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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 headquarters with its enterprise branch offices. At the enterprise headquarters, Cisco Unified CM provides call control for voice services. Remote enterprise branch offices have Cisco Unified SRST deployed for voice services. The Cisco Integrated Services Router (Cisco ISR) running the Cisco Unified Border Element (Cisco UBE) is placed at the edge of the network. Cisco UBE plays an important role in serving multiple functions when connecting to other networks.

This design guide discusses the components deployed in the network, and provides sample router configurations for the Cisco UBE functions tested for the features included in this document. This guide is an update to the existing SRND and validates the Cisco UBE functions on the second generation Cisco Integrated Services Router (Cisco ISR-G2) 29xx/39xx and 3945E platforms, All other solution components remain unchanged.

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

Network Topology

The components of the enterprise SIP trunk deployment with Cisco Unified CM and Cisco Unified SRST network topology is 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 31
- Enterprise 1 HQ Cisco Unified CM Example Configuration, page 34
- Enterprise 1 HQ Cisco ASA Firewall Example Configuration, page 123
- Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration, page 122
- Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration, page 122

Enterprise Branch

- Branch 1 Cisco UBE, TDM Gateway, and Cisco Unified SRST Example Configuration, page 124
- Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration, page 128
Service Provider
- PSTN hop-off gateway
- SIP Call Agent
- Multiprotocol Label Switching (MPLS) core network

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

Prerequisites are grouped into the following sections:

- Components Used, page 5
- Cisco IOS Software Releases, page 6
- Conventions, page 6

Components Used

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

- Cisco Unified Communications Manager, page 5
- Cisco Unified Border Element, page 5
- SCCP Analog Voice Gateway, page 5
- Voice Mail at the Enterprise Headquarter Site, page 5
- Cisco Adaptive Security Appliance Firewall Appliance, page 5
- Cisco Survivable Remote Site Telephony, page 6

Cisco Unified Communications Manager

The Cisco Unified CM at the enterprise headquarter site provides call control to voice services at the headquarter site and the branch offices. The Cisco Unified CM was tested using version 6.1.3.

Cisco Unified Border Element

A Cisco 3945 and 3945A series platforms were tested with Cisco IOS Release 15.1(1)T and Cisco UBE version 1.4. The Cisco 2900 series Integrated Services Router (Cisco ISR) can also be used as a Cisco UBE.

SCCP Analog Voice Gateway

A Cisco VG224 analog voice gateway was used at the enterprise headquarter site to provide connectivity to analog phones and fax machines. The Cisco VG224 analog voice gateway was tested with Cisco IOS Release 15.1(1)T.

Voice Mail at the Enterprise Headquarter Site

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

Cisco Adaptive Security Appliance Firewall Appliance

A Cisco ASA firewall appliance was placed at the ingress from the service provider servicing the enterprise headquarter site. It was tested with Cisco ASA 8.0(4).
The Cisco UBE at the enterprise headquarter site can also be used to provide Cisco IOS firewall functions. If the Cisco UBE is used to provide Cisco IOS zone-based firewall functions, the Cisco ASA firewall appliance is not needed.

**Cisco Survivable Remote Site Telephony**

A Cisco Unified SRST router is placed at the enterprise branch site. In addition to the Cisco Unified SRST functions, this router provides Cisco UBE, Cisco IOS firewall, conferencing transcoding, MTP, voice mail using Cisco Unity Express, TDM, and gateway functions. A Cisco 3800 series platform was tested with Cisco IOS Release 15.1(1)T. Cisco Unity Express was tested with version 3.2. The Cisco 2800 series Integrated Services Router (Cisco ISR) can also be used as an Cisco Unified SRST router.

**SIP Call Agent**

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

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

**PSTN Hop-Off Gateway**

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

**Cisco IOS Software Releases**

The test results described in this guide for the Cisco Unified Border Element were conducted using Cisco IOS Release 15.1(1)T. We recommend Cisco IOS Release 15.1(1)T or later releases for the deployment of the features described in this guide.

**Conventions**

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

**Solution Description**

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

When Cisco Unified CM fails, but the WAN connection remains active and SRST takes over, the remote phones are able to make WAN calls through SIP to the call agent. If a WAN connectivity failure occurs, the enterprise branch offices can continue to maintain basic IP phone and PSTN services.
The focus of services using this solution are:
- Voice services with call control provided by Cisco Unified CM at the enterprise headquarter site
- Voice services with Cisco Unified SRST at the enterprise branch offices

The following topics describe the solution:
- Feature Summary, page 7
- IP Connectivity, page 16
- Quality of Service, page 17
- Voice Mail, page 19
- Dial Plan, page 19
- Security, page 19
- Failover and Redundancy, page 20
- Fax and Modem, page 20
- Billing and Management, page 20
- Best Practices for SIP Trunk Implementation Using Cisco UBE, page 21
- Caveats, page 22

**Feature Summary**

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

**Enterprise Headquarter Site Features**
- Cisco Unified Communications Manager call control
- Cisco Unified Border Element
- Cisco ASA Firewall or Cisco IOS Zone-Based Firewall
- Cisco Unity Voice Mail Server
- Analog Phone and Fax Services

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

**Service Provider Features**
- Multiprotocol Label Switching (MPLS) in the service provider backbone network
- PSTN Hop-Off Services (using service provider shared PSTN gateway)
- Optional Voice Mail Server
Basic Phone Features Served in the Topology

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

SIP Trunking Design Considerations

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

- DTMF Transport, page 8
- SIP Delayed Offer and Early Offer, page 9
- Early Media Cut Through, page 9
- SIP Trunk Transport Protocols, page 10
- Monitoring SIP Trunk State, page 10

DTMF Transport

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

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

OOB SIP DTMF signaling methods include:

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

KPML (RFC 4730) is the preferred OOB signaling method used by Cisco. KPML is supported on Advanced Cisco 79X1 Series IP Phones, Cisco Unified CM, and Cisco IOS Gateways (Cisco IOS Release 15.1 and later).

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

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

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

**SIP Delayed Offer and Early Offer**

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

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

**Note**

Service providers sometimes mandate an Early Offer call from the enterprise. In such cases Cisco UBE (Cisco IOS Release 15.1(1) and later) can be configured to convert the Delayed Offer to the Early Offer.

**Early Media Cut Through**

The terms Early Offer and Early Media are often confused.

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

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

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

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

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

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

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

Monitoring SIP Trunk State

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

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

- OPTIONS Method—The SIP OPTIONS method allows a UA to query another UA or a proxy server as to determine its capabilities. This query allows a client to discover information about the supported methods, content types, extensions, codecs, and so on, without actually placing a call. Cisco UBE sends an Out of Dialogue OPTIONS message to the device at the far-end of the SIP trunk to determine its state. The OPTIONS method is used as an application-level ping. The returned ping response is generally not as important as the fact that the trunk has confirmed that it is alive. Cisco Unified CM SIP trunks support the receipt of OPTIONS messages but do not send OPTIONS messages as keepalives. Cisco Unified CM version 5.x SIP trunks respond to OPTIONS messages with a “405—Method Not Acceptable” response. In Cisco Unified CM version 6.0.1, SIP trunks respond to an OPTIONS message with a “200—OK” response.

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

SIP Trunk Redundancy and Load Balancing

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

- Dial-peer preferences and Dial-peer Hunting
- DNS SRV
- Route List & Route Group option from CCM

Dial-peer preferences and Dial-peer Hunting

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

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

Redundancy and Scaling in SIP networks

1. Max Number of calls can be defined on the outgoing VoIP dialpeer
2. If number of calls exceed, server error 500 is sent back
3. SIP Proxy chooses the next IP Address provided by the DNS SRV record
4. Call is now sent to the next IP Address in the DNS SRV
Route List & Route Group option from CCM

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

1. Configure one Route Group to each IPIPgw (see Figure 3).

Figure 3 Configuring Route Groups
2. Configure one Route List to club all Route Groups (see Figure 4).

**Figure 4 Configuring A Route List for Route Groups**
3. Configure Route List under Route Pattern Gateway or Route List (see Figure 5).

**Figure 5** Configuring A Route List Under Route Pattern Gateway or Route List
4. Configure Max-Con under IPIPgw dial-peers towards Meeting Place, or Set the Global Call Treatment for total-calls (see Figure 6).

**Figure 6 Configuring Max-Con**

**IP Connectivity**

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

The sample configuration in the “Configurations” section on page 22 shows a Gigabit Ethernet interface. Some service providers that offer both data and voice services over a single IP interface also offer MPLS services. With MPLS services, voice packets must be sent with an MPLS label so that the service provider can terminate the traffic, and data marked with a different label can be tunneled through the backbone network. Marking voice traffic with an MPLS label requires the Virtual Routing and Forwarding (VRF)-Aware voice feature available on the Cisco ISRs in Cisco IOS Release 15.1(1)T.
Quality of Service

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

- Congestion Management, page 17
- Packet Marking, page 18
- Call Admission Control, page 18
- Delay, page 18
- Echo, page 19

Congestion Management

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

VoIP signaling and media traffic can be identified and classified as priority traffic using the QoS tools available within Cisco IOS software. Use Low Latency Queuing (LLQ) for media traffic streams. During congestion, LLQ queues restrict throughput to the configured bandwidth and packets exceeding this bandwidth are dropped. Therefore, signaling traffic should use class-based weighted fair queuing (CBWFQ), because signaling traffic bursts during call setup and teardown. The configurations for LLQ and CBWFQ are shown in the “Configurations” section on page 22. See Quality of Service for Voice Over IP for more information.

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

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

The service provider can limit the maximum number of calls allowed across the SIP trunk based on the CAC techniques discussed in the “Billing and Management” section on page 20. This maximum number of calls allowed can be part of the service level agreement (SLA) between the service provider and the end customer.

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

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

QoS policies may use either DSCP or IP precedence to classify voice packets. IP precedence interprets the low order three bits of the 6-bit DSCP value. In this way DSCP EF maps to CS5, while DSCP AF31 maps to CS3, which are appropriate IP precedence settings for voice media and signaling traffic.

Call Admission Control

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

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

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

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

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

Note

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

Delay

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

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

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

Voice Mail

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

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

Dial Plan

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

Security

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

- **Authentication**, page 19
- **Encryption of Media and Signaling**, page 19
- **Firewall**, page 20

**Authentication**

SIP registration and call method authentication can be provided using Digest Authentication. This method uses a single username and password for the entire SIP trunk, as shown in the “Configurations” section on page 22. The password is encrypted using Message Digest 5 (MD5).

**Encryption of Media and Signaling**

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

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

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

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

Failover and Redundancy

If a complete SIP trunk failure or IP interface failure occurs, backup PSTN lines connected directly to Cisco Unified SRST can be used for PSTN access. In the Cisco Unified SRST router configuration shown in the “Configurations” section on page 22, backup PSTN access was tested for alternate call routing when SIP trunk access was down.

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

Fax and Modem

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

Billing and Management

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

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

- CDR Logging with Syslog Servers and Cisco IOS Gateways
- Equivalent MIB Objects for VoIP show Commands
- RADIUS VSA Voice Implementation Guide
Best Practices for SIP Trunk Implementation Using Cisco UBE

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

- Configure explicit incoming and outgoing dial-peers for Cisco UBE to apply the appropriate treatment to calls (for example, translations, codec, DTMF-type, SIP Normalization, and so on).
- Configure VoIP dial-peers with appropriate descriptions. For example:
  - description *** dial-peer to Service Provider ***
  - description *** dial-peer to Publisher Cisco Unified CM ***
  - description *** dial-peer to Subscriber Cisco Unified CM ***
- Configure valid descriptions for explicit incoming and outgoing VoIP dial-peers to and from the Service Provider/Enterprise to ease troubleshooting. For example:
  - dial-peer voice 100 voip
    description *** incoming calls from Service Provider ***
    incoming called number xxx
  - dial-peer voice 200 voip
    description *** outgoing calls to Enterprise ***
    destination-pattern xxx
- Use a keepalive mechanism, such as Out of Dialog OPTIONS-ping, over the SIP trunk to detect upstream entity failure before routing calls to the service provider.
- Configure the Cisco UBE for media inactivity based on RTP, RTCP, or both to accelerate the detection of hung calls.
- Use RFC 2833 to configure DTMF because it is the most widely deployed and most interoperable DTMF mechanism for SIP trunks.
- Enable PRACK on Cisco Unified CM if Cisco UBE is configured to do Delayed Offer to Early Offer conversions for call flows involving early media cut through (18x w/SDP).
- If using G.729 over WAN, make sure the following CLI command is configured for RFC 3555 backward compliance: sip-ua g729-annexb override.
- Tune the failover timers when using clustered/DNS-SRV addressing.
  To ensure minimum Post Dial Delay during failover situations, fine tune the sip-ua retry xxx parameters, where xxx is the request name and response code. We recommend the value for INVITES as retry invite 2.
- Do not configure Cisco HSRP on the router that runs Cisco UBE functionality.
  The Layer 3 and Layer 7 embedded SIP addresses can be unpredictable when Cisco HSRP is enabled. Refer to the caveats section for exact Bug-ID.
- Use SIP profiles to insert or remove elements in the SIP headers.
  SIP Profiles is a very powerful SIP message normalization and protocol repair tool that can quickly fix or create a workaround to minor interoperability issues when two SIP implementations communicate with each other. This feature is available in Cisco IOS 12.4(15)XZ and Cisco IOS 12.4(20)T and later.
• Use the Cisco Unified SIP Proxy and Cisco UBE scaling architecture at the HQ location, if SIP trunk capacity requires a stack of Cisco UBEs to scale capacity.

• Pay attention to DTMF interoperability and call flows.

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

• Adjust SIP incoming and outgoing ports as required to accommodate send and listen devices on non-standard SIP ports.

• Test call flows with supplementary services since they may present interoperability issues.

• Configure ACLs on Cisco UBE to allow traffic only from valid call agents and endpoints to avoid toll-fraud.

  You can configure CLI commands such as allow term.

• Configure fax traffic on TDM PSTN access if at all possible.

• Mark all the outbound voice traffic with the appropriate DSCP values so that it gets the right priority in the service provider network. All other traffic should be appropriately marked.

• Provision backup FXO trunks on the Cisco CPE router to provide emergency PSTN access if the SIP trunk is down.

• Routing for emergency (911) calls using the shared hop-off PSTN gateway should be ensured by the service providers.

Caveats

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

• Voice calls must use the same static codec. It can be any codec type, but the same codec must be maintained.

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

• Voice calls over the WAN must be configured with G.729 codecs.

• Video was not tested as part of this solution.

• H.323 calls were not tested as part of this solution.

• Use of Cisco HSRP is not recommended in this solution as it can cause unexpected results with SIP signaling.

Configurations

The “Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations” section on page 25 provides configuration examples, screen figures, and other helpful information you need to configure the features on the Cisco UBE router at the edge of the service provider network described in this guide.

Note

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:
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:


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.

- Cisco Unified Communications Manager Express 4.1 Multi-party Conferencing Enhancements
- CDR Logging with Syslog Servers and Cisco IOS Gateways
- Cisco 2800 Series Integrated Services Routers
- Cisco 2900 Series Integrated Services Routers (Cisco 2900 ISR-G2)
- Cisco 3800 Series Integrated Services Routers
- Cisco 3900 Series Integrated Services Routers (Cisco 3900 ISR-G2)
- Cisco Cable High-Speed WAN Interface Cards
- Cisco High Density Analog and Digital Extension Module for Voice and Fax
- Cisco IAD243X Business Class Integrated Access Device
- Cisco Systems - Support
- “Configuring Conferencing” chapter of Cisco Unified Communications Manager Express System Administrator Guide
- Configuring Frame Relay and Frame Relay Traffic Shaping
- Configuring SIP Support for Hookflash
- Echo Analysis for Voice over IP
- Enterprise QoS Solution Reference Network Design Guide
- Equivalent MIB Objects for VoIP show Commands
- IP Communications Voice/Fax Network Module
- Quality of Service for Voice Over IP
- RADIUS VSA Voice Implementation Guide
- Service Provider Quality-of-Service Overview
- Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)
Overview of Test Configurations

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

Enterprise 1 HQ Components

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

- Cisco Unified CM (version 6.1)
- SCCP IP Phones
- VG224 (version 15.1(1)T) analog lines for Fax/Modem support
- Cisco UBE (Cisco IOS Release 15.1(1)T)
Enterprise 1 and Branch 1 Components

The main components of the Enterprise 1 and Branch 1 include:
- Cisco UBE/Cisco Unified SRST/Analog lines for Fax/Modem
- SCCP IP Phones

Caveats

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

Global Caveats

In general, the following global caveats exist with this solution:
- The same static codec must be used on all voice calls. It can be any codec type, but the same codec must be maintained.
- Intra Enterprise calls were tested with G.711 codecs. SIP trunk calls were tested with G.729r8 codecs.
- Voice calls over the WAN must be configured with G.729 codecs.
- Video was not tested as part of this solution.
- H.323 calls were not tested as part of this solution.
- Use of Cisco HSRP is not recommended in this solution as it can cause unexpected results with SIP signaling.

Cisco Unified CM 6.1.0.9901-372 Caveats

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

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

Cisco Unity 5.0(1) Caveats

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

Cisco Unity Express 3.2 Caveats

To view the caveats for Cisco Unity Express 3.2, see Release Notes for Cisco Unity Express 3.2.
Cisco ASA 8.0(4) Caveats

High-Level Operation

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

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

Call Flow Within Enterprise 1

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

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

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

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

HQ Call Flow to Enterprise Offsite Remote Endpoint

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

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

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

Branch 1 Call Flow to Enterprise Offsite Remote Endpoint

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

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

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

Note

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

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

For purposes of security, the Cisco ASA can be placed at the front end of the HQ Cisco UBE.
High-Level Configuration Summaries

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

Protocols

The following is a list of protocols used between components:

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

Codecs

The following is a list of codecs used between components:

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

Note

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

DSP Farms

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

The following is a list of supplementary services:

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

**Call Admission Control**

The call admission control (CAC) restrictions that are imposed by Cisco Unified CM for the whole enterprise are as follows:
• **BANDWIDTH**—With Static Location. Cisco Unified CM restricts max voice and fax/modem calls to configured bandwidth threshold for both enterprise HQ and the Branch uplinks under “Location/Audio calls information.”

• **NUMBER of CALLS**—The Branch Cisco UBE must be configured to activate when in Cisco Unified SRST mode only, which means that the max-calls/bandwidth threshold should be larger than the setting for Cisco Unified CM. Cisco Unified CM would be the triggering mechanism under normal circumstances.

• **CPU%**—Cisco UBE at the enterprise HQ and the Branch restrict the maximum voice and fax/modem calls to configured CPU% threshold.

• **MEMORY**—Cisco UBE at the enterprise HQ and the Branch restrict the maximum voice and fax/modem calls to the configured available memory threshold.

**Test Topology**

Figure 7 shows the setup test topology used in example configurations described in the following sections.
Example Configuration Details

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

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

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

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

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

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

Enterprise 1 HQ Cisco UBE Example Configuration

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

Ent1_HQ_CUBE1#
  !
  voice-card 0
dspfarm
dsp services dspfarm
  !
voice service voip
  mode border-element
  allow-connections h323 to h323
  allow-connections h323 to sip
  allow-connections sip to h323
  allow-connections sip to sip
  signaling forward unconditional
  fax protocol pass-through g711ulaw
  modem passthrough nse codec g711ulaw
  h323
    emptycapability
  h245 passthru tcsnonstd-passthru
  sip
    bind control source-interface Loopback0
    bind media source-interface Loopback0
    min-se 2000
    header-passing error-passthru
    options-ping 1200
    listen-port non-secure 5090
    midcall-signaling passthru
  !
  voice translation-rule 1
    rule 1 /^61/ /1/
    rule 2 /^71/ /1/
  !
  voice translation-profile OUTGOING-SIP-TRK-DIGIT-STRIP
    translate called 1
  !
  interface Loopback0
    ip address 10.10.11.151 255.255.255.255
  !
  interface GigabitEthernet0/0
    ip address 10.40.97.1 255.255.255.0
    duplex full
    speed 100
    media-type rj45
    no keepalive
  !
  interface GigabitEthernet0/1
    ip address 10.40.99.2 255.255.255.0
    duplex full
    speed 100
    media-type rj45
    no keepalive
  !
  ip rtcp report interval 9000
  !
  sccp local GigabitEthernet0/0
  sccp ccm 10.40.97.2 identifier 5 priority 1 version 6.0
  sccp
  !
  sccp ccm group 10
    associate ccm 5 priority 1
    associate profile 10 register MTP111222333
    associate profile 12 register CON111222333
    associate profile 11 register XCODE111222333
  !
  dspfarm profile 11 transcode
    codec g711ulaw
    codec g729r8
    maximum sessions 10
    associate application SCCP
  !
Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

Enterprise 1 HQ Cisco UBE Example Configuration

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
Enterprise 1 HQ Cisco Unified CM Example Configuration

The following example shows the required field and parameter entries for example configuration of the Cisco Unified CM for the topology shown in Figure 7. Parameters are entered using the Cisco Unified CM GUI. The example parameters windows entries are shown in following sections:

- Configuring the Cisco Unified CM System Parameters, page 35
- Configuring the Cisco Unified CM Call Routing Parameters, page 65
- Configuring the Cisco Unified CM Media Resources Parameters, page 81
Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

Enterprise 1 HQ Cisco Unified CM Example Configuration

- Configuring the Cisco Unified CM Voice Mail Parameters, page 98
- Configuring the Cisco Unified CM Device Parameters, page 105

Configuring the Cisco Unified CM System Parameters

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

- **System: Server Parameters**, page 35
- **System: Region Parameters**, page 36
- **System: Device Pool Parameters**, page 49
- **System: Location Parameters**, page 58

System: Server Parameters

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

![System Server Enterprise 1 HQ Cisco Unified CM Administration Window](image)

- **Server Information**
  - **Database Replication**: Publisher
  - **Host Name/IP Address**: 10.40.97.2
  - **MAC Address**: Enter
  - **Description**: Ent-HQ-CUCM
Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

Enterprise 1 HQ Cisco Unified CM Example Configuration

System: Region Parameters

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

Figure 9 System Region Cisco Unified CM Administration Window

---

SIP-Based Trunk Managed Voice Services Solution Design and Implementation Guide
Figure 10  System Region Default Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_HQ_Phones_IP</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Wan</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

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

**Modify Relationship to other Regions**

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>Region_Br1_Phones_Analog</td>
<td>Use System Default</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>Region_Br1_DSPfarm</td>
<td>Use System Default</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Conference</td>
<td>Use System Default</td>
<td>None</td>
<td></td>
</tr>
<tr>
<td>Region_Br1_DSFarm_Transcoder</td>
<td>Use System Default</td>
<td>None</td>
<td></td>
</tr>
</tbody>
</table>

* - 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 Region-Region Branch 1 Phones Analog Cisco Unified CM Administration Window

### Region Information
- **Name**: Region Br1 Phones Analog

### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region_Br1_Phone_IP</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_HQ_Phone_Analog</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_Phone_IP</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_Phone_Analog</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

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

### Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>Region_Br1_Phone_Analog</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm</td>
<td>G.711</td>
<td>384</td>
<td>None</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Conference</td>
<td>G.711</td>
<td>384</td>
<td>64 kbps</td>
</tr>
</tbody>
</table>

**i**: *indicates required item.

**ii**: 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 DSP Farm Cisco Unified CM Administration Window

Cisco Unified CM Administration

For Cisco Unified Communications Solutions

Region Configuration

Region Information
- Name: Region_Br1_DSPfarm

Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region_Br1_DSPfarm</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_Phone/IP</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_HQ_Phone/IP</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Wan</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

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

Modify Relationship to other Regions

<table>
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<tr>
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<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>Region_Br1_Phone/Analog</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Region_Br1_DSPfarm</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Conference</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Transcoder</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

- Save  | Delete | Reset | Add New |

* - indicates required item.

**The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 kbps between regions and can be used interchangeably.
**Figure 13**  
System Region-Region Branch 1 DSP Farm Conference Cisco Unified CM Administration Window

<table>
<thead>
<tr>
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</tr>
<tr>
<td>Region_HQ_Phone_IP</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Wan</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

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

**Modify Relationship to Other Regions**

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<td></td>
</tr>
<tr>
<td>Region_Br1_DSPFarm_Conference</td>
<td></td>
<td></td>
<td></td>
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<tr>
<td>Region_Br1_DSPFarm_Transcoder</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

* 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.**
### Region Information

<table>
<thead>
<tr>
<th>Name</th>
<th>Region_ID</th>
</tr>
</thead>
<tbody>
<tr>
<td>region_bri_DSPFARM_Transcoder</td>
<td>region_bri_DSPFARM_Transcoder</td>
</tr>
</tbody>
</table>

### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>region_bri_DSPFARM_Transcoder</td>
<td>6.711</td>
<td>394</td>
<td>Use System Default</td>
</tr>
<tr>
<td>region_bri_Phone_IP</td>
<td>6.711</td>
<td>394</td>
<td>Use System Default</td>
</tr>
<tr>
<td>region_wan</td>
<td>6.729</td>
<td>394</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**Note:** Regions(s) not displayed: Use System Default

### Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
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<td>Default</td>
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<td>Keep Current Setting</td>
</tr>
<tr>
<td>region_bri_Phone_Analog</td>
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<td>region_bri_DSPFARM</td>
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</tr>
<tr>
<td>region_bri_DSPFARM_Conference</td>
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<td>region_bri_DSPFARM_Transcoder</td>
<td></td>
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</table>

* - indicates required item.

**The Audio Codec selection determines bandwidth only. The 6.711 and 6.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.
### Figure 15 System Region-Region Branch 1 Phones IP Cisco Unified CM Administration Window

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

**Navigation** Cisco Unified CM Administration

**Region Configuration**

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

---

**Region Information**

Name: Region_Br1_Phone_IP

---

**Region Relationships**

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region_Br1_DSPfarm</td>
<td>6.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Conference</td>
<td>6.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Transcoder</td>
<td>6.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_Phone_IP</td>
<td>6.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_HQ_DSPfarm</td>
<td>6.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_HQ_DSPfarm_Conference</td>
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<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_HQ_Phone_IP</td>
<td>6.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Wan</td>
<td>6.729</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_Phone_Analog</td>
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**NOTE:** Regions(s) not displayed

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**Modify Relationship to other Regions**

<table>
<thead>
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<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default</td>
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<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>Region_Br1_Phone_Analog</td>
<td>Keep Current Setting</td>
<td>Use System Default</td>
<td>None</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm</td>
<td>Keep Current Setting</td>
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<td>None</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Conference</td>
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<td>Use System Default</td>
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</tr>
</tbody>
</table>

---

* indicates required item.

**The Audio Codec selection determines bandwidth only. The 6.711 and 6.722 codecs both result in a maximum bandwidth of 64 Kbps between regions and can be used interchangeably.**
**Figure 16**  System Region-Region HQ DSP Farm Cisco Unified CM Administration Window

### Cisco Unified CM Administration

**For Cisco Unified Communications Solutions**  

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

**Region Configuration**

**Related Links:**  
- Back To Find/List

---

### Region Information

**Name:** Region_HQ_DSPFarm

---

### Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region_Br1_Phone_IP</td>
<td>G.729</td>
<td>384</td>
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<tr>
<td>Region_HQ_DSPFarm</td>
<td>G.729</td>
<td>384</td>
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</tr>
<tr>
<td>Region_HQ_Phone_IP</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Wan</td>
<td>G.729</td>
<td>384</td>
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</tr>
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**NOTE:** Regions(s) not displayed  
- Use System Default  
- Use System Default  
- Use System Default

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### Modify Relationship to other Regions

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<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
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<td></td>
<td>Keep Current Setting</td>
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<tr>
<td>Region_Br1_DSPFarm</td>
<td></td>
<td></td>
<td>Keep Current Setting</td>
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<tr>
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<tr>
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<td></td>
<td></td>
<td></td>
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---

**i** - indicates required item.  

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

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

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

![Cisco Unified CM Administration Window](image)

**Region Information**

<table>
<thead>
<tr>
<th>Name</th>
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</tr>
</thead>
</table>

**Region Relationships**

<table>
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<th>Link Loss Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>Region_Br1_Phones_IP</td>
<td>G.729</td>
<td>384</td>
<td>Use System Default</td>
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<tr>
<td>Region_HQ_Phones_IP</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Wan</td>
<td>G.729</td>
<td>384</td>
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**NOTE:** Regions(s) not displayed

<table>
<thead>
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<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
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<tbody>
<tr>
<td>Default</td>
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<tr>
<td>Region_Br1_Phones_Analog</td>
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<tr>
<td>Region_Br1_DSPfarm</td>
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**Modify Relationship to other Regions**

<table>
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<th>Link Loss Type</th>
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</thead>
<tbody>
<tr>
<td></td>
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<td></td>
<td></td>
</tr>
</tbody>
</table>

**Related Links:**

- Save
- Delete
- Reset
- Add New

---

* * indicates required item.

**The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 kbps between regions and can be used interchangeably.**
Figure 18  System Region-Region HQ DSP Farm Transcoder Cisco Unified CM Administration Window

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

<table>
<thead>
<tr>
<th>Region Relation</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
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</thead>
<tbody>
<tr>
<td>Region_HQ_DSPFarm_Transcoder</td>
<td>G.711</td>
<td>364</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_HQ_Phone_IP</td>
<td>G.711</td>
<td>364</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Wan</td>
<td>G.729</td>
<td>364</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

**Region Information**

Name: Region_HQ_DSPFarm_Transcoder

**Region Relationships**

- **Region Name**: Region_HQ_DSPFarm_Transcoder
- **Audio Codec**: G.711
- **Video Call Bandwidth**: 364
- **Link Loss Type**: Use System Default

**Modify Relationship to other Regions**

<table>
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<th>Link Loss Type</th>
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</thead>
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<td>Keep Current Setting</td>
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<td></td>
</tr>
<tr>
<td>Region_Br1_DSPFarm</td>
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<td></td>
</tr>
<tr>
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</table>

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

#### Region Information
- **Name**: Region_HQ_Phones_Analog

#### Region Relationships
<table>
<thead>
<tr>
<th>Region</th>
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<tr>
<td>Region_HQ_Phones_IP</td>
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<tr>
<td>Region_Wan</td>
<td>G.711</td>
<td>384</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_Phones_Analog</td>
<td>G.711</td>
<td>384</td>
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**NOTE**: Regions(s) not displayed
- Use System Default
- Use System Default
- Use System Default

#### Modify Relationship to other Regions
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</thead>
<tbody>
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<td>Keep Current Setting</td>
</tr>
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<td>Region_Br1_Phones_Analog</td>
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<td>Region_Br1_DSPfarm</td>
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<td></td>
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<tr>
<td>Region_Br1_DSPfarm_Conference</td>
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<td></td>
<td></td>
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<tr>
<td>Region_Br1_DSPfarm_Transcorder</td>
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</tr>
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</table>

**i**: *indicates required item.

**ii**: The Audio Codec selection determines bandwidth only. The G.711 and G.722 codecs both result in a maximum bandwidth of 64 kbps between regions and can be used interchangeably.
Figure 20  
System Region-Region HQ Phones IP Cisco Unified CM Administration Window

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

Region Configuration  
Related Links:

<table>
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<tr>
<th>Region Information</th>
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</thead>
<tbody>
<tr>
<td>Name*</td>
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</tbody>
</table>

<table>
<thead>
<tr>
<th>Region Relationships</th>
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</tr>
<tr>
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</tr>
<tr>
<td>Region_Br1_DSPFarm</td>
</tr>
<tr>
<td>Region_Br1_DSPFarm_Conference</td>
</tr>
<tr>
<td>Region_Br1_Phone_IP</td>
</tr>
<tr>
<td>Region_HQ_DSPFarm</td>
</tr>
<tr>
<td>Region_HQ_DSPFarm_Conference</td>
</tr>
<tr>
<td>Region_HQ_DSPFarm_Transcorder</td>
</tr>
<tr>
<td>Region_HQ_Phone_Analog</td>
</tr>
<tr>
<td>Region_HQ_Phone_IP</td>
</tr>
<tr>
<td>Region_Wan</td>
</tr>
<tr>
<td>Region_Br1_Phone_Analog</td>
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NOTE: Regions(s) not displayed
Use System Default  Use System Default  Use System Default

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<th>Modify Relationship to other Regions</th>
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</thead>
<tbody>
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</tr>
<tr>
<td>Region_Br1_Phone_Analog</td>
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<td>Region_Br1_DSPFarm</td>
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<tr>
<td>Region_Br1_DSPFarm_Conference</td>
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<td>Region_Br1_DSPFarm_Transcorder</td>
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</tbody>
</table>

- 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 21  System Region-Region WAN Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Region Configuration

Region Information
Name: Region_Wan

Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
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</thead>
<tbody>
<tr>
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</tr>
<tr>
<td>Region_Br1_DSPfarm_Conference</td>
<td>0.729</td>
<td>304</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Transcoder</td>
<td>0.729</td>
<td>304</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Region_Br1_Phone_IP</td>
<td>0.729</td>
<td>304</td>
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</tr>
<tr>
<td>Region_HQ_DSPfarm</td>
<td>0.729</td>
<td>304</td>
<td>Use System Default</td>
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<td>Region_HQ_DSPfarm_Conference</td>
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<tr>
<td>Region_HQ_DSPfarm_Transcoder</td>
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<td>304</td>
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</tr>
<tr>
<td>Region_HQ_Phone_IP</td>
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<td>384</td>
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<td>Region_HQ_Phone_Analog</td>
<td>6.711</td>
<td>384</td>
<td>Use System Default</td>
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<td>Region_Wan</td>
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<td>Region_Br1_Phone_Analog</td>
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NOTE: Regions(s) not displayed: Use System Default

Modify Relationship to other Regions

<table>
<thead>
<tr>
<th>Regions</th>
<th>Audio Codec</th>
<th>Video Call Bandwidth</th>
<th>Link Loss Type</th>
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<tbody>
<tr>
<td>Default</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>Region_Br1_Phone_Analog</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
<td>Keep Current Setting</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm</td>
<td>Use System Default</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Conference</td>
<td>Use System Default</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>Region_Br1_DSPfarm_Transcoder</td>
<td>Use System Default</td>
<td>None</td>
<td>None</td>
</tr>
</tbody>
</table>

- Save  Delete  Reset  Add New

* indicates required item.
** The Audio Codec selection determines bandwidth only. The 6.711 and 6.722 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** in the Cisco Unified CM Administration window.

*Figure 22  System Device Pool Cisco Unified CM Administration Window*
### Figure 23  System Device Pool Default Cisco Unified CM Administration Window

**Cisco Unified CM Administration**

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td>Save</td>
<td>Delete</td>
</tr>
</tbody>
</table>

#### 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

- **Data/Time Group**: CMLocal
- **Region**: Default
- **Media Resource Group List**: < None >
- **Location**: < None >
- **Network Locale**: < None >
- **SRST Reference**: Disable
- **Connection Monitor Duration**: ***
- **Single Button Range**: 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 >

---

* 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 Device Pool-DevicePool Branch 1 Analog Phones Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
</table>

#### Status
- Status: Ready

#### Device Pool Information
- Device Pool: DevicePool_Br1_Analog_Phone (2 members**)

#### Device Pool Settings
- **Device Pool Name**: DevicePool_Br1_Analog_Phone
- **Calling Search Space for Auto-registration**: Branch 1
- **Reverted Call Focus Priority**: Default

#### Roaming Sensitive Settings
- **Date/Time Group**: CM Local
- **Region**: Region Branch 1 Phones Analog
- **Media Resource Group List**: Br1_HW_MRGCL
- **Location**: Hub Br1
- **Network Locale**: < None >
- **SRST Reference**: SRST_Emtl_Br1
- **Connection Monitor Duration**: < None >
- **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 >

---

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

#### Figure 25: System Device Pool-DevicePool Branch 1 DSP Farm Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links:</th>
<th>Back To Find/List</th>
</tr>
</thead>
</table>

#### Cisco Unified CM Administration
For Cisco Unified Communications Solutions

<table>
<thead>
<tr>
<th>System</th>
<th>Call Reading</th>
<th>Media Resources</th>
<th>Voice Mail</th>
<th>Device</th>
<th>Application</th>
<th>User Management</th>
<th>Bulk Administration</th>
<th>Help</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

#### Device Pool Configuration

- **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**
  - Data/Time Group*:
    - CMLocal
  - Region*:
    - Region_Br1_DSPfarm
  - Media Resource Group List:
    - Br1 HW MRLG
  - Location:
    - Hub_Br1
  - Network Locale:
    - < None >
  - SRST Reference*:
    - Disable
  - Connection Monitor Duration***:
    - Disable
  - Single Button Range*:
    - 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 >

---

**i**: indicates required item.

**ii**: 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.

**iii**: Leave blank to use default.

**iv**: 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 IP Phones Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status</td>
<td>Status: Ready</td>
</tr>
<tr>
<td><strong>Device Pool Information</strong></td>
<td></td>
</tr>
<tr>
<td>Device Pool: DevicePool_Br1_IP_Phone (5 members***)</td>
<td></td>
</tr>
<tr>
<td>Device Pool Name*:</td>
<td>DevicePool_Br1_IP_Phone</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Group*:</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space for Auto-registration</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Reverted Call Focus Priority</td>
<td>Default</td>
</tr>
<tr>
<td><strong>Roaming Sensitive Settings:</strong></td>
<td></td>
</tr>
<tr>
<td>Date/Time Group*:</td>
<td>CMLocal</td>
</tr>
<tr>
<td>Region*:</td>
<td>Region_Br1_IPPhones</td>
</tr>
<tr>
<td>Media Resource Group List:</td>
<td>Br1 HW MRGL</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_Br1</td>
</tr>
<tr>
<td>Network Locale:</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>SRST Reference*:</td>
<td>SRST_Ent1_Br1</td>
</tr>
<tr>
<td>Connection Monitor Duration**:</td>
<td></td>
</tr>
<tr>
<td>Single Button Barge*:</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines*:</td>
<td>Default</td>
</tr>
<tr>
<td>Physical Location:</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Device Mobility Group:</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td><strong>Device Mobility Related Information</strong>**:**</td>
<td></td>
</tr>
<tr>
<td>Device Mobility Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt;None&gt;</td>
</tr>
</tbody>
</table>

* 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 Dependence 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 HQ Analog Phones Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Device Pool Configuration

Status

Device Pool Information
Device Pool: DevicePool HQ Analog Phones (3 members**)

Device Pool Settings
Device Pool Name* CiscoUnifiedCommunicationsManagerGroup*
DevicePool HQ Analog Phones Default

Calling Search Space for Auto-registration
Reverted Call Focus Priority

Roaming Sensitive Settings
Data/Time Group*
Region*
Media Resource Group List
Location
Network Locale
SRST Reference* Connection Monitor Duration***

Single Button Digits*
Join Across Lines*
Physical Location
Device Mobility Group

Device Mobility Related Information****
Device Mobility Calling Search Space
AAR Calling Search Space
AAR Group

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.

23/07/2012
### Figure 28  System Device Pool-DevicePool HQ DSP Farm Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Device Pool Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Pool Name*</td>
<td>DevicePool_HQ_DSPFarm</td>
</tr>
<tr>
<td>Cisco Unified Communications Manager Group*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space for Auto-registration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Reverted Call Focus Priority</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Roaming Sensitive Settings</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Date/Time Group*</td>
<td>CMLocal</td>
</tr>
<tr>
<td>Region*</td>
<td>Region_HQ_DSPFarm</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>HQ HW MGRL</td>
</tr>
<tr>
<td>Location</td>
<td>Hub_HQ</td>
</tr>
<tr>
<td>Network Locale</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>SRST Reference*</td>
<td>Disable</td>
</tr>
<tr>
<td>Connection Monitor Duration***</td>
<td></td>
</tr>
<tr>
<td>Single Button Barge*</td>
<td>Default</td>
</tr>
<tr>
<td>Join Across Lines*</td>
<td>Default</td>
</tr>
<tr>
<td>Physical Location</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Device Mobility Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Device Mobility Related Information****</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Mobility Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

---

1. * indicates required item.
2. **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 Dependence Records.
3. ***Leave blank to use default.
4. ****These three parameters will overwrite device level settings when device is roaming and in the same device mobility group.
**Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations**

### Enterprise 1 HQ Cisco Unified CM Example Configuration

#### Figure 29  System Device Pool-DevicePool HQ IP Phones Cisco Unified CM Administration Window

![Cisco Unified CM Administration Window](image)

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

**Device Pool Configuration**

<table>
<thead>
<tr>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
</table>

- **Status**
  - Status: Ready

- **Device Pool Information**
  - Device Pool: DevicePool_HQ_IP_Phone (12 members***)

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

- **Roaming Sensitive Settings**
  - Data/Time Group*: CMLocal
  - Region*: Region_HQPhones_IP
  - Media Resource Group List: HQ HW MRGL
  - Location: Hub_HQ
  - Network Locale: < None >
  - SRST Reference*: SRST_Ent1_Br1
  - Connection Monitor Duration***: 
  - Single Button Range*: 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 >

---

* 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.
### Device Pool Information

**Device Pool**: DevicePool_WAN (2 members)

<table>
<thead>
<tr>
<th>Device Pool Configuration</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Device Pool Name</strong></td>
<td>DevicePool_WAN</td>
</tr>
<tr>
<td><strong>Calling Search Space for Auto-registration</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Reverted Call Focus Priority</strong></td>
<td>Default</td>
</tr>
</tbody>
</table>

### Roaming Sensitive Settings

- **Date/Time Group**: CMLocal
- **Region**: Region_Wan
- **Media Resource Group List**: HQ HW MPR1
- **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 >

---

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

**Figure 31 System Location Cisco Unified CM Administration Window**

<table>
<thead>
<tr>
<th>Location</th>
<th>Audio Bandwidth</th>
<th>Video Bandwidth</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hub Br1</td>
<td>05</td>
<td>NONE</td>
<td><img src="#" alt="Copy" /></td>
</tr>
<tr>
<td>Hub HQ</td>
<td>1.10</td>
<td>NONE</td>
<td><img src="#" alt="Copy" /></td>
</tr>
<tr>
<td>Hub None</td>
<td>UNLIMITED</td>
<td>UNLIMITED</td>
<td><img src="#" alt="Copy" /></td>
</tr>
<tr>
<td>Trunk Br1</td>
<td>05</td>
<td>NONE</td>
<td><img src="#" alt="Copy" /></td>
</tr>
<tr>
<td>Trunk HQ</td>
<td>1.10</td>
<td>NONE</td>
<td><img src="#" alt="Copy" /></td>
</tr>
</tbody>
</table>
Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 32  System Location Hub Branch 1 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Location Configuration

---

**Status:**

Status: Ready

---

**Location Information**

Name: **Hub_Br1**

---

**Audio Calls Information**

Audio Bandwidth: **Unlimited** / 65 kbps

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

---

**Video Calls Information**

Video bandwidth: **Unlimited** / kbps

---

**Location RSVP Settings**

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

---

**Modify Setting(s) to Other Locations**

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hub_Br1</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Hub_HQ</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Hub_Name</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Trunk_Br1</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Trunk_HQ</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

---

* indicates required item.
### System Location Hub HQ Cisco Unified CM Administration Window

![Cisco Unified CM Administration](image)

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

**Location Configuration**

#### 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

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use System Default</td>
<td></td>
</tr>
</tbody>
</table>

#### Modify Setting(s) to Other Locations

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hub_Br1</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Hub_HQ</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Hub_None</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Trunk_Br1</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Trunk_HQ</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

- Status: Indicates required item.
Figure 34  System Location Hub None Cisco Unified CM Administration Window

### Cisco Unified CM Administration

**For Cisco Unified Communications Solutions**

#### System Configuration

<table>
<thead>
<tr>
<th>Location Configuration</th>
<th>Related Links</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status: Ready</td>
<td></td>
</tr>
</tbody>
</table>

#### Location Information

- **Name**: Hub_Name

#### Audio Calls Information

- **Audio Bandwidth**: Unlimited
- **Kbps**: [Value]

*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**: Unlimited
- **Kbps**: [Value]

#### Location RSVP Settings

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Hub_Brl</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Hub_HQ</td>
<td></td>
</tr>
<tr>
<td>Hub_Name</td>
<td></td>
</tr>
<tr>
<td>Trunk_Brl</td>
<td></td>
</tr>
<tr>
<td>Trunk_HQ</td>
<td></td>
</tr>
</tbody>
</table>

---

* indicates required item.
### System Location-Location Trunk Branch 1 Cisco Unified CM Administration Window

#### Location Configuration
- **Name**: [Trunk Br1]

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

#### Video Calls Information
- **Video Bandwidth**: 0 kbps

#### Location RSVP Settings

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use System Default</td>
<td></td>
</tr>
</tbody>
</table>

#### Modify Setting(s) to Other Locations

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Use System Default</td>
<td></td>
</tr>
</tbody>
</table>

---

* Indicates required item.
### Figure 36  System Location-Location Trunk HQ Cisco Unified CM Administration Window

#### Cisco Unified CM Administration

For Cisco Unified Communications Solutions

### Location Configuration

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunk HQ</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

#### 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 64 kbps or 64 kbps.

#### Video Calls Information

Video Bandwidth: Unlimited

#### Location RSVP Settings

<table>
<thead>
<tr>
<th>Location</th>
<th>RSVP Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

---

* indicates required item.
Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

System: SRST Parameters

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

Figure 37  System SRST-SRST Enterprise 1 Branch 1 Cisco Unified CM Administration Window

Status:
- Status: Ready

SRST Reference Status
- SRST Reference: SRST_Ent1_Bri (used by 13 devices)

SRST Reference Information
- Name*: SRST_Ent1_Bri
- Port*: 2000
- IP Address*: 10.40.103.1
- SIP Network/IP Address
- SIP Port*: 5060
- SRST Certificate Provider Port*: 2445
- Is SRST Secure?

- Save  Delete  Copy  Reset  Add New

*- indicates required item.
Configuring the Cisco Unified CM Call Routing Parameters

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

- **Call Routing: Route/Hunt Parameters**, page 65
- **Call Routing: Class of Control Parameters**, page 71

**Call Routing: Route/Hunt Parameters**

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

![Figure 38 Call Routing Route/Hunt Route Pattern Cisco Unified CM Administration Window](image-url)
### 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

**Route Pattern Configuration**

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
</table>

#### Pattern Definition
- **Route Pattern**: 9.1.XXXX.XXXX
- **Route Partition**: Partition-HQ_Phone_IP
- **Description**: RP Ent1-HQ IP Phone LongDistance
- **Numbering Plan**: Not selected
- **Route Filter**: None
- **MUP Precedence**: Default
- **Gateway/Route List**: 10.10.10.151
- **Route Option**: Block this pattern
- **Call Classification**: Offnet

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

- **Require Forced Authentication Code**
- **Authorization Level**: 0
- **Require Client Matter Code**

#### Calling Party Transformations
- **Use Calling Party's External Phone Number 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)**

#### ISDN Network Specific Facilities Information Element
- **Network Service Protocol**: Not selected
- **Carrier Identification Code**
- **Network Service**: Not selected
- **Service Parameter Name**: Not selected
- **Service Parameter Value**

- **Save**
- **Delete**
- **Copy**
- **Add New**

* Indicates required item.
**Figure 40** Call Routing RouteHunt Route Pattern RP Ent1 HQ Analog Phone LongDistance Administration Window

### Route Pattern Configuration

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>9.1XXXXXXX00000</td>
</tr>
<tr>
<td>Route Partition</td>
<td>Partition-HQ_Phones_Analog</td>
</tr>
<tr>
<td>Description</td>
<td>RP Ent1-HQ Analog Phone LongDistance</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLLP Precedence</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>16.10.11.151</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Offnet</td>
</tr>
<tr>
<td>Allow Device Override</td>
<td></td>
</tr>
<tr>
<td>Provide Outside Dial Tone</td>
<td></td>
</tr>
<tr>
<td>Allow Overlap Sending</td>
<td></td>
</tr>
<tr>
<td>Urgent Priority</td>
<td></td>
</tr>
<tr>
<td>Require Forced Authorization Code</td>
<td></td>
</tr>
<tr>
<td>Authorization Level</td>
<td>0</td>
</tr>
<tr>
<td>Require Client Matter Code</td>
<td></td>
</tr>
</tbody>
</table>

### Calling Party Transformations

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

### Connected Party Transformations

- **Connected Line ID Presentation**
- **Connected Name Presentation**

### Called Party Transformations

- **Discard Digits**
- **Called Party Transform Mask**
- **Prefix Digits (Outgoing Calls)**

### ISDN Network-Specific Facilities Information Element

- **Network Service Protocol**
- **Carrier Identification Code**

### Additional Information

- *- indicates required item.
Figure 41 Call Routing Route Hunt Route Pattern RP Ent1 Br1 Analog Phone Long Distance Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Route Pattern Configuration

- Status
  Status: Ready

- Pattern Definition
  Route Pattern: 9.1,XXX,XXX,XXX
  Route Partition: Partition-Br1_Analog
  Description: RP Ent1-Br1 Analog Phone Long Distance
  Numbering Plan: Not Selected
  Route Filter: None
  MIP Precedence: Default
  Gateway/Route List: 1.80.80.82 (Edit)
  Route Option: Route this pattern
  Default Route Pattern: No Error
  Call Classification: OffNet
  Allow Device Override: Provide Outside Dial Tone: Allow Overlap Sending: Urgent Priority
  Require Forced Authentication Code
  Authorization Level: 0
  Require Client Master Code

- Calling Party Transformations
  Use Calling Party's External Number Mask
  Calling Party Transform Mask: 415555XXX
  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
  Called Party Transform Mask: PreDot
  Prefix Digits (Outgoing Calls)

- ISDN Network-Specific Facilities Information Element
  Network Service Protocol: Not Selected
  Carrier Identification Code
  Network Service: Not Selected
  Service Parameter Name: < Not Exist >
  Service Parameter Value

- Save Delete Copy Add New

* indicates required item.
Figure 42 Call Routing Route Hunt Route Pattern RP Ent1 Br1 IP Phone Long Distance Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Route Pattern Configuration

Status
Status: Ready

Pattern Definition
Route Pattern: 9.1000000000
Route Partition: Partition-Br1_Phones_IP
Description: RP Ent1-BR1 IP Phone LongDistance
Numbering Plan: -- Not Selected --
Route Filter:
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: 1555550000
Prefix Digits (Outgoing Calls):
Calling Line ID Presentation:
Calling Name Presentation:

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): 5

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 Exists

Save Delete Copy Add New

* Indicates required item.
Call Routing: Class of Control Parameters

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

**Figure 43  Call Routing Class of Control Partition Cisco Unified CM Administration Window**
Figure 44  Call Routing Class of Control Partition-Partition Br1 Phones Analog Administration Window

Partition Information

- Name*: Partition-Br1_Phone_Analog
- Description: Analog Phones
- Time Schedule: <None>
- Time Zone: Greenwich Standard Time

* indicates required item.
### Figure 45 Call Routing Class of Control Partition-Partition Br1 Phones IP Cisco Unified CM Administration Window

**Partition Configuration**

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
</table>

**Partition Information**

<table>
<thead>
<tr>
<th>Name*</th>
<th>Partition-Br1_Phone_IP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>IP Phones</td>
</tr>
<tr>
<td>Time Schedule</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Time Zone</td>
<td>Originating Device</td>
</tr>
<tr>
<td>Specific Time Zone</td>
<td>Greenwich Standard Time</td>
</tr>
</tbody>
</table>

* - indicates required item.
**Figure 46 Call Routing Class of Control Partition-Partition HQ Phones Analog Administration Window**

<table>
<thead>
<tr>
<th>Partition Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>System</strong></td>
<td>Navigation: Cisco Unified CM Administration</td>
</tr>
<tr>
<td><strong>Call Routing</strong></td>
<td>admin</td>
</tr>
<tr>
<td><strong>Media Resources</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Voice Mail</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Device</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Application</strong></td>
<td></td>
</tr>
<tr>
<td><strong>User Management</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Bulk Administration</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Help</strong></td>
<td></td>
</tr>
</tbody>
</table>

### Partition Information

- **Name**: Partition-HQ_Phone_Analog
- **Description**: Analog Phones
- **Time Schedule**: None
- **Time Zone**: Originating
- **Specific Time Zone**: Greenwich Standard Time

- **Status**: Ready

<i>* indicates required item.*
Figure 47 Call Routing Class of Control Partition-Partition HQ Phones IP Cisco Unified CM Administration Window

**Partition Information**

- **Name**: Partition-HQ_Phone_IP
- **Description**: IP Phones
- **Time Schedule**: <None>
- **Time Zone**: Originating Device

* - indicates required item.
### Figure 48  Call Routing Class of Control CSS Cisco Unified CM Administration Window

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

![Cisco Unified CM Administration Window](image)

#### Find and List Calling Search Spaces

- Add New
- Select All
- Clear All
- Delete Selected

**Status**

5 records found

#### Calling Search Space (2 - 5 of 5)

Find Calling Search Space where **CSS Name** begins with **x**

<table>
<thead>
<tr>
<th>CSS Name</th>
<th>Description</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>CSS-Br1_Phones_Analog</td>
<td>CSS-Br1_Phones_Analog</td>
<td></td>
</tr>
<tr>
<td>CSS-Br1_Phones_IP</td>
<td>CSS-Br1_Phones_IP</td>
<td></td>
</tr>
<tr>
<td>CSS-HQ_Phones_Analog</td>
<td>CSS-HQ_Phones_Analog</td>
<td></td>
</tr>
<tr>
<td>CSS-HQ_Phones_IP</td>
<td>CSS-HQ_Phones_IP</td>
<td></td>
</tr>
</tbody>
</table>

Add New | Select All | Clear All | Delete Selected |
**Figure 49**  
**Call Routing Class of Control CSS-CSS Branch 1 Phones Analog Cisco Unified CM Administration Window**

**Cisco Unified CM Administration**

*For Cisco Unified Communications Solutions*

**Calling Search Space Configuration**

---

**Status**

Status: Ready

---

**Calling Search Space Information**

Name: CSS-Bri1_Phones_Analog

Description: CSS-Bri1_Phones_Analog

---

**Route Partitions for this Calling Search Space**

Available Partitions:

- Partition-Bri1_Phones_Analog
- Partition-Bri1_Phones_IP
- Partition-HQ_Phones_Analog
- Partition-HQ_Phones_IP

Selected Partitions:

- Partition-Bri1_Phones_Analog
- Partition-Bri1_Phones_IP
- Partition-HQ_Phones_Analog
- Partition-HQ_Phones_IP

---

* - Indicates required item.
** - Selected Partitions are ordered by highest priority.
Figure 50  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

Calling Search Space Configuration

Status
Status: Ready

Calling Search Space Information
Name: CSS-Br1_Phone_IP
Description: CSS-Br1_Phone_IP

Route Partitions for this Calling Search Space
Available Partitions:

Selected Partitions
Partition-Br1_Phone_IP
Partition-Br1_Phone_Analog
Partition-HQ_Phone_Analog
Partition-HQ_Phone_IP

Save  Delete  Copy  Add New

* indicates required item.
**Selected Partitions are ordered by highest priority
Figure 51  Call Routing Class of Control CSS-CSS HQ Phones Analog Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Calling Search Space Configuration

- Status
  Status: Ready

- Calling Search Space Information
  Name: CSS-HQ_Phonexs_Analog
  Description: CSS-HQ_Phonexs_Analog

- Route Partitions for this Calling Search Space
  Available Partitions:
  Selected Partitions:
  Partition-HQ_Phonexs_Analog
  Partition-B1_Phonexs_Analog
  Partition-B1_Phonexs_IP
  Partition-HQ_Phonexs_IP

Save  Delete  Copy  Add New

* Indicates required item.
** Selected Partitions are ordered by highest priority.
### Figure 52: 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

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

### Calling Search Space Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Status: Ready</th>
</tr>
</thead>
</table>

#### Calling Search Space Information

- **Name:** CSS-HQ_Phone_IP
- **Description:** CSS-HQ_Phone_IP

#### Route Partitions for this Calling Search Space

- **Available Partitions:**
  - Partition-HQ_Phone_IP
  - Partition-Branch1_Phone_Analog
  - Partition-Branch2_Phone_IP
  - Partition-Branch2_Phone_Analog

- **Selected Partitions:**
  - Partition-HQ_Phone_IP
  - Partition-Branch1_Phone_Analog
  - Partition-Branch2_Phone_IP
  - Partition-Branch2_Phone_Analog

- **Save** | **Delete** | **Copy** | **Add New**

---

*• Indicates required item.

**•** Selected Partitions are ordered by highest priority
Configuring the Cisco Unified CM Media Resources Parameters

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

- Media Resources: Annunciator Parameters, page 81
- Media Resources: Conference Bridge Parameters, page 82
- Media Resources: Media Termination Point Parameters, page 85
- Media Resources: Music on Hold Server Parameters, page 88
- Media Resources: Transcoder Parameters, page 89
- Media Resources: Media Resource Group Parameters, page 92
- Media Resources: Media Resource Group List Parameters, page 95

Media Resources: Annunciator Parameters

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

Figure 53 Media Resources Annunciator ANN 2 Cisco Unified CM Administration Window
Media Resources: Conference Bridge Parameters

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

**Figure 54  Media Resources Conference Bridges Cisco Unified CM Administration Window**
Enterprise 1 HQ Cisco Unified CM Example Configuration

Figure 55  Media Resources Conference Bridges CFB Enterprise 1 Branch 1 Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Conference Bridge Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
</table>

- **Status**
  - Status: Ready

---

- **Conference Bridge Information**
  - Conference Bridge: CONC01AA29DF631 (CFB-Ent1-Bri)
  - Registration: Registered with Cisco Unified Communications Manager 10.40.97.2
  - IP Address: 10.40.130.1

- **Cisco IOS Conference Bridge Info**
  - Conference Bridge Type: Cisco IOS Enhanced Conference Bridge
  - Conference Bridge Name: CONC01AA29DF631
  - Description: CFB-Ent1-Bri
  - Device Pool: DevicePool_Bri_DSPfarm
  - Common Device Configuration: < None >
  - Location: Hub_Bri
  - Device Security Mode: Non Secure Conference Bridge

---

* indicates required item.
Figure 56  Media Resources Conference Bridges CFB Enterprise 1 HQ Cisco Unified CM Administration Window

Conference Bridge Configuration

- 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

* indicates required item.
Media Resources: Media Termination Point Parameters

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

Figure 57 Media Resources Media Termination Point Cisco Unified CM Administration Window
Figure 58  Media Resources Media Termination Point MTP Enterprise 1 Branch 1 Administration Window

---

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

---

**Media Termination Point Configuration**

---

**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**

MTP-Ent1-Brl

**Description**

MTP-Ent1-Brl

**Device Pool**

DevicePool_brl_DSPfarm

---

* - indicates required item.
Figure 59  Media Resources Media Termination Point MTP Enterprise 1 HQ Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Media Resource</th>
<th>Media Termination Point MTP Enterprise 1 HQ Cisco Unified CM Administration Window</th>
</tr>
</thead>
<tbody>
<tr>
<td>Registration</td>
<td>Registered with Cisco Unified Communications Manager 10.4.0.97.2</td>
</tr>
<tr>
<td>IP Address</td>
<td>10.4.0.97.1</td>
</tr>
<tr>
<td>Media Termination Point Type</td>
<td>Cisco IOS Enhanced Software Media Termination Point</td>
</tr>
<tr>
<td>Media Termination Point Name</td>
<td>MTP111222333</td>
</tr>
<tr>
<td>Description</td>
<td>MTP-Ent1-HQ</td>
</tr>
<tr>
<td>Device Pool</td>
<td>DevicePool_HQ_DSPFarm</td>
</tr>
</tbody>
</table>

- Status
- Status: Ready

*: Indicates required item.
Media Resources: Music on Hold Server Parameters

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

Figure 60  Media Resources Music on Hold Server MOH Enterprise 1 HQ Cisco Unified CM Administration Window

---

Device Information

| Registered with Cisco Unified Communications Manager 10.40.97.2 |
| 10.40.97.2 |
| Host Server* |
| 10.40.97.2 |
| Music On Hold Server Name* |
| MOH_Hub_HQ |
| Description |
| Default |
| Location* |
| Hub_HQ |
| Maximum Half Duplex Streams* |
| 250 |
| Maximum Multicast Connections* |
| 50 |
| 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
- 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.
Media Resources: Transcoder Parameters

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

Figure 61  Media Resources Transcoder Cisco Unified CM Administration Window
### Media Resources Transcoder XCODE Enterprise 1 Branch 1 Cisco Unified CM Administration Window

**Cisco Unified CM Administration**

For Cisco Unified Communications Solutions

**Navigation:** Cisco Unified CM Administration

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

**Transcoder Configuration**

| Save | Delete | Copy | Reset | Add New |
---|---|---|---|---|

**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**

<table>
<thead>
<tr>
<th>Transcoder Type</th>
<th>Cisco IOS Enhanced Media Termination Point</th>
</tr>
</thead>
<tbody>
<tr>
<td>Description</td>
<td>XCODE-Ent1-Br1</td>
</tr>
<tr>
<td>Device Name</td>
<td>XCD001AA29DF631</td>
</tr>
<tr>
<td>Device Pool</td>
<td>DevicePool_Br1_DSPfarm</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Special Load Information</td>
<td>Leave blank to use default</td>
</tr>
</tbody>
</table>

* - indicates required item.
Figure 63  Media Resources Transcoder XCODE Enterprise 1 HQ Cisco Unified CM Administration Window

- 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_OSRfarm
  - Common Device Configuration: < None >
  - Special Load Information: Leave blank to use default

* - indicates required item.
Media Resources: Media Resource Group Parameters

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

**Figure 64 Media Resources-Media Resource Group Cisco Unified CM Administration Window**
## Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

### Enterprise 1 HQ Cisco Unified CM Example Configuration

#### Figure 65  Media Resources-Media Resource Group Enterprise 1 Branch 1 Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Media Resource Group Configuration</th>
<th>Related Links:  Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td><img src="image" alt="Cisco Unified CM Administration" /></td>
<td><img src="admin" alt="admin" /> About Logout</td>
</tr>
<tr>
<td>System Call Reading Media Resources Voice Mail Device Application User Management Bulk Administration Help</td>
<td><img src="Save" alt="Save" /> Delete Copy Reset Add New</td>
</tr>
<tr>
<td>Media Resource Group Configuration</td>
<td>![Status](Status: Ready)</td>
</tr>
<tr>
<td>Status: Ready</td>
<td>![Media Resource Group Status](Media Resource Group: BRL_HK_MRG (used by 11 devices))</td>
</tr>
<tr>
<td>![Media Resource Group Information](Name: BRL_HK_MRG Description: Ent 1 Brl)</td>
<td>![Devices for this Group](Available Media Resources: ANN_2 CPB_2 CON111122333 MTP111122333 MTP_2)</td>
</tr>
<tr>
<td>![Selected Media Resources](CON001AA29DF631 (CFB) MDH-Ent1 (MOH) MTP001AA29DF631 (MTP) XCC001AA29DF631 (XCODE))</td>
<td><img src="on" alt="Use Multicast for MOH Audio (if at least one multicast MOH resource is available)" /></td>
</tr>
</tbody>
</table>

---

* - Indicates required item.

** Includes Announciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)
### Media Resources - Media Resource Group Enterprise 1 HQ Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Available Media Resources</th>
<th>Selected Media Resources</th>
</tr>
</thead>
<tbody>
<tr>
<td>ANN_2</td>
<td>CON111222333 (CFB)</td>
</tr>
<tr>
<td>CPU_2</td>
<td>MOH-Ent1 (MOH)</td>
</tr>
<tr>
<td>CONB1A1A92BDE31</td>
<td>MTP111222333 (MTP)</td>
</tr>
<tr>
<td>MTP_2</td>
<td>XCODE111222333 (XCODE)</td>
</tr>
</tbody>
</table>

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

---

- **Status**
  - Status: Ready

- **Media Resource Group Information**
  - Name: HQ_HW_MRG
  - Description: Ent 1 HQ

---

* Indicates required item.

** Includes Annunciators (ANN), Conference Bridges (CFB), Media Termination Points (MTP), Music On Hold Servers (MOH) and Transcoders (XCODE)
Media Resources: Media Resource Group List Parameters

To configure the media resources media resource group list parameters for the Cisco Unified CM, click Media Resources > Media Resource Group List in the Cisco Unified CM Administration window.
**Figure 68** Media Resources-Media Resource Group List Branch 1 HW MRGL Cisco Unified CM Administration Window

- **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_MRGL

- **Selected Media Resource Groups**
  - Br1_HW_MRGL

- **Save** | **Delete** | **Copy** | **Reset** | **Add New**

- * indicates required item.
Figure 69  Media Resources-Media Resource Group List HQ HW MRGL Cisco Unified CM Administration Window

- **Status**
  
  Status: Ready

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

- **Media Resource Group List Information**
  
  Name: HQ HW MRGL

- **Media Resource Groups for this List**
  
  Available Media Resource Groups

  - BR1_HW_MRG
  
  Selected Media Resource Groups

  - HQ HW MRGL

  - ![Image of Media Resources window with selected groups]

  - ![Image of Cisco Unified CM Administration window with media resource group list]

- **Related Links:**
  
  - Back To Find/List

- **Notes:**
  
  - * - indicates required item.
Configuring the Cisco Unified CM Voice Mail Parameters

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

- Voice Mail: Cisco Voice Mail Port Parameters, page 98
- Voice Mail: Message Waiting Parameters, page 100
- Voice Mail: Voice Mail Pilot Parameters, page 103
- Voice Mail: Voice Mail Profile Parameters, page 104

Voice Mail: Cisco Voice Mail Port Parameters

To configure the voice mail Cisco voice mail port parameters for the Cisco Unified CM, click Voice Mail > Cisco Voice Mail Port in the Cisco Unified CM Administration window.
### Figure 71: Voice Mail - Voice Mail Port CiscoUM1 V11 Cisco Unified CM Administration Window

#### Cisco Unified CM Administration

**For Cisco Unified Communications Solutions**

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
</table>

#### Device Information

<table>
<thead>
<tr>
<th>Registration</th>
<th>Registered with Cisco Unified Communications Manager 10.40.97.2</th>
</tr>
</thead>
<tbody>
<tr>
<td>IP Address</td>
<td>10.40.97.258</td>
</tr>
<tr>
<td>Port Name*</td>
<td>CiscoUM1-V1</td>
</tr>
<tr>
<td>Description</td>
<td>Voicemail for Enterprise1</td>
</tr>
<tr>
<td>Device Pool*</td>
<td>DevicePool_HQ_IP_Phone</td>
</tr>
<tr>
<td>Common Device Configuration</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>CSS-HQ_Phone_IP</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Location*</td>
<td>Hub_HQ</td>
</tr>
<tr>
<td>Device Security Mode*</td>
<td>Non Secure Voice Mail Port</td>
</tr>
</tbody>
</table>

#### Directory Number Information

<table>
<thead>
<tr>
<th>Directory Number*</th>
<th>1050</th>
</tr>
</thead>
<tbody>
<tr>
<td>Partition</td>
<td>Partition-HQ_Phone_IP</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>CSS-HQ_Phone_IP</td>
</tr>
<tr>
<td>AAR Group</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Internal Caller ID Display</td>
<td>VoiceMail</td>
</tr>
<tr>
<td>External Number Mask</td>
<td>$15555555</td>
</tr>
</tbody>
</table>

---

* indicates required item.
## Appendix: Enterprise 1 and Branch 1 SIP-Based Trunk Managed Voice Services Solution Example Configurations

### Enterprise 1 HQ Cisco Unified CM Example Configuration

### Voice Mail: Message Waiting Parameters

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

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

<table>
<thead>
<tr>
<th>Directory Number</th>
<th>Description</th>
<th>Partition</th>
<th>Calling Search Space</th>
<th>Copy</th>
</tr>
</thead>
<tbody>
<tr>
<td>1080</td>
<td>MWI-On</td>
<td>Partition-HQ_Phones 1P</td>
<td>CSS-HQ_Phones 1P</td>
<td>☐</td>
</tr>
<tr>
<td>1081</td>
<td>MWI-Off</td>
<td>Partition-HQ_Phones 1P</td>
<td>CSS-HQ_Phones 1P</td>
<td>☐</td>
</tr>
</tbody>
</table>

---

273783
### Figure 73 Voice Mail Message Waiting MWI ON Cisco Unified CM Administration Window

**Cisco Unified CM Administration**

#### Message Waiting Configuration

<table>
<thead>
<tr>
<th>Status</th>
<th>Ready</th>
</tr>
</thead>
</table>

**Message Waiting Information**

- **Message Waiting Number**: 100
- **Partition**: Partition-HQ_Phone/IP
- **Description**: MWI-On
- **Message Waiting Indicator**: On
- **Calling Search Space**: CSS-HQ_Phone/IP

**Related Links**: Back To Find/List
### Figure 74: Voice Mail Message Waiting MWI Off Cisco Unified CM Administration Window

#### Message Waiting Configuration

<table>
<thead>
<tr>
<th>Status: Ready</th>
</tr>
</thead>
</table>

**Message Waiting Information**

<table>
<thead>
<tr>
<th>Message Waiting Number</th>
<th>1001</th>
</tr>
</thead>
<tbody>
<tr>
<td>Partition</td>
<td>Partition-HQ_Phone_IP</td>
</tr>
<tr>
<td>Description</td>
<td>MWI-Off</td>
</tr>
<tr>
<td>Message Waiting Indicator</td>
<td>On/Off</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>CSS-HQ_Phone_IP</td>
</tr>
</tbody>
</table>

*- Indicates required item.*
Voice Mail: Voice Mail Pilot Parameters

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

Figure 75 Voice Mail-Voice Mail Pilot 1099 Cisco Unified CM Administration Window

- Status: Ready

Voice Mail Pilot Information

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

- Save  Delete  Add New

* Indicates required item.
Voice Mail: Voice Mail Profile Parameters

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

**Figure 76 Voice Mail-Voice Mail Profile VM Profile Enterprise 1 HQ Cisco Unified CM Administration Window**

---

**Voice Mail Profile Information**

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

Make this the default Voice Mail Profile for the System

---

* Indicates required item.

** Indicates 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 >).
Configuring the Cisco Unified CM Device Parameters

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

- Device: Gateway Parameters, page 105
- Device: Phone Parameters, page 112
- Device: Trunk Parameters, page 117

Device: Gateway Parameters

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

Figure 77 Device Gateway Cisco Unified CM Administration Window
### Figure 78  Device Gateway Enterprise 1 Branch 1 Enterprise1.com Cisco Unified CM Administration Window

![Cisco Unified CM Administration Window](image)

#### Cisco Unified CM Administration

<table>
<thead>
<tr>
<th>Gateway Configuration</th>
<th>Related Links: Back To Find/List</th>
</tr>
</thead>
<tbody>
<tr>
<td>Status:</td>
<td>Status: Ready</td>
</tr>
</tbody>
</table>

#### Gateway Details

<table>
<thead>
<tr>
<th>Product</th>
<th>Cisco 3645</th>
</tr>
</thead>
<tbody>
<tr>
<td>Gateway</td>
<td>En1_Bri.Ent1.com</td>
</tr>
<tr>
<td>Protocol</td>
<td>MGCP</td>
</tr>
<tr>
<td>Domain Name*</td>
<td>Ent1_Bri.Ent1.com</td>
</tr>
<tr>
<td>Description</td>
<td>Ent1_Bri</td>
</tr>
</tbody>
</table>

#### Configured Slots, VICs and Endpoints

<table>
<thead>
<tr>
<th>Module in Slot 0</th>
<th>&lt; None &gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Module in Slot 1</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Module in Slot 2</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Module in Slot 3</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Module in Slot 4</td>
<td>NEC-HDV2-2PORT-T1</td>
</tr>
<tr>
<td></td>
<td>Subunit 0: VIC2-2FXS</td>
</tr>
<tr>
<td></td>
<td>Subunit 1: &lt; None &gt;</td>
</tr>
</tbody>
</table>

#### Product Specific Configuration Layout

<table>
<thead>
<tr>
<th>Global ISDN Switch Type</th>
<th>&lt;ESS&gt;</th>
</tr>
</thead>
<tbody>
<tr>
<td>Switchback Timing*</td>
<td>Graceful</td>
</tr>
<tr>
<td>Switchback uptime-delay (min)</td>
<td>10</td>
</tr>
<tr>
<td>Switchback schedule (hh:mm)</td>
<td>12:00</td>
</tr>
<tr>
<td>Type Of DTMF Relay*</td>
<td>Current GW Config</td>
</tr>
</tbody>
</table>

---

* - indicates required item.
Figure 79  Device Gateway Enterprise 1 Branch 1 Enterprise 1.com pots 1110 Cisco Unified CM Administration Window
### Cisco Unified CM Administration

#### Directory Number Informations
- **Context Number**: 110
- **Route Plan**
  - Branch 1 Felicity, Vigilant
- **Description**: 110
- **Hosting Name**: 
  - Branch 1 Felicity, Vigilant
- **Associated Devices**
  - 

#### Add Settings
- **Voice Mail**
  - 
- **Additional trunk**: 
  - 
- **AAM Group**: 
  - 

### Call Forward and Call Pickup Settings
- **Display Search Space Activation Rule**
  - 
- **Forward All**
  - 
- **Signalling Calling Search Space for Forward All**
  - 
- **Signalling Calling Search Space for Forward External**
  - 
- **Signalling Calling Search Space for Forward Internal**
  - 
- **Signalling Calling Search Space for Forward Registered Internal**
  - 
- **Signalling Calling Search Space for Forward Unregistered Internal**
  - 
- **Signalling Calling Search Space for Forward Unregistered External**
  - 

### Multi Alternate Party Settings
- **Target (Context)**
  - 
- **MUP Calling Search Space**
  - 
- **MUP No Answer Ring Duration (Seconds)**
  - 

### Line Settings for All Devices
- **Hold Reservation Ring Duration (Seconds)**
  - 
- **Hold Reservation Notification Interval (Seconds)**
  - 

### Line 1 Line Device Aki/1/1/057/1/000109_1_Felicit
- **Display (Optional Caller ID)**
  - 
- **External Phone Number Type**
  - 
- **Extension Calling Hold Settings on Device Aki/1/1/057/1/000109_1_Felicit**
  - Note: The range to avoid the max. number of digits is 1-100
  - 

### Users Associated with Line
- 

---

**Figure 80** Device Gateway Enterprise 1 Branch 1 Enterprise 1.com pots 1110 Line Administration Window

---

*SIP-Based Trunk Managed Voice Services Solution Design and Implementation Guide*
Figure 81  Device Gateway Enterprise 1 HQ VG224 Cisco Unified CM Administration Window

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Gateway Configuration

- Status
  Status: Ready

- Gateway Details
  Product: VG224
  Gateway: 8815400C88972F5
  Protocol: S0CP
  Mac Address (Last 10 Characters)*: 0066382F31
  Description: Ent1-HQ-VG224
  Cisco Unified Communications Manager Group*: Default

- Configured Slots, VICs and Endpoints
  Module in Slot: ANALOG
  | Subunit 1 | 2/0/0 | 2/0/1 | 2/0/2 | 2/0/3 | 2/0/4 | 2/0/5 |
  | Subunit 2 | 2/0/6 | 2/0/7 | 2/0/8 | 2/0/9 | 2/0/10 | 2/0/11 |
  | Subunit 3 | 2/0/12 | 2/0/13 | 2/0/14 | 2/0/15 | 2/0/16 | 2/0/17 |
  | Subunit 4 | 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.
Figure 82  
Device Gateway Enterprise 1 HQ VG224 ANA 1050 Cisco Unified CM Administration Window

**Phone Type:** Analog Phone

**Device Protocol:** SCCP

**Device Information**
- **Registration:** Registered with Cisco Unified Communications Manager 10.4.0.9.2
  - **IP Address:** 10.40.97.254
  - **MAC Address:** 00:06:32:77:06:40
  - **Description:** VG224_ANA 1050
  - **Device Pool:** DevicePool_HQ_Analog_Phone
  - **Common Device Configuration:** HQ_MRG Slave
  - **Phone Button Template:** Standard Analog
  - **Common Phone Profile:** Standard Common Phone Profile
  - **Calling Search Space:** HQ_HW_MRG Slave
  - **Media Resource Group List:** HQ_HW_MRG Slave
  - **Location:** Hub_HQ
  - **User Locale:** HQ_HQ
  - **Network Locale:** HQ_HQ
  - **Device Mobility Mode:** Default
  - **Owner User ID:** HQ

**MLPP Information**
- **MLPP Domain:** None
- **MLPP Indication:** Default
- **MLPP Preemption:** Default

---

* 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 83 Device-Gateway Enterprise 1 HQ VG224 ANA 1050 Line Cisco Unified CM Administration Window
Device: Phone Parameters

To configure the device phone parameters for the Cisco Unified CM, click **Device > Phone** in the Cisco Unified CM Administration window.
Figure 86  Device Phone 4155551170 Cisco Unified CM Administration Window
Figure 87 Device Phone 1170 Cisco Unified CM Administration Window
Device: Trunk Parameters

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

Figure 88  Device Trunk Cisco Unified CM Administration Window

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Calling Search Space</th>
<th>Device Pool</th>
<th>Route Pattern</th>
<th>Partition</th>
<th>Route Group</th>
<th>Priority</th>
<th>Trunk Type</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.10.11.151</td>
<td>Ent1-HQ-CUBE1</td>
<td>CSS-HQ Phones IP</td>
<td>DevicePool-WAN</td>
<td>9.1XXXXXX</td>
<td>Partition-HQ Phones Analog</td>
<td>1</td>
<td>100</td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>10.10.11.151</td>
<td>Ent1-HQ-CUBE1</td>
<td>CSS-HQ Phones IP</td>
<td>DevicePool-WAN</td>
<td>9.1XXXXXX</td>
<td>Partition-HQ Phones IP</td>
<td>1</td>
<td>100</td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>10.60.60.62</td>
<td>Ent1-Bri-CUBE1</td>
<td>CSS-Bri Phones IP</td>
<td>DevicePool-WAN</td>
<td>9.1XXXXXX</td>
<td>Partition-Bri Phones IP</td>
<td>1</td>
<td>100</td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
<tr>
<td>10.60.60.62</td>
<td>Ent1-Bri-CUBE1</td>
<td>CSS-Bri Phones IP</td>
<td>DevicePool-WAN</td>
<td>9.1XXXXXX</td>
<td>Partition-Bri Phones Analog</td>
<td>1</td>
<td>100</td>
<td>SIP Trunk</td>
<td>Non Secure SIP Trunk Profile</td>
</tr>
</tbody>
</table>
Figure 89  Device Trunk Enterprise 1 HQ CUBE1 Phones Analog Cisco Unified CM Administration Window

Cisco Unified CM Administration

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

Trunk Configuration

Status: Ready

Device Information

- Product: SIP Trunk
- Device Protocol: SIP
- Device Name*: 10.10.11.151
- Description: HNT-HQ-CUBE1
- Device Pool*: DevicePool_WAN
- Common Device Configuration: None
- Call Identification*: Use System Default
- Media Resource Group List: HQ-MW-900
- Location*: HQ-HQ-1A
- AAR Group: None
- Packet Capture Mode*: None
- Packet Capture Duration: 0
- Media Termination Point Required: No
- Retry Video Call as Audio: No
- Transmit UTF-8 for Calling Party Name: No
- Unaltered Port: Yes

Multilevel Precedence and Preemption (MLPP) Information

MLPP Domain: None

Cell Routing Information

- Inbound Calls
  - Significant Digits*: 4
  - Connected Line ID Presentation*: Default
  - Connected Name Presentation*: Default
  - Calling Search Space: HQ-HQ-Phones_IP
  - AAR Calling Search Space: None
  - Prefix On: No
  - Redirecting Diversion Header Delivery - Inbound: No

- Outbound Calls
  - Calling Party Selection*: Last Redirect Number (Internal)
  - Calling Line ID Presentation*: Default
  - Calling Name Presentation*: Default
  - Caller ID On: No
  - Caller Name: No
  - Redirecting Diversion Header Delivery - Outbound: No

SIP Information

- Destination Address*: 10.10.11.151
- Destination Address is an SIVR Domain: No
- Destination Port*: 5060
- HTTP Preferred Originating Codec*: G.711
- Presence Group*: Standard Presence group
- SIP Trunk Security Profile*: Non secure SIP Trunk Profile
- Outgoing Calling Search Space: None
- Out-of-Dialing Refer Calling Search Space: None
- SUBSCRIBE Calling Search Space: None
- SIP Profile*: Standard SIP Profile
- DTMF Signaling Method*: Unaltered

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## Enterprise 1 HQ Cisco Unified CM Example Configuration

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

<table>
<thead>
<tr>
<th><strong>Cisco Unified CM Administration</strong></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>System</strong></td>
</tr>
<tr>
<td>---------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Trunk Configuration</strong></td>
</tr>
<tr>
<td>---------------------------------------------------------------</td>
</tr>
<tr>
<td><strong>Status</strong></td>
</tr>
</tbody>
</table>

### Call Routing Information

#### Inbound Calls
- **Significant Digits**: 9
- **Connected Line ID Presentation**: Default
- **Connected Name Presentation**: Default
- **Calling Search Space**: CSS-HQ_Routes_IP
- **AUR Calling Search Space**: <None>
- **Profile ON**: 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 On**: <None>
- **Caller Name**: Redirecting Diversion Header Delivery - Outbound

### SIP Information

#### Destination Address
- **Destination Address**: 10.10.11.151
- **Destination Port**: 5060
- **SIP Trunk Security Profile**: Non Secure SIP Trunk Profile
- **Out-Of-Dialing Referral**: Non Secure SIP Trunk Profile
- **SUBSCRIBER Calling Search Space**: <None>
- **SIP Profile**: Standard SIP Profile
- **DTMF Signaling Method**: No Preference

---

* indicates required item.

** Device reset is not required for changes to Packet Capture Mode and Packet Capture Duration.
### Figure 91 Device Trunk Enterprise 1 Branch 1 CUBE1 Phones Analog Cisco Unified CM Administration Window

#### Device Information
- **Product**: SIP Trunk
- **Device Protocol**: SIP
- **Device IP**: 10.80.80.82
- **Device Name**: Ent1-B1-CUBE1
- **Description**: Ent1-B1-CUBE1
- **Device Pool**: DevicePool_WIN
- **Common Device Configuration**
- **Call Classification**: Use System Default
- **Media Resource Group List**: B1_WFI_RSPL
- **Trunk**: B1
- **AAR Group**: < None >
- **Packet Capture Mode**: None
- **Packet Capture Duration**: 0
- **Media Termination Point Required**: False
- **Video Call as Audio**: False
- **Transmit UTF-8 for Calling Party Name**: False
- **Uncompressed Port**: False

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

#### Call Routing Information

<table>
<thead>
<tr>
<th>Inbound Calls</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Significant Digits*</td>
<td>4</td>
</tr>
<tr>
<td>Connected Line ID Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation*</td>
<td>Default</td>
</tr>
<tr>
<td>Calling Search Space</td>
<td>CSS-B1_Phone_ID_PS</td>
</tr>
<tr>
<td>AAR Calling Search Space</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Prefix DN</td>
<td></td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Inbound</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Outbound Calls</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Party Selection*</td>
</tr>
<tr>
<td>Calling Line ID Presentation*</td>
</tr>
<tr>
<td>Calling Name Presentation*</td>
</tr>
<tr>
<td>Caller ID DN</td>
</tr>
<tr>
<td>Caller Name</td>
</tr>
<tr>
<td>Redirecting Diversion Header Delivery - Outbound</td>
</tr>
</tbody>
</table>

#### SIP Information

<table>
<thead>
<tr>
<th>Destination Address*</th>
</tr>
</thead>
<tbody>
<tr>
<td>10.80.80.82</td>
</tr>
<tr>
<td>Destination Port*</td>
</tr>
<tr>
<td>5060</td>
</tr>
</tbody>
</table>

#### Additional Information

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

---

SIP-Based Trunk Managed Voice Services Solution Design and Implementation Guide
Figure 92  Device Trunk Enterprise 1 Branch 1 CUBE1 Phones IP Cisco Unified CM Administration Window

Cisco Unified CM Administration

Trunk Configuration

Status

---

Device Information

- Product: SIP Trunk
- Device Protocol: SIP
- Device Name: 10.80.80.82
- Description: Ent-1-B1-CUBE1
- Device Pool: DevicePool_WON
- Common Device Configuration: < None >
- Call Classification: CUC System Default
- Media Resource Group List: B11-HTTP-MRG
- Location: Trunk B1
- AAR Group: < None >
- Packet Capture Mode: None
- Packet Capture Duration: <>

Multilevel Precedence and Preemption (MPLP) Information

- MLPP Domain: < None >

Call Routing Information

---

Inbound Calls

- Significant Digits: <>
- Connected Line ID Presentation: Default
- Connected Name Presentation: Default
- Calling Search Space: CSS-B11-Phones_IP
- AUR Calling Search Space: < None >
- Premis DN: <>

Outbound Calls

- Calling Party Selection: Originator
- Calling Line ID Presentation: Default
- Calling Name Presentation: Default
- Caller ID DN: <>
- Caller Name: <>

SIP Information

- Destination Address: 10.80.80.82
- Destination Address is an SRV
- Destination Port: <>
- MTP Preferred Originating Codes: <>
- Presence Group: Standard Presence group
- SIP Trunk Security Profile: Non Secure SIP Trunk Profile
- Redirecting Calling Search Space: < None >
- OK-CSCF Allocated Calling Search Space: < None >
- SUBSCRIBER 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.
Enterprise 1 HQ Cisco Unity and Cisco Unity Express Example Configuration

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

Enterprise 1 HQ and Cisco VG224 Analog Phone Gateway Example Configuration

The following example shows a CLI configuration for the enterprise 1 HQ the Cisco VG224 Analog Phone Gateway for the test topology described in Figure 7.

```
Ent1_HQ_VG224#  
!  
stcapp ccm-group 1  
stcapp  
!  
voice service voip  
fax protocol pass-through g711ulaw  
modem passthrough nse codec g711ulaw  
!  
interface FastEthernet0/0  
ip address 10.40.97.254 255.255.0.0  
load-interval 30  
duplex full  
speed 100  
!  
interface FastEthernet0/1  
no ip address  
shutdown  
duplex auto  
speed auto  
!  
interface FastEthernet0/1  
no 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
```
Enterprise 1 HQ Cisco ASA Firewall Example Configuration

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

```
Ent1-HQ-ASA#
interface Vlan65
  nameif inside
  security-level 100
  ip address 10.40.99.1 255.255.255.0

interface Vlan70
  nameif outside
  security-level 0
  ip address 10.40.98.2 255.255.255.0

interface Ethernet0/0
  description *** To WAN ***
  switchport access vlan 70

interface Ethernet0/1
  description *** To LAN ***
  switchport access vlan 65

ftp mode passive
access-list 100 extended permit icmp any any
access-list 100 extended permit icmp any any echo
access-list 100 extended permit icmp any any echo-reply
access-list 100 extended permit tcp any host 40.40.97.2 eq 2000
access-list 100 extended permit udp any host 40.40.97.2 eq sip
access-list 100 extended permit tcp any host 40.40.97.2 range h323 h323
access-list 100 extended permit tcp any host 10.10.11.151 eq 5090
access-list 100 extended permit udp any host 10.10.11.151 eq 5090
access-list 100 extended permit tcp any host 40.40.97.2 eq 2428
access-list 100 extended permit udp any host 40.40.97.2 eq 2427
pager lines 24
logging enable
logging buffered debugging
logging asdm informational
mtu inside 1500
mtu outside 1500
icmp unreachable rate-limit 1 burst-size 1
asdm image disk0:/asdm-524.bin
no asdm history enable
arp timeout 14400
access-group 100 in interface outside

timeout xlate 3:00:00
timeout conn 1:00:00 half-closed 0:10:00 udp 0:02:00 icmp 0:00:02
timeout sunrpc 0:10:00 h323 0:05:00 h225 1:00:00 mgcp 0:05:00 mgcp-pat 0:05:00
timeout sip 0:30:00 sip_media 0:02:00 sip-invite 0:03:00 sip-disconnect 0:02:00
timeout sip-provisional-media 0:02:00 uauth 0:05:00 absolute
http server enable
no snmp-server location
no snmp-server contact
snmp-server enable traps snmp authentication linkup linkdown coldstart
```
Branch 1 Cisco UBE, TDM Gateway, and Cisco Unified SRST Example Configuration

The following example shows a CLI configuration for the branch 1 Cisco Unified Border Element, TDM Switching in the Cisco AS5000 Gateway, and Cisco Unified SRST for the test topology described in Figure 7.

Ent1_Br1#
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
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
call-agent 10.40.97.2 2427 service-type mgcp version 0.1
dtmf-relay voip codec all mode out-of-band
sdp simple
fax t38 inhibit
call control source-interface GigabitEthernet0/1.1
call media source-interface GigabitEthernet0/1.1
mgcp profile default
ccp local GigabitEthernet0/1.1
ccp ccm 10.40.97.2 identifier 1 priority 1 version 6.0
ip precedence 3
ccp
ccp 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
Branch 1 Cisco Unity Express 3.2 and Cisco Unified CM Example Configuration

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

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

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