name* Partition-HQ_Phones_IP

Description IP Phones

Time Schedule < None >


Time Zone
☒ Originating Device
☐ Specific Time Zone Greenwich Standard Time

Save Delete Reset Add New

*- indicates required item.

273712

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



Cisco Unified CM Administration


For Cisco Unified Communications Solutions


Navigation Cisco Unified CM Administration


admin | About | Logout


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

Find and List Calling Search Spaces


 Add New

 Select All



 Clear All





 Delete Selected

Status

 5 records found

Calling Search Space (1 - 5 of 5) Rows per Page 50

Find Calling Search Space where CSS Name begins with Find Clear Filter  

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

Add New

Select All

Clear All

Delete Selected

273713

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Copy Add New

Status
Status: Ready

Calling Search Space Information
Name* CSS-Br1_Phones_Analog
Description CSS-Br1_Phones_Analog

Route Partitions for this Calling Search Space
Available Partitions**

Selected Partitions


- Partition-Br1_Phones_Analog
- Partition-Br1_Phones_IP
- Partition-HQ_Phones_Analog
- Partition-HQ_Phones_IP

Save Delete Copy Add New

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

273714

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



Cisco Unified CM Administration

For Cisco Unified Communications Solutions




Navigation Cisco Unified CM Administration

admin | About | Logout


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

Calling Search Space Configuration

Related Links: Back To Find/List

Save  Delete  Copy  Add New

Status

 Status: Ready

Calling Search Space Information

Name*

CSS-Br1_Phones_IP

Description

CSS-Br1_Phones_IP

Route Partitions for this Calling Search Space

Available Partitions**

Selected Partitions


Partition-Br1_Phones_IP


Partition-Br1_Phones_Analog

Partition-HQ_Phones_Analog

Partition-HQ_Phones_IP

Save Delete Copy Add New

 *- indicates required item.

 **Selected Partitions are ordered by highest priority

273715

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

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title 'Cisco Unified CM Administration', and a navigation dropdown menu. Below the navigation bar is a breadcrumb trail: System > Call Routing > Media Resources > Voice Mail > Device > Application > User Management > Bulk Administration > Help. The main content area is titled 'Calling Search Space Configuration' and includes a 'Related Links' section with a 'Back To Find/List' link. Below this is a toolbar with 'Save', 'Delete', 'Copy', and 'Add New' buttons. The configuration details are as follows:


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

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

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

273716

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



Cisco Unified CM Administration

For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

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

Calling Search Space Configuration

Related Links: Back To Find/List

Save Delete Copy Add New

Status

Status: Ready

Calling Search Space Information

Name*

CSS-HQ_Phones_IP

Description

CSS-HQ_Phones_IP

Route Partitions for this Calling Search Space

Available Partitions**

Selected Partitions

Partition-HQ_Phones_IP

Partition-Br1_Phones_Analog

Partition-Br1_Phones_IP

Partition-HQ_Phones_Analog

Save Delete Copy Add New

*- indicates required item.

**Selected Partitions are ordered by highest priority

273717

Configuring the Cisco Unified CM Media Resources Parameters

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

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

Media Resources: Annunciator Parameters

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

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

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu set to "Cisco Unified CM Administration". Below this is a secondary navigation bar with links for "admin", "About", and "Logout". A main navigation menu contains various system components: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The "Media Resources" menu is expanded, showing "Annunciator Configuration".

The "Annunciator Configuration" page features a "Related Links" section with a "Back To Find/List" link. Below this are "Save" and "Reset" buttons. The configuration details are as follows:

- Status:** Status: Ready
- Device Information:**
 - Registration: Registered with Cisco Unified Communications Manager 40.40.97.2
 - IP Address: 10.40.97.2
 - Server*: 10.40.97.2
 - Name*: ANN_2
 - Description: ANN_2_Ent1-HQ-CUCM
 - Device Pool*: DevicePool_HQ_IP_Phones
 - Location*: Hub_HQ

At the bottom of the configuration section are "Save" and "Reset" buttons. A note at the bottom left states: "i *- indicates required item."

273734

Media Resources: Conference Bridge Parameters

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

Figure 55 Media Resources Conference Bridges Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation menu with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The "Media Resources" menu is expanded, showing "Find and List Conference Bridges". Below this, there are buttons for "Add New", "Select All", "Clear All", "Delete Selected", and "Reset Selected". A status bar indicates "3 records found". The main content area displays a table of conference bridges with columns for Name, Description, Device Pool, Status, IP Address, and Copy. The table lists three bridges: CFB_2 (Default device pool), CON001AA29DF631 (DevicePool_Br1_DSPfarm), and CON111222333 (DevicePool_HQ_DSPfarm). At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", "Delete Selected", and "Reset Selected".

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

273735

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Conference Bridge Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Conference Bridge Information

Conference Bridge : CON001AA29DF631 (CFB-Ent1-Br1)

Registration Registered with Cisco Unified Communications Manager 10.40.97.2

IP Address 10.40.103.1

IOS Conference Bridge Info

Conference Bridge Type* Cisco IOS Enhanced Conference Bridge

Conference Bridge Name* CON001AA29DF631

Description CFB-Ent1-Br1

Device Pool* DevicePool_Br1_DSPfarm

Common Device Configuration < None >

Location* Hub_Br1

Device Security Mode* Non Secure Conference Bridge

Save Delete Copy Reset Add New

*- indicates required item.

273736

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Conference Bridge Configuration Related Links: Back To Find/List

Save Delete Copy Reset Add New

Status

Status: Ready

Conference Bridge Information

Conference Bridge : CON111222333 (CFB-Ent1-HQ)

Registration Registered with Cisco Unified Communications Manager 10.40.97.2

IP Address 10.40.97.1

IOS Conference Bridge Info

Conference Bridge Type* Cisco IOS Enhanced Conference Bridge

Conference Bridge Name* CON111222333

Description CFB-Ent1-HQ

Device Pool* DevicePool_HQ_DSPfarm

Common Device Configuration < None >

Location* Hub_HQ

Device Security Mode* Non Secure Conference Bridge

Save Delete Copy Reset Add New

i *- indicates required item.

273737

Media Resources: Media Termination Point Parameters

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

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

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

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

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

273738

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Copy Reset Add New

Status

Status: Ready

Media Termination Point Information

Registration	Registered with Cisco Unified Communications Manager 10.40.97.2
IP Address	10.40.103.1
Media Termination Point Type*	Cisco IOS Enhanced Software Media Termination Point
Media Termination Point Name*	MTP001AA29DF631
Description	MTP-Ent1-Br1
Device Pool*	DevicePool_Br1_DSPfarm

Save Delete Copy Reset Add New

*- indicates required item.

273739

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Copy Reset Add New

Status

Status: Ready

Media Termination Point Information

Registration Registered with Cisco Unified Communications Manager 10.40.97.2

IP Address 10.40.97.1

Media Termination Point Type* Cisco IOS Enhanced Software Media Termination Point

Media Termination Point Name* MTP111222333

Description MTP-Ent1-HQ

Device Pool* DevicePool_HQ_DSPfarm

Save Delete Copy Reset Add New

*- indicates required item.

273740

Media Resources: Music on Hold Server Parameters

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Reset

Status

Status: Ready

Device Information

Registration Registered with Cisco Unified Communications Manager 10.40.97.2

IP Address 10.40.97.2

Host Server* 10.40.97.2

Music On Hold Server Name* MOH-Ent1

Description MOH_Ent1-HQ

Device Pool* Default

Location* Hub_HQ

Maximum Half Duplex Streams* 250

Maximum Multicast Connections* 30

Fixed Audio Source Device

Run Flag* Yes

Multicast Audio Source Information

☐ Enable Multicast Audio Sources on this MOH Server

Base Multicast IP Address* 0.0.0.0

Base Multicast Port Number* 0 (Even numbers only)

Increment Multicast on* ☒ Port Number ☐ IP Address

Selected Multicast Audio Sources

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

Save Reset

***** indicates required item.

273741

Media Resources: Transcoder Parameters

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

Figure 62 Media Resources Transcoder Cisco Unified CM Administration Window

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below the navigation bar is a menu bar with options: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The "Media Resources" menu is expanded, showing "Find and List Transcoders".

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

The main content area displays a table of transcoders. The table has columns for Name, Description, Device Pool, Status, IP Address, and Copy. There are two records listed:

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

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

273742

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

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a "Navigation" dropdown menu. Below this is a secondary navigation bar with tabs for System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Transcoder Configuration" and includes a "Related Links" section with a "Back To Find/List" button. A toolbar at the top of the configuration area contains icons for Save, Delete, Copy, Reset, and Add New. The configuration is divided into two sections: "Transcoder Information" and "IOS Transcoder Info".

Transcoder Information

Transcoder: XCD001AA29DF631 (XCODE-Ent1-Br1)
Registration: Registered with Cisco Unified Communications Manager 10.40.97.2
IP Address: 10.40.103.1

IOS Transcoder Info

Transcoder Type* Cisco IOS Enhanced Media Termination Point
Description XCODE-Ent1-Br1
Device Name* XCD001AA29DF631
Device Pool* DevicePool_Br1_DSPfarm [View Details](#)
Common Device Configuration < None > [View Details](#)
Special Load Information Leave blank to use default

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

273743

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

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". A navigation dropdown menu is set to "Cisco Unified CM Administration". Below this, a secondary navigation bar lists various system components: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The "Media Resources" section is expanded, showing "Transcoder Configuration". A "Related Links" dropdown is set to "Back To Find/List".

Below the navigation bar, there is a toolbar with icons for Save, Delete, Copy, Reset, and Add New. The main content area is titled "Transcoder Configuration" and contains the following information:

- Transcoder Information**
 - Transcoder:** XCODE111222333 (XCODE-Ent1-HQ)
 - Registration:** Registered with Cisco Unified Communications Manager 10.40.97.2
 - IP Address:** 10.40.97.1
- IOS Transcoder Info**
 - Transcoder Type*:** Cisco IOS Enhanced Media Termination Point
 - Description:** XCODE-Ent1-HQ
 - Device Name*:** XCODE111222333
 - Device Pool*:** DevicePool_HQ_DSPfarm [View Details](#)
 - Common Device Configuration:** < None > [View Details](#)
 - Special Load Information:** Leave blank to use default


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

273744

Media Resources: Media Resource Group Parameters

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

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


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


Navigation Cisco Unified CM Administration


admin | About | Logout


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

Find and List Media Resource Groups


 Add New

 Select All



 Clear All



 Delete Selected

Status

 2 records found

Media Resource Group (1 - 2 of 2) Rows per Page 50

Find Media Resource Group where Name ▾ begins with ▾ Find Clear Filter  

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

Add New

Select All

Clear All

Delete Selected

273745

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Copy Reset Add New

Status
Status: Ready

Media Resource Group Status
Media Resource Group: Br1_HW_MRG (used by 11 devices)

Media Resource Group Information
Name* Br1_HW_MRG
Description Ent 1 Br1

Devices for this Group
Available Media Resources**
ANN_2
CFB_2
CON111222333
MTP111222333
MTP_2

Selected Media Resources*
CON001AA29DF631 (CFB)
MOH-Ent1 (MOH)
MTP001AA29DF631 (MTP)
XCD001AA29DF631 (XCODE)

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

Save Delete Copy Reset Add New

*- indicates required item.

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

273746

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Copy Reset Add New

Status
Status: Ready

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

Media Resource Group Information
Name* HQ_HW_MRG
Description Ent 1 HQ

Devices for this Group
Available Media Resources**
ANN_2
CFB_2
CON001AA29DF631
MTP001AA29DF631
MTP_2

Selected Media Resources*
CON111222333 (CFB)
MOH-Ent1 (MOH)
MTP111222333 (MTP)
XCODE111222333 (XCODE)

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

Save Delete Copy Reset Add New

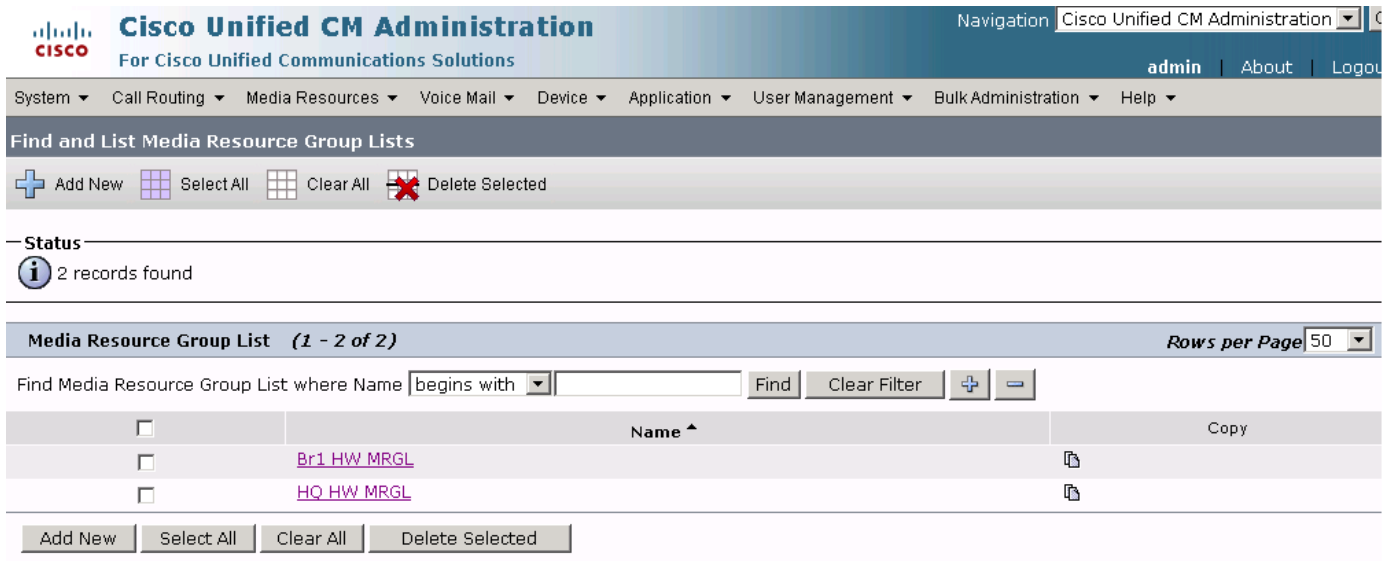
*- indicates required item.
**Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)

273747

Media Resources: Media Resource Group List Parameters

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

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



The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below the navigation bar is a menu bar with options: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Find and List Media Resource Group Lists". It features a toolbar with buttons for "Add New", "Select All", "Clear All", and "Delete Selected". Below the toolbar, a status bar indicates "2 records found". The main table displays the "Media Resource Group List" with a filter set to "begins with". The table has two columns: "Name" and "Copy". The first row is "Br1 HW MRGL" and the second row is "HQ HW MRGL". At the bottom of the table, there are buttons for "Add New", "Select All", "Clear All", and "Delete Selected".

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

273748

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

The screenshot displays the Cisco Unified CM Administration interface for configuring a Media Resource Group List. The page title is "Media Resource Group List Configuration". The navigation bar includes "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Media Resources" menu is expanded, showing "Media Resource Group List Configuration". The "Related Links" section contains a "Back To Find/List" link. The "Status" section shows "Status: Ready". The "Media Resource Group List Status" section shows "Media Resource Group List: Br1 HW MRGL (used by 11 devices)". The "Media Resource Group List Information" section shows "Name* Br1 HW MRGL". The "Media Resource Groups for this List" section shows "Available Media Resource Groups" with "HQ_HW_MRG" and "Selected Media Resource Groups" with "Br1_HW_MRG". The "Save", "Delete", "Copy", "Reset", and "Add New" buttons are visible at the bottom. A note at the bottom left states: "* - indicates required item."

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Copy Reset Add New

Status
Status: Ready

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

Media Resource Group List Information
Name* Br1 HW MRGL

Media Resource Groups for this List
Available Media Resource Groups HQ_HW_MRG

Selected Media Resource Groups Br1_HW_MRG

Save Delete Copy Reset Add New

* - indicates required item.

273749

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

The screenshot displays the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and the subtitle "For Cisco Unified Communications Solutions". The navigation menu shows "System", "Call Routing", "Media Resources", "Voice Mail", "Device", "Application", "User Management", "Bulk Administration", and "Help". The "Media Resources" menu is expanded, showing "Media Resource Group List Configuration". The "Related Links" section includes "Back To Find/List".

The main content area is titled "Media Resource Group List Configuration". It includes a toolbar with "Save", "Delete", "Copy", "Reset", and "Add New" buttons. The "Status" section shows "Status: Ready". The "Media Resource Group List Status" section shows "Media Resource Group List: HQ HW MRGL (used by 19 devices)". The "Media Resource Group List Information" section shows "Name* HQ HW MRGL". The "Media Resource Groups for this List" section shows "Available Media Resource Groups" with "Br1_HW_MRG" and "Selected Media Resource Groups" with "HQ_HW_MRG".

At the bottom, there is a toolbar with "Save", "Delete", "Copy", "Reset", and "Add New" buttons. A note indicates that "*" indicates a required item.

273750

Configuring the Cisco Unified CM Voice Mail Parameters

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

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

Voice Mail: Cisco Voice Mail Port Parameters

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Find and List Voice Mail Ports

+ Add New Select All Clear All Delete Selected Reset Selected

—Status—
5 records found

Voice Mail Port (1 - 5 of 5) Rows per Page: 50

Find Voice Mail Port where: Device Name begins with Find Clear Filter + -

Select item or enter search text

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

Add New Select All Clear All Delete Selected Reset Selected

273781

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Copy Reset Add New

Status
Status: Ready

Device Information

Registration: Registered with Cisco Unified Communications Manager 10.40.97.2
 IP Address: 10.40.97.253
 Port Name*: CiscoUM1-VI1
 Description: Voicemail for Enterprise1
 Device Pool*: DevicePool_HQ_IP_Phones
 Common Device Configuration: < None >
 Calling Search Space: CSS-HQ_Phones_IP
 AAR Calling Search Space: < None >
 Location*: Hub_HQ
 Device Security Mode*: Non Secure Voice Mail Port

Directory Number Information

Directory Number*: 1090
 Partition: Partition-HQ_Phones_IP
 Calling Search Space: CSS-HQ_Phones_IP
 AAR Group: < None >
 Internal Caller ID Display: VoiceMail
 Internal Caller ID Display (ASCII format): VoiceMail
 External Number Mask: 41555XXXX

Save Delete Copy Reset Add New


*- Indicates required item.

273782

Voice Mail: Message Waiting Parameters

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

Figure 73 Voice Mail Message Waiting Cisco Unified CM Administration Window


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


Navigation Cisco Unified CM Administration


admin | About | Logout


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

Find and List Message Waiting Numbers


 Add New

 Select All



 Clear All



 Delete Selected

Status

 2 records found

Message Waiting Numbers (1 - 2 of 2) Rows per Page 50

Find Message Waiting Numbers where Directory Number ▾ begins with ▾ and where Message Waiting Indicator is Both ▾ Find Clear Filter  

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

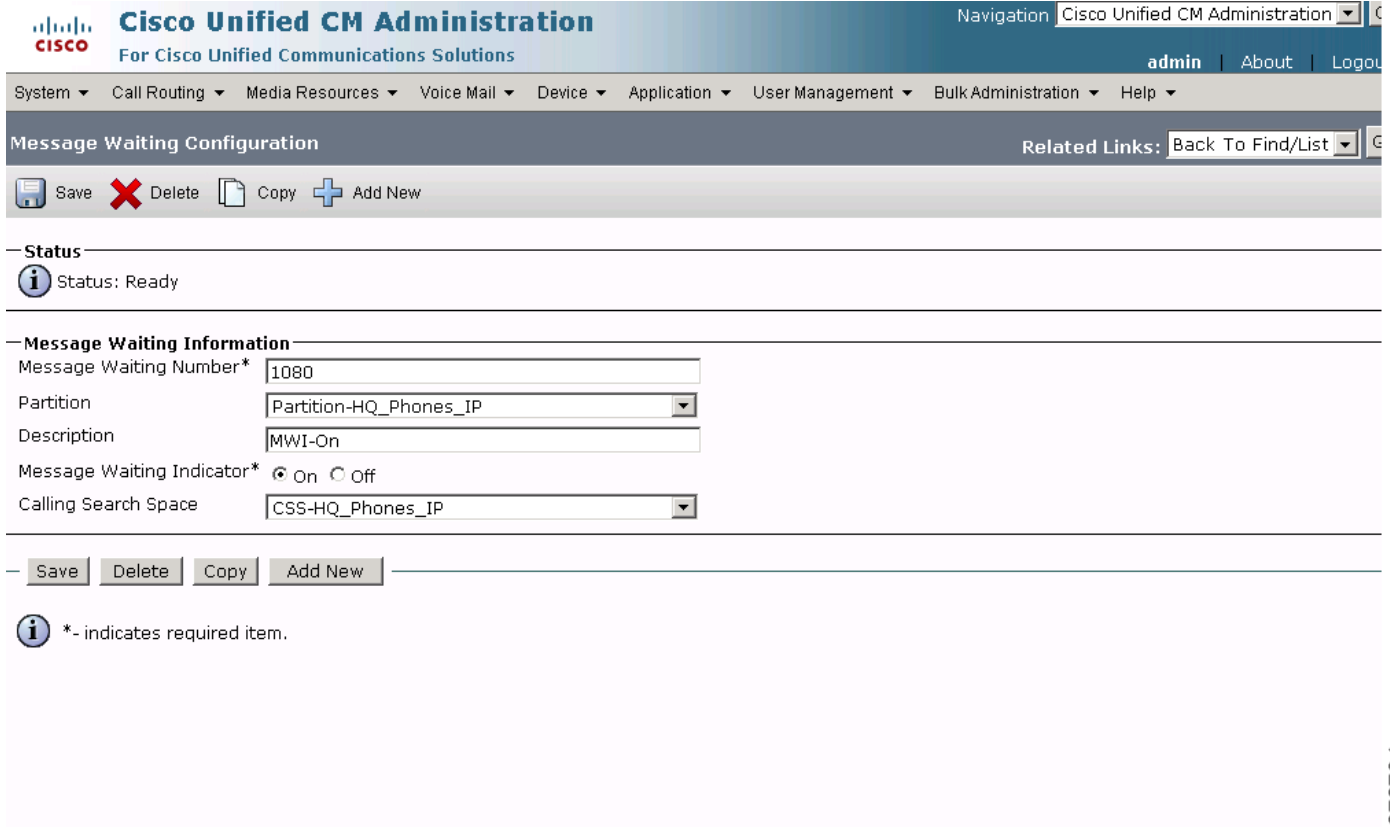
Add New

Select All

Clear All

Delete Selected

273783

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


Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Message Waiting Configuration Related Links: Back To Find/List

Save Delete Copy Add New

Status
Status: Ready

Message Waiting Information

Message Waiting Number* 1080

Partition Partition-HQ_Phones_IP

Description MWI-On

Message Waiting Indicator* ☒ On ☐ Off


Calling Search Space CSS-HQ_Phones_IP

Save Delete Copy Add New

*- indicates required item.

273784

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



Cisco Unified CM Administration

For Cisco Unified Communications Solutions

Navigation Cisco Unified CM Administration

admin | About | Logout

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

Message Waiting Configuration

Related Links: Back To Find/List

Save Delete Copy Add New

Status

Status: Ready

Message Waiting Information

Message Waiting Number*1081

PartitionPartition-HQ_Phones_IP

DescriptionMWI-Off

Message Waiting Indicator*☐ On ☒ Off

Calling Search SpaceCSS-HQ_Phones_IP

Save Delete Copy Add New

*- indicates required item.

273785

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

99

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

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration", and a navigation dropdown menu. Below the navigation bar is a menu bar with options: System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "Voice Mail Pilot Configuration" and includes a "Related Links" section with a "Back To Find/List" link. Below this is a section for "Status" showing "Status: Ready". The "Voice Mail Pilot Information" section contains the following fields:

- Voice Mail Pilot Number: 1099
- Calling Search Space: CSS-HQ_Phones_IP
- Description: Voicemail Pilot
- ☒ Make this the default Voice Mail Pilot for the system

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

273786

Voice Mail: Voice Mail Profile Parameters

To configure the voice mail voice mail profile parameters for the Cisco Unified CM, click **Voice Mail > Voice Mail Profile** 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

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes the Cisco logo, the title "Cisco Unified CM Administration For Cisco Unified Communications Solutions", and a navigation menu with options like System, Call Routing, Media Resources, Voice Mail, Device, Application, User Management, Bulk Administration, and Help. The "Voice Mail" menu is selected, and the "Voice Mail Profile Configuration" page is displayed. The page has a "Related Links" section with a "Back To Find/List" link. Below this is a "Status" section showing "Status: Ready". The main configuration section is titled "Voice Mail Profile Information" and contains the following fields:

- Voice Mail Profile: VM-Profile-Ent1-HQ (used by 15 devices)
- Voice Mail Profile Name*: VM-Profile-Ent1-HQ
- Description: Default voice messaging profile
- Voice Mail Pilot**: 1099/CSS-HQ_Phones_IP
- Voice Mail Box Mask:

There is a checkbox labeled "Make this the default Voice Mail Profile for the System" which is checked. At the bottom of the configuration section are buttons for "Save", "Delete", "Copy", "Reset", and "Add New". Below the configuration section are two informational messages:

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

273787

Configuring the Cisco Unified CM Device Parameters

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

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

Device: Gateway Parameters

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

Figure 78 Device Gateway Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Find and List Gateway

+ Add New Select All Clear All Delete Selected Reset Selected

— Status —
 2 records found

Gateways (1 - 2 of 2) Rows per Page: 50

Find Gateways where Name ▾ begins with ▾ Hide ▾ endpoints Find Clear Filter + -
Select item or enter search text ▾

<input type="checkbox"/>	Device Name ^	Description	Device Pool	Calling Search Space	Extension	Partition	Route Group	Priority	Port	Device Type	Status	IP Address
<input type="checkbox"/>	Ent1_Br1.Ent1.com	Ent1_Br1								Cisco 3845	See Endpoints	
<input type="checkbox"/>	SKIGWQC863972F5	Ent1-HQ-VG224								VG224	See Endpoints	

Add New Select All Clear All Delete Selected Reset Selected

273718

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

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

Save Delete Reset Add New

Status
Status: Ready

Gateway Details

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

Configured Slots, VICs and Endpoints

Module in Slot 0	< None >
Module in Slot 1	< None >
Module in Slot 2	< None >
Module in Slot 3	< None >
Module in Slot 4	NM-HDV2-2PORT-T1

Subunit 0	VIC2-2FXS	Begin Port	0	4/0/ 0	POTS
Subunit 1	< None >	Begin Port	0	4/0/ 1	POTS

Product Specific Configuration Layout

Global ISDN Switch Type	4ESS
Switchback Timing*	Graceful
Switchback uptime-delay (min)	10
Switchback schedule (hh:mm)	12:00
Type Of DTMF Relay*	Current GW Config

Save Delete Reset Add New

*- indicates required item.

273719

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Gateway Configuration Related Links: Back to MGCP Configuration

Save Delete Reset Add New

Status
Status: Ready

Directory Number Information

Line [1] - 1110 in Partition-

Br1 Phones Analog

Device Information

Product: Cisco MGCP FXS Port

Gateway: Ent1_Br1.Ent1.com

Device Protocol: Analog Access

Registration: Registered with Cisco Unified Communications Manager 10.40.97.2

IP Address: 10.40.103.1

End-Point Name *: AALN/S4/SU0/0@Ent1_Br1.Ent1.com

Description: Ent1_Br1_FXS

Device Pool*: DevicePool_Br1_Analog_Phones

Common Device Configuration: < None >

Media Resource Group List: Br1 HW MRGL

Calling Search Space: CSS-Br1_Phones_Analog

AAR Calling Search Space: < None >

Location*: Hub_Br1

AAR Group: < None >

Network Locale: < None >

☐ Transmit UTF-8 for Calling Party Name

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain: < None >

MLPP Indication: Not available on this device

MLPP Preemption: Not available on this device

Port Information (POTS)

Port Direction*: Bothways

Prefix DN:

Num Digits*: 4

Expected Digits*: 0

SMDI Port Number(0-4096)*: 0

☐ Unattended Port

Save Delete Reset Add New

*- indicates required item.

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

273720

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

Cisco Unified CM Administration Navigation: Cisco Unified CM Administration

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

Directory Number Configuration Related Links: Configure Device (AALN/54/SUO/0@Ent1_Br1.Ent1.com)

Save Delete Reset Add New

Status
Status: Ready

Directory Number Information

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

Dissociate Devices

Directory Number Settings

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

AAR Settings

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

☒ Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

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

MLPP Alternate Party Settings

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

Line Settings for All Devices

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

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

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

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

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

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

☒ Caller Name
☒ Caller Number
☒ Redirected Number
☒ Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Add New

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

273721

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Gateway Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Gateway Details

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

Configured Slots, VICs and Endpoints

Module in Slot 2: ANALOG

Subunit 0: 24FXS-SCCP

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

Save Delete Reset Add New

* - indicates required item.

273722

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Phone Configuration Related Links: [Back to Gateway](#)

Save Delete Copy Reset Add New

Status
 Status: Ready

Association Information

[Modify Button Items](#)

1 [Line \[1\] - 1050 in Partition-HQ_Phones_Analog](#)

----- Unassigned Associated Items -----

2 [Line \[2\] - Add a new DN](#)

Phone Type

Product Type: Analog Phone

Device Protocol: SCCP

Device Information

Registration: Registered with Cisco Unified Communications Manager 10.40.97.2

IP Address: 10.40.97.254

MAC Address*: 0C863972F5400

Description: 415555XXXX

Device Pool*: [DevicePool_HQ_Analog_Phones](#) [View Details](#)

Common Device Configuration: < None > [View Details](#)

Phone Button Template*: Standard Analog

Common Phone Profile*: Standard Common Phone Profile

Calling Search Space: CSS-HQ_Phones_Analog

Media Resource Group List: HQ HW MRGL

Location*: Hub_HQ

User Locale: < None >

Network Locale: < None >

Device Mobility Mode*: Default [View Current](#)
[Device Mobility Settings](#)

Owner User ID: < None >

☒ Is Active

☐ Ignore Presentation Indicators (internal calls only)

☐ Allow Control of Device from CTI

☐ Logged Into Hunt Group

☐ Remote Device

Protocol Specific Information

Presence Group*: Standard Presence group

Device Security Profile*: Analog Phone - Standard SCCP Non-Secure Pr

SUBSCRIBE Calling Search Space: < None >

☐ Unattended Port

MLPP Information

MLPP Domain: < None >

MLPP Indication*: Default

MLPP Preemption*: Default

Save Delete Copy Reset Add New

*- indicates required item.

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

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

273723

Enterprise 1 HQ Cisco Unified CM Example Configuration

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

Cisco Unified CM Administration For Cisco Unified Communications Solutions

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

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

Directory Number Configuration Related Links: Configure Device (ANOC863972F5400)

Save Delete Reset Add New

Status: Ready

Directory Number Information

Directory Number* 1050

Route Partition Partition-HQ_Phones_Analog

Description 1050

Alerting Name

ASCII Alerting Name

☒ Allow Control of Device from CTI

Associated Devices ANOC863972F5400

Edit Device Edit Line Appearance

Dissociate Devices

Directory Number Settings

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

Calling Search Space CSS-HQ_Phones_Analog

Presence Group* Standard Presence group

User Hold MOH Audio Source 1-SampleAudioSource

Network Hold MOH Audio Source 1-SampleAudioSource

AAR Settings

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

☒ Retain this destination in the call forwarding history

Call Forward and Call Pickup Settings

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

MLPP Alternate Party Settings

Target (Destination)

MLPP Calling Search Space < None >

MLPP No Answer Ring Duration (seconds)

Line Settings for All Devices

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

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

Line 1 on Device ANOC863972F5400

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

ASCII Display (Internal Caller ID)

External Phone Number Mask 415555XXXX

Monitoring Calling Search Space < None >

Multiple Call/Call Waiting Settings on Device ANOC863972F5400

Note: The range to select the Max Number of calls is: 1-2

Maximum Number of Calls* 1

Busy Trigger* 1 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device ANOC863972F5400

☒ Caller Name

☒ Caller Number

☒ Redirected Number

☒ Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Add New

* - indicates required item.

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

273724

Device: Phone Parameters

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

Figure 85 **Device Phone 4155551000 Cisco Unified CM Administration Window**

[illegible]

273725

Cisco Unified CM Administration
 For Cisco Unified Communications Solutions

Navigation: [Cisco Unified CM Administration](#) | [Admins](#) | [About](#) | [Logout](#)

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

Directory Number Configuration

Related Links: [Configure Device \(SEP001B7371C3FA\)](#)

Save
 Delete
 Reset
 Add New

Status:
 (1) Status: Ready

Directory Number Information

Directory Number* 1000
 Router Partition CSS-HQ_Phones_IP
 Description 1000
 Alerting Name
 ASCII Alerting Name
☐ Allow Control of Device from CTI
 Associated Devices SEP001B7371C3FA

Edit Device
 Edit Line Appearance

Dissociate Devices

Directory Number Settings

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

AAR Settings

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

☒ Retain this destination in the call forwarding history

Call Forwarding and Call Pickup Settings

	Yoice Mail	Destination	Calling Search Space
Calling Search Space Activation Policy			Use System Default
Forward All	<input type="checkbox"/>		<None>
Secondary Calling Search Space for Forward All			<None>
Forward Busy Internal	<input checked="" type="checkbox"/>		CSS-HQ_Phones_IP
Forward Busy External	<input checked="" type="checkbox"/>		CSS-HQ_Phones_IP
Forward No Answer Internal	<input checked="" type="checkbox"/>		CSS-HQ_Phones_IP
Forward No Answer External	<input checked="" type="checkbox"/>		CSS-HQ_Phones_IP
Forward No Coverage Internal	<input checked="" type="checkbox"/>		<None>
Forward No Coverage External	<input checked="" type="checkbox"/>		<None>
Forward on CTI Failure	<input checked="" type="checkbox"/>		<None>
Forward Unregistered Internal	<input checked="" type="checkbox"/>		CSS-HQ_Phones_IP
Forward Unregistered External	<input checked="" type="checkbox"/>		CSS-HQ_Phones_IP
No Answer Ring Duration (seconds)		0	
Call Pickup Group		<None>	

MLPP Alternate Party Settings

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

Line Settings for All Devices

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

Line 1 on Device SEP001B7371C3FA

Display (Internal Caller ID) Display text for a line appearance is intended for displaying text such as a name instead of a directory number for internal calls. If you specify a number, the person receiving a call may not see the proper identity of the caller.
 ASCII Display (Internal Caller ID)
 Line Text Label
 ASCII Line Text Label
 External Phone Number Mask #15551000X
 Visual Message Waiting Indicator Use System Policy
 Audible Message Waiting Indicator Policy Off
 Ring Setting (Phone SPS)* Use System Default
 Ring Setting (Phone Active) Use System Default Applies to this line when any line on the phone has a call in progress.
 Call Pickup Group Audio Alert Setting (Phone Idle) Use System Default
 Call Pickup Group Audio Alert Setting (Phone Active)
 Recording Option* Call Recording Disabled
 Recording Profile <None>
 Monitoring Calling Search Space <None>

Multiple Call/Call Waiting Settings on Device SEP001B7371C3FA

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

Forwarded Call Information Display on Device SEP001B7371C3FA

☒ Caller Name
☒ Caller Number
☒ Redirected Number
☒ Dialed Number

Users Associated with Line

Associate End Users

Save
 Delete
 Reset
 Add New

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

Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 87 Device Phone 4155551170 Cisco Unified CM Administration Window

Cisco Unified CM Administration

Phone Configuration: **Device Phone 4155551170**

Phone Type: **Cisco 7961**

Product Type: **Cisco 7961**

Device Protocol: **SIP**

Registration: **Registered with Cisco Unified Communications Manager 10.40.97.2**

IP Address: **10.40.97.2**

MAC Address: **9C:05:06:00:00:00**

Description: **4155551170**

Device Pool: **DefaultPool**

Common Device Configuration: **Standard T81 SCCP**

Template: **Standard User**

Software Template: **Standard Common Phone Profile**

Common Phone Profile: **Standard Common Phone Profile**

Call Forward Search: **CC-BK-Phones_IP**

AMR Calling Search: **None**

Search: **None**

Media Resource Group: **None**

User Hold MCH Audio: **None**

Network Hold MCH Audio: **None**

Location: **None**

AMR Group: **None**

User Locale: **None**

Network Locale: **None**

Built In Bridge: **Default**

Privacy: **Default**

Device Mobility Mode: **Default**

Owner User ID: **None**

Phone Personalization: **Default**

Phone Label Name: **None**

Single Button Merge: **Default**

RF Is Active: **Default**

Join Across Lines: **Default**

Party Video Call as Audio: **None**

Ignore Presentation Indicators (Internal calls only): **None**

Allow Control of Device from CTI: **None**

Logged Into Hunt Group: **None**

Remote Device: **None**

Packet Capture Mode: **None**

Packet Capture Duration: **0**

Presence Group: **Standard Presence group**

Device Security Profile: **CCM T81 - Standard SCCP Non-Secure Mode**

Unsecured Calling Search Space: **None**

Unsecured Port: **None**

Require CTI Reception: **None**

Require CTI Reception: **None**

Certificate Authority Proxy Function (CAPF) Information

Certificate Operation: **No Pending Operation**

Authentication Mode: **Not Strong**

Authentication String: **None**

Key Size (bits): **1024**

Operation Completes By: **2011-01-01 00:00:00**

Certificate Operation Status: **None**

Note: Security Profile Contains Additional CAPF Settings

Exemption Module Information

Module 1 Load Name: **None**

Module 2 Load Name: **None**

External Data Locations Information (Leave blank to use default)

Information: **None**

Directions: **None**

Messages: **None**

Services: **None**

Authentication Server: **None**

Press Server: **None**

Site: **None**

Site Timer (seconds): **None**

Extension Information

Enable Extension Mobility: **None**

Log Out Profile: **Use Current Device Settings**

Log In Time: **None**

Log Out Time: **None**

MLPP Information

MLPP Domain: **None**

MLPP Indication: **Default**

MLPP Preemption: **Default**

Do Not Disturb

Do Not Disturb: **None**

Out of Office: **None**

Out of Office Call Alert: **None**

Secure Shell Information

Secure Shell User: **None**

Secure Shell Password: **None**

Product Specific Configuration Legend

Enable Speakerphone: **None**

Enable Speakerphone and Headset: **None**

Forwarding Delay: **None**

PC Port: **None**

Settings Access: **None**

Gratuitous ARP: **None**

PC Voice VLAN Access: **None**

Video Capability: **None**

Auto Line Select: **None**

Web Access: **None**

Open to PC Port: **None**

Logging Display: **None**

Load Server: **None**

Recording Tone: **None**

Recording Tone Local Volume: **None**

Recording Tone Remote Volume: **None**

Recording Tone Duration: **None**

MLPP: **None**

MLPP Soft Key Tone: **None**

Auto Call Select: **None**

Log Server: **None**

Adaptive T12 Codec: **None**

Wideband handset UC Control: **None**

Wideband handset UC Control: **None**

Wideband handset: **None**

Wideband handset: **None**

Peer Firmware Sharing: **None**

Cisco Discovery Protocol (CDP): Switch Port: **None**

Cisco Discovery Protocol (CDP): PC Port: **None**

Link Layer Discovery Protocol (LLDP): Media: **None**

Link Layer Discovery Protocol (LLDP): Switch Port: **None**

LLDP Power ID: **None**

LLDP Power Priority: **None**

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* Indicates required item.

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

*** Note: Security Profile Contains Additional CAPF Settings.

Figure 88 **Device Phone 1170 Cisco Unified CM Administration Window**

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

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

Directory Number Configuration Related Links: Configure Device (SEP0019E8ABE7B) > G

Save Delete Reset Add New

Status
 Status: Ready

Directory Number Information

Directory Number* 1170
 Route Partition Partition-Br1_Phones_IP
 Description 1170
 Alerting Name
 ASCII Alerting Name
☒ Allow Control of Device from CTI
 Associated Devices SEP0019E8ABE7B **Edit Device**
 Edit Line Appearance

Dissociate Devices

Directory Number Settings

Voice Mail Profile <None> (Choose <None> to use system default)
 Calling Search Space CSS-Br1_Phones_IP
 Presence Group* Standard Presence group
 User Hold MOH Audio Source 1-SampleAudioSource
 Network Hold MOH Audio Source 1-SampleAudioSource
 Auto Answer* Auto Answer with Speakerphone

AAR Settings

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

Call Forward and Call Pickup Settings

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

MLPP Alternate Party Settings

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

Line Settings for All Devices

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

Line 1 on Device SEP0019E8ABE7B

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

ASCII Display (Internal Caller ID)
 Line Text Label
 ASCII Line Text Label
 External Phone Number Mask #155555XXXX
 Visual Message Waiting Indicator Policy* Use System Policy
 Audible Message Waiting Indicator Policy* Off
 Ring Setting (Phone Idle)* Use System Default
 Ring Setting (Phone Active) Use System Default Applies to this line when any line on the phone has a call in progress.
 Call Pickup Group Use System Default
 Audio Alert Setting (Phone Idle) Use System Default
 Call Pickup Group Audio Alert Setting (Phone Active) Use System Default
 Recording Option* Call Recording Disabled
 Recording Profile <None>
 Monitoring Calling Search Space <None>

Multiple Call/Call Waiting Settings on Device SEP0019E8ABE7B

Note: The range to select the Max Number of calls is: 1-200
 Maximum Number of Calls* 6
 Busy Trigger* 2 (Less than or equal to Max. Calls)

Forwarded Call Information Display on Device SEP0019E8ABE7B

☒ Caller Name
☒ Caller Number
☒ Redirected Number
☒ Dialed Number

Users Associated with Line

Associate End Users

Save Delete Reset Add New

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

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Device: Trunk Parameters

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

Figure 89 Device Trunk Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Find and List Trunks

+ Add New Select All Clear All Delete Selected Reset Selected

— Status —
4 records found

Trunks (1 - 4 of 4) Rows per Page 50

Find Trunks where Device Name begins with Find Clear Filter + -
Select item or enter search text

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

+ Add New Select All Clear All Delete Selected Reset Selected

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Device Information

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

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

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

Call Routing Information

Inbound Calls

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

Outbound Calls

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

SIP Information

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

Save Delete Reset Add New

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Enterprise 1 HQ Cisco Unified CM Example Configuration

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin About Logout

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

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Device Information

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

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

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

Call Routing Information

Inbound Calls

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

☐ Redirecting Diversion Header Delivery - Inbound

Outbound Calls

Calling Party Selection*: Last Redirect Number (External)
 Calling Line ID Presentation*: Default
 Calling Name Presentation*: Default
 Caller ID DN:
 Caller Name:

☐ Redirecting Diversion Header Delivery - Outbound

SIP Information

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

Save Delete Reset Add New

*- indicates required item.

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

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin | About | Logout

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

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Device Information

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

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

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

Call Routing Information

Inbound Calls

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

Outbound Calls

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

SIP Information

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

Save Delete Reset Add New

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

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Enterprise 1 HQ Cisco Unified CM Example Configuration

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

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration

admin About Logout

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

Trunk Configuration Related Links: Back To Find/List

Save Delete Reset Add New

Status
Status: Ready

Device Information

Product: SIP Trunk
Device Protocol: SIP
Device Name*: 10.80.80.82
Description: Ent1-Br1-CUBE1
Device Pool*: DevicePool_WAN
Common Device Configuration: < None >
Call Classification*: Use System Default
Media Resource Group List: Br1 HW MRGL
Location*: Trunk Br1
AAR Group: < None >
Packet Capture Mode*: None
Packet Capture Duration: 0
☐ Media Termination Point Required
☒ Retry Video Call as Audio
☐ Transmit UTF-8 for Calling Party Name
☐ Unattended Port

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

Call Routing Information

Inbound Calls

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

Outbound Calls

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

SIP Information

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

Save Delete Reset Add New

*- indicates required item.

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

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Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration

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

Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration

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

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

```

port 2/1
!
Ent1_HQ_VG224#

```

Enterprise 1 HQ Cisco ASA Firewall Example Configuration

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

```

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

```



```

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

```

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

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

```

Ent1_Br1#

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

```

```

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

```

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

```

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

```

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

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

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

Cisco Validated Design

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