



Cisco Unified CM Trunks

Revised: April 2, 2013; OL-29367-04

A trunk is a communications channel on Cisco Unified Communications Manager (Unified CM) that enables Unified CM to connect to other servers. Using one or more trunks, Unified CM can receive or place voice, video, and encrypted calls, exchange real-time event information, and communicate in other ways with call control servers and other external servers.

Trunks are an integral and a crucial part of a Cisco Unified Communications deployment, hence it is important to understand the types of trunks available, their capabilities, and design and deployment considerations such as resiliency, capacity, load balancing, and so forth.

There are two basic types of trunks that can be configured in Unified CM:

- SIP and H.323 trunks, both of which can be used for external communications
- Intercluster trunks (ICTs)

This chapter describes the general capabilities and functions of these trunks. Additional discussion on specific applications of Unified CM trunks can be found in other pertinent chapters of this document.

This chapter discusses the following topics:

- [A Comparison of SIP and H.323 Trunks, page 6-3](#)
- [SIP Trunks Overview, page 6-6](#)
- [H.323 Trunks Overview, page 6-38](#)
- [General SIP and H.323 Trunk Design Considerations, page 6-57](#)
- [IP PSTN and IP Trunks to Service Provider Networks, page 6-64](#)
- [Trunk Aggregation Platforms, page 6-65](#)

For more details on the applications of Unified CM trunks, refer to their respective sections in the following chapters:

- [Collaboration Deployment Models, page 10-1](#)
- [Media Resources, page 7-1](#)
- [Call Admission Control, page 13-1](#)
- [Cisco IM and Presence, page 20-1](#)

What's New in This Chapter

Table 6-1 lists the topics that are new in this chapter or that have changed significantly from previous releases of this document.

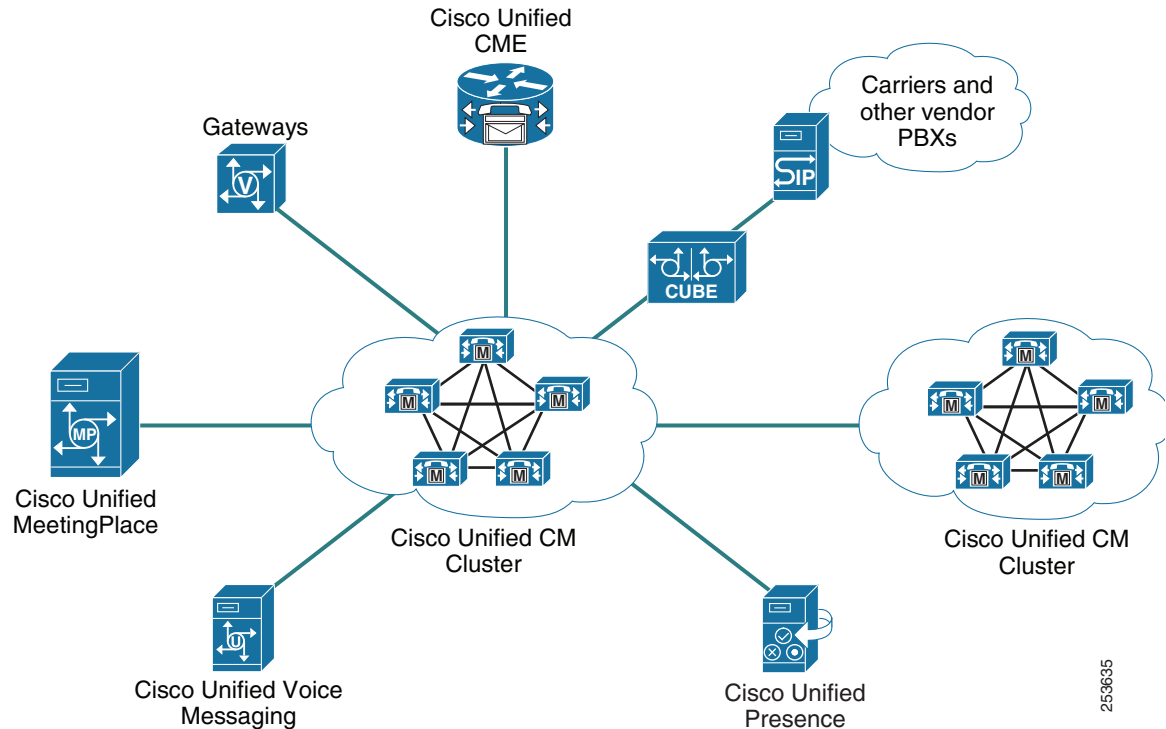
Table 6-1 *New or Changed Information Since the Previous Release of This Document*

New or Revised Topic	Described in:	Revision Date
Integration of Unified CM with Cisco TelePresence Video Communication Server (VCS)	Interconnecting Unified CM and Cisco VCS, page 6-63	April 2, 2013
Clustering over the WAN with Unified CM Session Management Edition Cluster Trunks	Design Guidance for Clustering over the WAN with Unified CM Session Management Edition Cluster Trunks, page 6-37	August 31, 2012
Audio codec preference lists	Audio Codec Preference (Accept Audio Codec Preference in Received Offer), page 6-11 Accept Audio Codec Preferences in Received Offer, page 6-60	June 28, 2012
Normalization and transparency scripts	Normalization and Transparency Scripts in Unified CM, page 6-15	June 28, 2012
Other minor updates for Cisco Unified Communications System Release 9.0	Various sections throughout this chapter	June 28, 2012

Unified CM Trunks Solution Architecture

Unified CM uses the mechanism of IP trunks to exchange call-related information with other components of a Unified Communications solution. Given their importance in this respect, it is important to develop the system architecture of the IP trunks with proper regard to the protocol, feature and service expectations, performance requirements, and so forth.

Figure 6-1 illustrates the role of IP trunks in system connectivity. The illustration does not show all possible connections from the Unified CM cluster.

Figure 6-1 *IP Trunks Provide Connections to Unified CM*

Calls are directed toward trunks as defined by the dial plan using the route pattern construct. A route pattern can use a trunk either directly or through a route list. The route list, if used, consists of one or more route groups, each of which contains one or more trunks. An individual trunk within a route group may be configured to be selected in either a top-down or circular fashion. For outgoing calls, Unified CM selects one of the trunks associated in this fashion with the route pattern. Before it accepts an incoming call, Unified CM verifies whether a trunk is defined to the remote address from which the call is received.

A Comparison of SIP and H.323 Trunks

Cisco Unified CM trunk connections support both SIP and H.323. In many cases, the decision to use SIP or H.323 is driven by the unique feature(s) offered by each protocol. There are also a number of external factors that can affect the choice of trunk protocol, such as customer preference or the protocol's maturity and degree of interoperability offered between various vendors' products.

For trunk connections between Cisco devices, this decision is relatively straightforward. For trunk connections to other vendors' products and to service provider networks, it is important to understand which features are required by the customer and the extent of interoperability between any two vendors' products.

Table 6-2 compares some of the features offered over SIP and H.323 trunks between Unified CM clusters.

Table 6-2 Comparison of SIP and H.323 Features on Cisco Unified CM Trunks

Feature	SIP	QSIG over SIP	H.323	QSIG over H.323
Calling Line (Number) Identification Presentation	Yes	Yes	Yes	Yes
Calling Line (Number) Identification Restriction	Yes	Yes	Yes	Yes
Calling Name Identification Presentation	Yes	Yes	Yes	Yes
Calling Name Identification Restriction	Yes	Yes	Yes	Yes
Connected Line (Number) Identification Presentation	Yes	Yes	Yes	Yes
Connected Line (Number) Identification Restriction	Yes	Yes	Yes	Yes
Connected Name Identification Presentation	Yes	Yes	Yes	Yes
Connected Name Identification Restriction	Yes	Yes	Yes	Yes
Alerting Name	Yes	Yes	No	Yes
Call Transfer (Blind/Attended)	Yes/Yes	Yes/Yes	Yes/Yes	Yes/Yes
Call Forward All	Yes	Yes	Yes	Yes
Call Forward Busy	Yes	Yes	Yes	Yes
Call Forward No Reply	Yes	Yes	Yes	Yes
Call Completion to Busy Subscriber	No	Yes	No	Yes
Call Completion No Reply	No	Yes	No	Yes
Subscribe/Notify, Publish – Presence	Yes	Yes	No	No
Message Waiting Indication (MWI: lamp ON, lamp OFF)	Yes	Yes	No	Yes
Path Replacement	No	Yes	No	Yes
Call Hold/Resume	Yes	Yes	Yes	Yes
Music On Hold (unicast and multicast)	Yes	Yes	Yes	Yes
DTMF-relay	RFC 2833, KPML (OOB), Unsolicited Notify (OOB)	RFC 2833, KPML (OOB), Unsolicited Notify (OOB)	H.245 Out Of Band (OOB) ¹	H.245 Out Of Band (OOB) ¹
SIP Early Offer	Yes – MTP may be required	Yes – MTP may be required	N/A	N/A
SIP Delayed Offer	Yes	Yes	N/A	N/A
H.323 Fast Start	N/A	N/A	Yes – MTP always required for Outbound Fast Start	Yes – MTP always required for Outbound Fast Start
H.323 Slow Start	N/A	N/A	Yes	Yes
Audio codecs	G.711, G.722, G.723, G.729, iLBC, AAC, iSAC	G.711, G.722, G.723, G.729, iLBC, AAC, iSAC	G.711, G.722, G.723, G.729	G.711, G.722, G.723, G.729

Table 6-2 Comparison of SIP and H.323 Features on Cisco Unified CM Trunks (continued)

Feature	SIP	QSIG over SIP	H.323	QSIG over H.323
Accept Audio Codec Preference in Received Offer	Yes	Yes	No	No
Codecs with MTP	All codecs supported when Early Offer support for voice and video calls (insert MTP if needed) is checked G.711, G.729 when MTP Required is checked	All codecs supported when Early Offer support for voice and video calls (insert MTP if needed) is checked G.711, G.729 when MTP Required is checked	G.711, G.723, G.729	G.711, G.723, G.729
Video	Yes	Yes	Yes	Yes
Video codecs	H.261, H.263, H.263+, H.264 AVC	H.261, H.263, H.263+, H.264 AVC	H.261, H.263, H.263+, H.264 AVC	H.261, H.263, H.263+, H.264 AVC
T.38 Fax	Yes	Yes	Yes	Yes
Signaling Authentication	Digest, TLS	Digest, TLS	No	No
Signaling Encryption	TLS	TLS	No	No
Media Encryption (audio)	SRTP	SRTP	SRTP	SRTP
RSVP-based QoS and call admission control	Yes	Yes	No	No
Support for + character	Yes	Yes	No	No
Inbound Calls — Called Party: Significant Digits, Prefix-Digits	Yes	Yes	Yes	Yes
Incoming Calling Party Settings: Strip Digits, Prefix-Digits based on Number Type	SIP does not support Number Type - "Unknown" used for all calls	SIP does not support Number Type - "Unknown" used for all calls	Unified CM, Unknown, National, International, Subscriber	Unified CM, Unknown, National, International, Subscriber
Incoming Called Party Settings: Strip Digits, Prefix-Digits based on Number Type	N/A	N/A	Unified CM, Unknown, National, International, Subscriber	Unified CM, Unknown, National, International, Subscriber
Connected Party Transformation	Yes	Yes	No	No
Outbound Calling Party Transformations	Yes	Yes	Yes	Yes
Outbound Called Party Transformations	Yes	Yes	Yes	Yes

Table 6-2 Comparison of SIP and H.323 Features on Cisco Unified CM Trunks (continued)

Feature	SIP	QSIG over SIP	H.323	QSIG over H.323
Outbound Calling/Called Party Number Type Setting	SIP does not support Number Type	SIP does not support Number Type	Unified CM, Unknown, National, International, Subscriber	Unified CM, Unknown, National, International, Subscriber
Outbound Called/Called Party Numbering Plan Setting	SIP does not support Number Plan	SIP does not support Number Plan	Unified CM, ISDN, National Standard, Private, Unknown	Unified CM, ISDN, National Standard, Private, Unknown
Trunk destination — State detection mechanism	OPTIONS Ping	OPTIONS Ping	Per call attempt	Per call attempt

1. H.323 trunks support signaling of RFC 2833 for certain connection types.

SIP Trunks Overview

SIP trunks provide connectivity to other SIP devices such as gateways, Cisco Unified CM Session Management Edition, SIP proxies, Unified Communications applications, and other Unified CM clusters. Today, SIP is arguably the most commonly chosen protocol when connecting to service providers and Unified Communications applications. Cisco Unified CM provides the following SIP trunk and call routing enhancements:

- Can run on all Unified CM nodes
- Up to 16 destination IP addresses per trunk
- SIP OPTIONS ping keepalives
- SIP Early Offer support for voice and video calls (insert MTP if needed)
- Audio codec preference (Accept Audio Codec Preference in Received Offer)
- QSIG over SIP
- SIP trunk normalization and transparency
- SIP REFER transparency
- Supports the use of route lists on all Unified CM nodes

The SIP trunk features available in the current release of Unified CM make SIP the preferred choice for new and existing trunk connections. The QSIG over SIP feature provides parity with H.323 intercluster trunks and can also be used to provide QSIG over SIP trunk connections to Cisco IOS gateways (and on to QSIG-based TDM PBXs). The ability to run on all Unified CM nodes and to handle up to 16 destination IP addresses improves outbound call distribution from Unified CM clusters and reduces the number of SIP trunks required between clusters and devices. SIP OPTIONS ping provides dynamic reachability detection for SIP trunk destinations, rather than per-call reachability determination. SIP Early Offer support for voice and video calls (insert MTP if needed) can reduce or eliminate the need to use MTPs and allows voice, video, and encrypted calls to be made over SIP Early Offer trunks.

SIP trunk normalization and transparency improve native Unified CM interoperability with and between third-party unified communications systems. Normalization allows inbound and outbound SIP messages and SDP information to be modified on a per-SIP-trunk basis. Transparency allows Unified CM to pass SIP headers, parameters, and content bodies from one SIP trunk call leg to another, even if Unified CM does not understand or support the parts of the message that are being passed through.

These features are discussed in detail later in this section.

For the complete list of new enhancements for SIP trunks, refer to the Cisco Unified Communications Manager product release notes available at

http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_release_notes_list.html

General Deployment Considerations

Unified CM SIP trunks offer a greater set of features in comparison with H.323 intercluster trunks, thus making SIP the protocol of choice for intercluster trunk connections (although H.323 Annex M1 may still be preferred for intercluster trunk connections to Unified CM clusters using earlier software versions). Also, given the wide support of SIP in the industry, SIP trunks are usually a good choice for connectivity to third-party applications and service providers.

SIP Trunk Features and Operation

This section explains how Unified CM SIP trunks operate and describes several key SIP trunk features that should be taken into account when designing and deploying Unified CM SIP trunks.

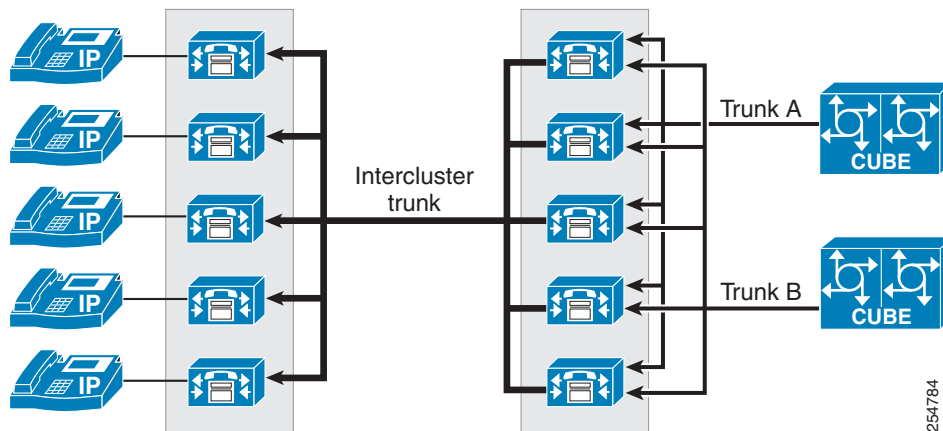
SIP Trunks Can Run on All Active Unified CM Nodes

When the **Run on all Active Unified CM Nodes** option is checked on a SIP trunk, Unified CM creates an instance of the SIP trunk daemon on every call processing subscriber within the cluster, thus allowing SIP trunk calls to be made or received on any call processing subscriber. (Prior to this feature, up to three nodes could be selected per trunk by using Unified CM Groups.) With **Run on all Active Unified CM Nodes** enabled, outbound SIP trunk calls originate from the same node on which the inbound call (for example, from a phone or trunk) is received. As with all Unified CM SIP trunks, the SIP daemons associated with the trunk will accept inbound calls only from end systems with IP addresses that are defined in the trunk's destination address fields. Running SIP trunks on all nodes is recommended where the SIP trunk is required to process a large number of calls so that outbound and inbound call distribution can be evenly spread across all call processing subscribers within a cluster. Also, when multiple SIP trunks to the same destination(s) are using the same subscriber, a unique incoming and destination port number must be defined per trunk to allow each trunk to be identified uniquely.

Up to 16 SIP Trunk Destination IP Addresses

SIP trunks can be configured with up to 16 destination IP addresses, 16 fully qualified domain names, or a single DNS SRV entry. Support for additional destination IP addresses reduces the need to create multiple trunks associated with route lists and route groups for call distribution between two Unified Communications systems, thus simplifying Unified CM trunk design. (See [Figure 6-2](#).) This feature can be used in conjunction with the **Run on all Active Unified CM Nodes** feature or with a SIP trunk that uses standard Unified CM Groups to create a SIP daemon on up to three nodes within the cluster. Bear in mind, however, that the SIP daemons associated with a Unified CM SIP trunk will accept inbound calls only from end systems with IP addresses that are defined in the trunk's destination address fields.

Figure 6-2 *SIP Trunks with Multiple Destination IP Addresses Running on All Active Nodes*



SIP OPTIONS Ping

The SIP OPTIONS Ping feature can be enabled on the SIP Profile associated with a SIP trunk to dynamically track the state of the trunk's destination(s). When this feature is enabled, each node running the trunk's SIP daemon will periodically send an OPTIONS Request to each of the trunk's destination IP addresses to determine its reachability and will send calls only to reachable nodes. A destination address is considered to be "out of service" if it fails to respond to an OPTIONS Request, if it sends a Service Unavailable (503) response or Request Timeout (408) response, or if a TCP connection cannot be established. The trunk state is considered to be "in service" when at least one node receives a response (other than a 408 or 503) from a least one destination address. SIP trunk nodes can send OPTIONS Requests to the trunk's configured destination IP addresses or to the resolved IP addresses of the trunk's DNS SRV entry. Enabling SIP OPTIONS Ping is recommended for all SIP trunks because it allows Unified CM to dynamically track trunk state rather than determining trunk state on a per-call and timeout basis.

SIP Early Offer Support over Unified CM SIP Trunks

SIP negotiates media exchange by means of the Session Description Protocol (SDP), where one side offers a set of capabilities to which the other side answers, thus converging on a set of media characteristics. SIP allows the initial offer to be sent either by the caller in the initial INVITE message (Early Offer) or, if the caller chooses not to, the called party can send the initial offer in the first reliable response (Delayed Offer).

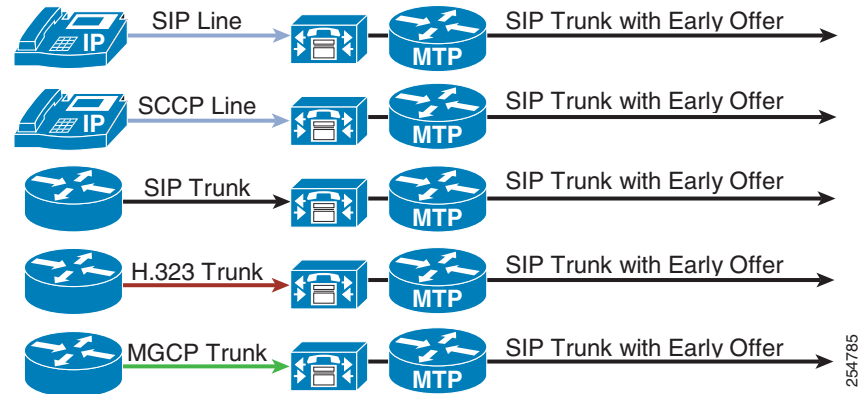
By default, Unified CM SIP trunks send the INVITE without an initial offer (Delayed Offer). In general SIP Delayed Offer is preferred for Unified CM SIP trunks because MTPs are not needed to establish a Delayed Offer call for voice, video, or encrypted media. If SIP Early Offer is desired, Unified CM has two configurable options to enable a SIP trunk to send the offer in the INVITE:

- [Media Termination Point Required](#), page 6-9
- [Early Offer Support for Voice and Video Calls \(Insert MTP If Needed\)](#), page 6-9

Media Termination Point Required

Enabling the **Media Termination Point Required** option on the SIP trunk assigns an MTP from the trunk's media resources group (MRG) to every outbound call. (See [Figure 6-3](#).) This statically assigned MTP supports only the G.711 or G.729 codecs, thus limiting media to voice calls only.

Figure 6-3 *SIP Early Offer with Media Termination Point Required*



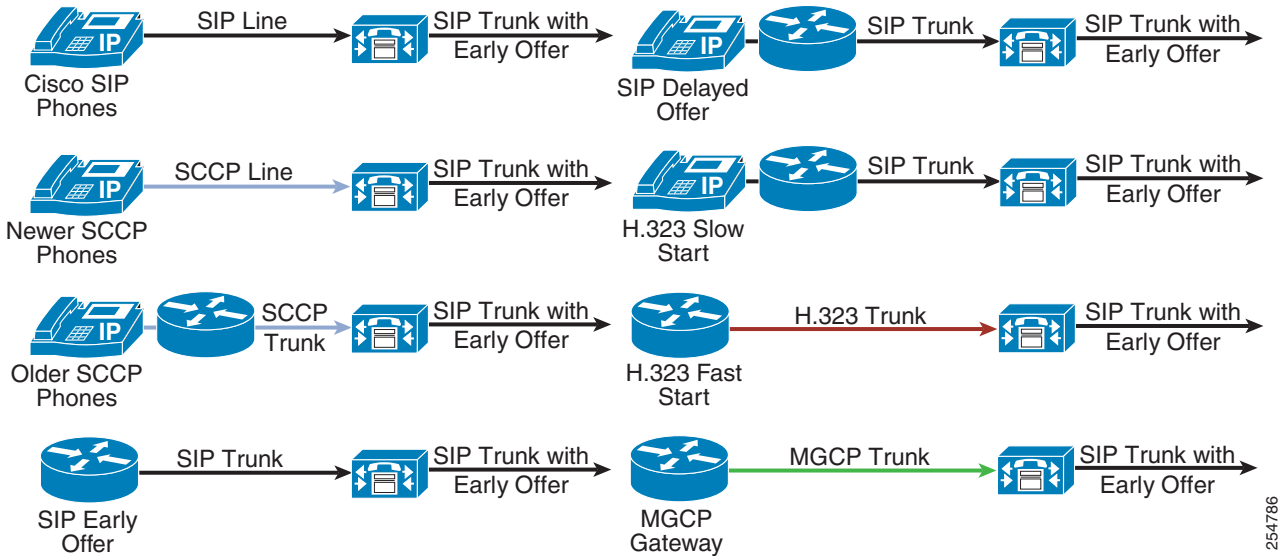
Early Offer Support for Voice and Video Calls (Insert MTP If Needed)

Enabling **Early Offer support for voice and video calls (insert MTP if needed)** on the SIP Profile associated with the SIP trunk inserts an MTP only if the calling device cannot provide Unified CM with the media characteristics required to create the Early Offer. In general, **Early Offer support for voice and video calls (insert MTP if needed)** is recommended over **Media Termination Point Required** because this configuration option reduces MTP usage (see [Figure 6-4](#)). Calls from older SCCP-based phones registered to Unified CM over SIP Early Offer trunks configured with this option will use an MTP to create the Offer SDP, and these calls support voice, video, and encrypted media. Inbound calls to Unified CM from SIP Delayed Offer trunks or H.323 Slow Start trunks that are extended over an outbound SIP Early Offer trunk will use an MTP to create the Offer SDP; however, these calls support audio only in the initial call set up but can be escalated mid-call to support video and SRTP if the call media is renegotiated (for example, after hold/resume). For guidance on when to use **Early Offer support for voice and video calls (insert MTP if needed)**, see [Design Considerations for SIP Trunks](#), page 6-29.



Note

MTP resources are not required for incoming INVITE messages, whether or not they contain an initial offer SDP.

Figure 6-4 Early Offer Support for Voice and Video Calls

Unified CM does not need to insert an MTP to create an outbound Early Offer call over a SIP trunk if the inbound call to Unified CM is received by any of the following means:

- On a SIP trunk using Early Offer
- On an H.323 trunk using Fast Start
- On an MGCP trunk
- From a SIP-based IP phone registered to Unified CM
- From newer SCCP-based Cisco Unified IP Phone models registered to Unified CM

For the above devices, Unified CM uses the media capabilities of the endpoint and applies the codec filtering rules based on the region-pair of the calling device and outgoing SIP trunk to create the offer SDP for the outbound SIP trunk call. In most cases, the offer SDP will have the IP address and port number of the endpoint initiating the call. This is assuming that Unified CM does not have to insert an MTP for other reasons such as a DTMF mismatch, TRP requirements, or a transcoder requirement when there is no common codec between the regions of the calling device and the SIP trunk.

When **Early Offer support for voice and video calls (insert MTP if needed)** is configured on a trunk's SIP Profile, calls from older SCCP-based phones, SIP Delayed Offer trunks, and H.323 Slow Start trunks will cause Unified CM to allocate an MTP if an MTP or transcoder has not already been allocated for that call for another reason. The MTP is used to generate an offer SDP with a valid media port and IP address. The MTP will be allocated from the media resources associated with the calling device rather than from the outbound SIP trunk's media resources. (This prevents the media path from being anchored to the outbound SIP trunk's MTP). If the MTP cannot be allocated from the calling device's media resource group list (MRGL), then the MTP allocation is attempted from the SIP trunk's MRGL.

For calls from older SCCP phones registered to Unified CM, some of the media capabilities of the calling device (for example, supported voice codecs, video codecs, and encryption keys if supported) are available for media exchange through the Session Description Protocol (SDP). Unified CM will create a superset of the endpoint and MTP codec capabilities and apply the codec filtering based on the applicable region-pair settings. The outbound Offer SDP will use the MTP's IP address and port number and can support voice, video, and encrypted media. Note that the MTP should be configured to support the pass-through codec.

**Note**

Older SCCP-based IP phones such as the Cisco Unified IP Phone 7902, 7905, 7910, 7912, 7920, 7935, 7940, and 7960 require the use of an MTP when they make calls over a SIP trunk with the **Early Offer for voice and video (insert MTP if needed)** feature enabled. If you have a significant number of these phone types deployed in a cluster, provision MTP resources in the cluster equivalent to the number of busy hour calls over those SIP trunks using the **Early Offer for voice and video (insert MTP if needed)** feature.

When Unified CM receives an inbound call on an H.323 Slow Start or SIP Delayed Offer trunk, the media capabilities of the calling device are not available when the call is initiated. In this case, Unified CM must insert an MTP and will use its IP address and UDP port number to advertise all supported audio codecs (after region pair filtering) in the Offer SDP of the initial INVITE sent over the outbound SIP trunk. When the Answer SDP is received on the SIP trunk, if it contains a codec that is supported by the calling endpoint, then no additional offer-answer transaction is needed. In case of codec mismatch, Unified CM can either insert a transcoder to address the mismatch or send a reINVITE or UPDATE to trigger media negotiation. Calls from H.323 Slow Start or SIP Delayed Offer trunks support audio only in the initial call setup, but they can be escalated mid-call to support video and SRTP if the call media is renegotiated (for example, after Hold/Resume).

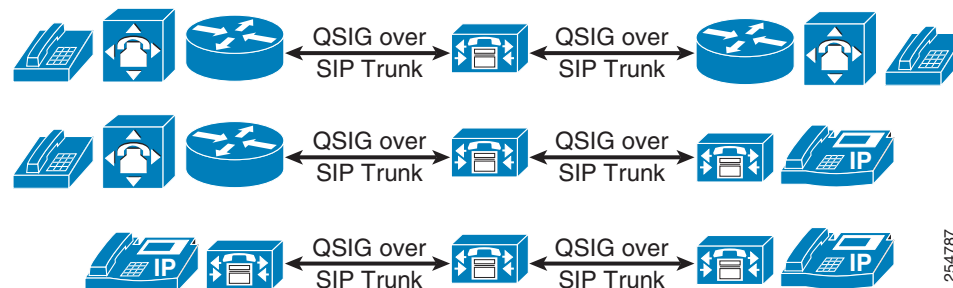
Audio Codec Preference (Accept Audio Codec Preference in Received Offer)

Cisco Unified CM 9.x provides configurable audio codec preference lists, which can be used to prioritize codec preferences for calls within regions, between regions, and between clusters. For calls over SIP trunks, the configurable SIP Profile option **Accept Audio Codec Preference in Received Offer** allows the trunk to override the codec preferences configured for the trunk's region or region pair and to use the codec preferences received in an Offer from an off-cluster device. This feature is particularly useful in deployments where a SIP call passes through two or more Unified CM clusters (for example, in SME deployments where the end codec preference needs to be preserved for a call). Codec preference is discussed in detail in the section on [Codec Selection Over IP Trunks](#), page 6-58.

QSIG over SIP Trunks

Unified CM can encapsulate QSIG content in SIP messages, thus allowing features such as Call Back, MWI, and Path Replacement to be invoked over SIP QSIG intercluster trunks and over SIP QSIG trunks to Cisco IOS gateways. (See [Figure 6-5](#).) QSIG over SIP trunks provides parity with the QSIG feature set on H.323 Annex M1 intercluster trunks and MGCP QSIG trunks. (ISO and ECMA variants of QSIG are supported on a per-trunk basis.)

Figure 6-5 QSIG over SIP Trunks



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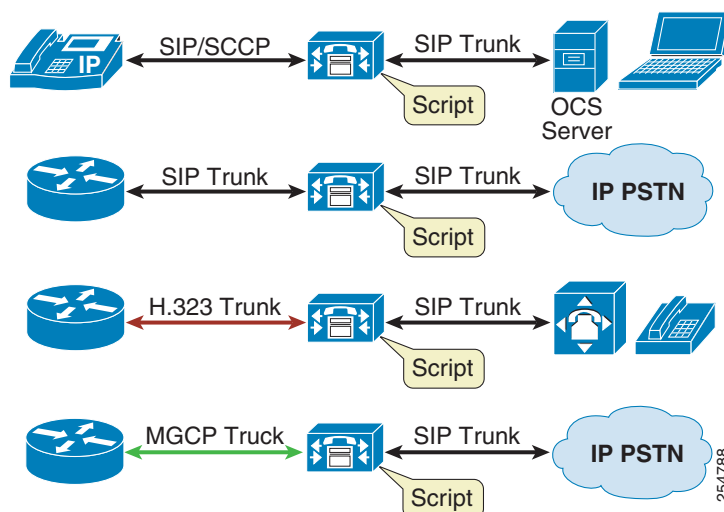
SIP Trunk Message Normalization and Transparency

Normalization and transparency provide powerful script-based functionality for SIP trunks that can be used to transparently forward and/or modify SIP messages and message body contents as they traverse Unified CM. Normalization and transparency scripts are designed to address SIP interoperability issues, allowing Unified CM to interoperate with SIP-based third-party PBXs, applications, and IP PSTN services.

SIP Trunk Normalization

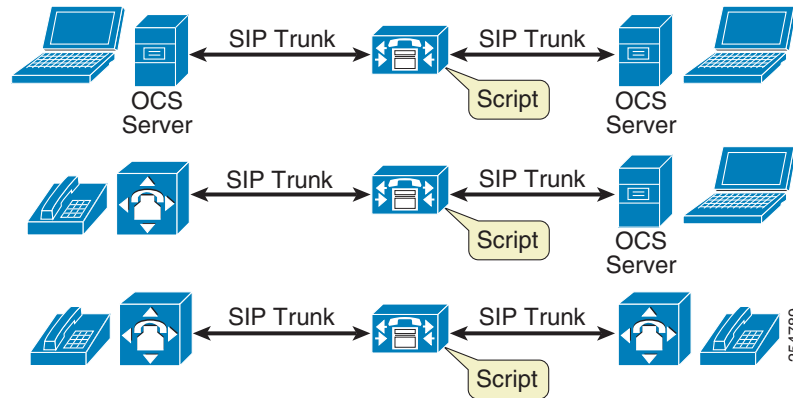
Normalization allows incoming and outgoing SIP messages to be modified on their way through Unified CM. Normalization applies to all calls that traverse a SIP trunk with an associated script, regardless of what protocol is being used for the other endpoint involved in the call. For example, a SIP trunk normalization script can operate on a call from a SIP line device to a SIP trunk, from an SCCP-based device to a SIP trunk, from MGCP to SIP trunk, from H.323 to SIP trunk, and so forth. (See [Figure 6-6](#).) Normalization does not require end-to-end SIP.

Figure 6-6 SIP Trunk Normalization



SIP Trunk Transparency

Transparency allows Unified CM to pass SIP headers, parameters, and content bodies from one SIP trunk call leg to another, even if Unified CM does not understand or support the parts of the message that are being passed through. Transparency (or transparent pass-through) is applicable only when the call through Unified CM is from SIP trunk to SIP trunk, as illustrated in [Figure 6-7](#).

Figure 6-7 SIP Trunk Transparency

Normalization and transparency scripts use Lua, a powerful, fast, lightweight, embeddable scripting language to modify SIP messages and SDP body content on SIP trunks. (For more information on Lua, refer to the documentation available at <http://lua-users.org/wiki/LuaOrgGuide>.)

Cisco has created a library of Lua-based SIP Message APIs that allow specified information in the SIP message and SDP body to be retrieved, modified, replaced, removed, passed through, ignored, appended to, transformed, and so on. The underlying Lua language allows retrieved information to be stored as variables and operated on using a series of operations such as: If, elseif, while, do, <, >, =, and so forth. The scripting approach naturally supports multiple variables and state-specific contexts for making script decisions. The combination of Cisco's SIP Message Library APIs and the functionality underlying the Lua language creates a very powerful scripting environment that allows almost any SIP message and/or its SDP body content to be modified.

For inbound messages on a SIP trunk, normalization and transparency script processing occurs immediately after receiving the message from the network. For outbound messages, script processing occurs immediately before sending the message to the network.

Within a Lua script, callback functions (also known as message handlers) are used to request message types of interest. The Cisco Lua environment constructs the name of the message handler based on the message direction and method for requests (for example, `inbound_INVITE`) and based on the message direction, response code, and method (from the CSeq header) for responses (for example, `outbound_180_INVITE`). A message object (for example, `msg`) is passed to the message handler, thereby allowing the script to modify the message (for example, `inbound_INVITE(msg)`).

Callback Function (message Handler) examples:

<code>inbound_INVITE()</code>	<code>outbound_INVITE()</code>
<code>inbound_UPDATE()</code>	<code>outbound_SUBSCRIBE()</code>
<code>inbound_3XX_INVITE()</code>	<code>outbound_180_INVITE()</code>

The Lua script then uses APIs defined in the Cisco SIP Message library to access and manipulate message parameters. For example:

- **getHeader**(*header-name*) returns header-value or ""
- **getHeaderValues**(*header-name*) returns a table of header values
- **addHeaderValueParameter**(*header-name*, *parameter-name*, [*parameter-value*])
- **getUri**(*header-name*) retrieves the URI from the specified header

- **block()** blocks the specified SIP message
- **applyNumberMask(*header-name*, *mask*)** retrieves the specified header and applies the specified number mask to the URI
- **getSdp()** returns the SDP content
- **sdp:getLine(*start of line*, *line contains*)** returns line in SDP that starts with "start of line" and also has string "line contains"
- **sdp:modifyLine(*start of line*, *line contains*, *new-line*)** finds the in SDP that starts with "start of line", the line matching "line contains" is replaced with the *new-line* parameter

The following examples illustrate the use of SIP Message API scripts.

Example 6-1 SIP Message API — getRequestLine

getRequestLine() returns the method, request-uri, and version.

This method returns three values:

- The method name
- The request-uri
- The protocol version

Example script:

Line 1	M = { }
Line 2	function M.outbound_INVITE(message)
Line 3	local method, ruri, ver = message:getRequestLine()
Line 4	end
Line 5	return M

Line 1 initializes the set of callback functions to an empty value. This set of callback functions, named M, is essentially a Lua table.

Lines 2 to 4 define a message handler. This callback function is executed when an outbound INVITE is sent from Unified CM. The script then gets the method, request-uri, and version from the request line and stores these values.

The script can define multiple message handlers. The name of the message handler dictates which message handler is invoked (if any) for a given SIP message.

The last line returns the set of callbacks. This line is absolutely required.

Message:

```
INVITE sip:1234@10.10.10.1 SIP/2.0
```

Output and result:

```
method == "INVITE"
ruri == "sip:1234@10.10.10.1"
version == "SIP/2.0"
```

Example 6-2 A script that simply removes the "Cisco-Guid" header in an outbound INVITE

Line 1	M = { }
Line 2	function M.outbound_INVITE(message)
Line 3	message:removeHeader("Cisco-Guid")
Line 4	end
Line 5	return M

Line 1 initializes the set of callback functions to an empty value. This set of callback functions, named M, is essentially a Lua Table.

Lines 2 to 4 define a message handler. This callback function is executed when an outbound INVITE is sent from Unified CM. The script can define multiple message handlers. The name of the message handler dictates which message handler is invoked (if any) for a given SIP message.

The last line returns the set of callbacks. This line is absolutely required.

Message:

```
INVITE sip:1234@10.10.10.1 SIP/2.0
.
P-Asserted-Identity: "1234" <1234@10.10.10.1>
Cisco-Guid: 1234-4567-1234
Session-Expires: 1800
```

Output and results:

```
INVITE sip:1234@10.10.10.1 SIP/2.0
.
P-Asserted-Identity: "1234"
```

For more information on SIP trunk normalization and transparency scripts, refer to the latest version of the *Developer Guide for SIP Transparency and Normalization*, available at

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_programming_reference_guides_list.html

Normalization and Transparency Scripts in Unified CM

A number of normalization and transparency scripts are pre-loaded into Unified CM, and the following scripts are a representative sample of them:

- Refer-passthrough script — This script allows Unified CM to be removed from the call signaling path when a blind transfer (using an in-dialog REFER) is invoked between two SIP trunks.
- ContactHeader script — This script removes the audio and video attributes from the contact header in an inbound Delayed Offer mid-call re-invite.
- HCS-PCV-PAI-passthrough script — This script is used for integration with IMS-based networks and passes through/ adds the P-Charging-Vector header in INVITE, UPDATE & 200 OK messages.
- Diversion-Counter script — This script provides the capability to adjust the diversion counter for various Call Forward scenarios.
- VCS-interop script — This script provides interoperability for endpoints registered to the Video Communication Server (VCS).

Route Lists Run on All Active Unified CM Nodes

Although this is not specifically a SIP trunk feature, running route lists on all nodes provides benefits for trunks in route lists and route groups. Running route lists on all nodes improves outbound call distribution by using the "route local" rule to avoid unnecessary intra-cluster traffic.

For route lists, the route local rule operates as follows:

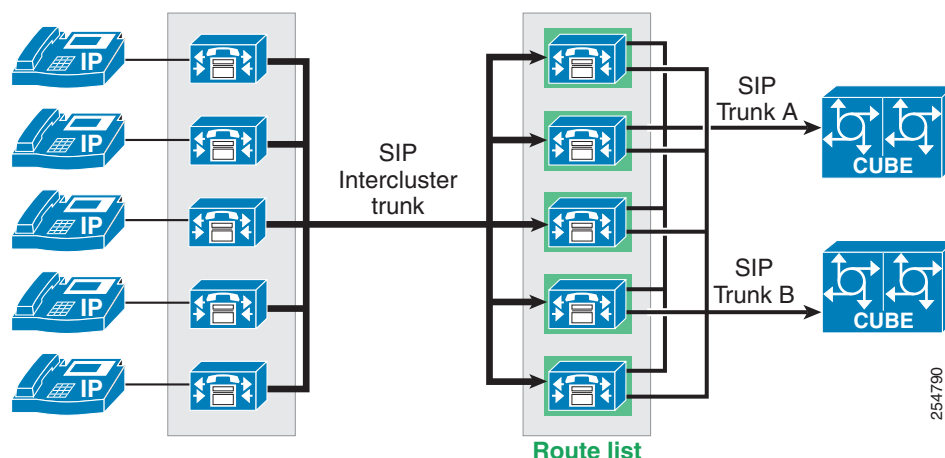
For outbound calls that use route lists (and associated route groups and trunks), when a call from a registered phone or inbound trunk arrives at the node with the route list instance, Unified CM checks to see if an instance of the selected outbound trunk exists on the same node as the route list. If so, Unified CM will use this node to establish the outbound trunk call.

If both the route list and the trunk have **Run on all Active Unified CM Nodes** enabled, outbound call distribution will be determined by the node on which the inbound call arrives. When the selected outbound trunk uses Unified CM Groups instead of running on all nodes, Unified CM will apply the route local rule if an instance of the selected outbound trunk exists on the same node on which the inbound call arrived. If an instance of the trunk does not exist on this node, then Unified CM will forward the call (within the cluster) to a node where the trunk is active.

If the route list does not have **Run on all Active Unified CM Nodes** enabled, an instance of the route list will be active on one node within the cluster (the primary node of the route list's Unified CM Group) and the route local rule will be applied on this node. With this configuration, Cisco recommends that you do not use the primary node of the route list's Unified CM Group in the Unified CM Group of any of the trunks associated with the route list because this can result in sub-optimal outbound call distribution.

As a general recommendation, **Run on all Active Unified CM Nodes** should be enabled for all route lists. (See [Figure 6-8](#).)

Figure 6-8 Route Lists Running on All Active Unified CM Nodes



SIP Trunks Using DNS

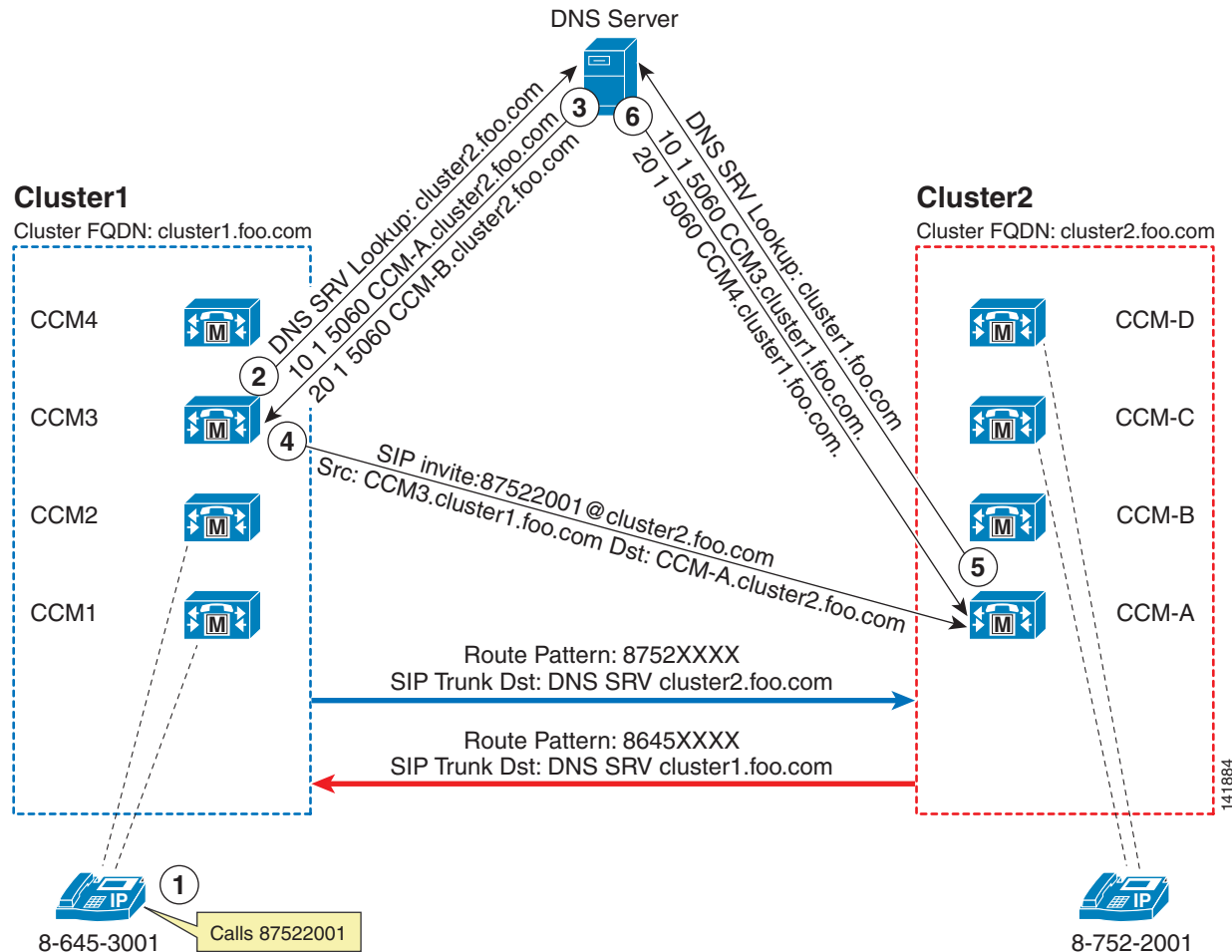
Using a DNS SRV entry as the destination of a SIP trunk might be preferable to defining multiple destination IP addresses in certain situations such as the following:

- SRV host prioritization is required
- SRV host weighting is required
- More than 16 destination IP addresses are required
- DNS SRV resolution is a requirement of the destination Unified Communications system

**Note**

If the configuration option **Destination Address is an SRV** is selected, only a single SRV entry can be added as the trunk destination. (For example, Destination Address = cluster1.cisco.com. Port = 0.)

[Figure 6-9](#) shows the call flow for a SIP trunk using DNS SRV to resolve the addresses to a destination Unified CM cluster. However, this destination could also be a third-party unified communications system.

Figure 6-9 Call Flow for Intercluster SIP Trunk Using DNS SRV

Note: The DNS A Lookup has been removed from this call flow

Figure 6-9 illustrates the following steps in the call flow:

1. The IP phone in Cluster1 calls 87522001.
2. The call matches a route pattern of 8752XXXX that is pointing to the SIP trunk with DNS SRV of cluster2.foo.com. CCM3 in Cluster1 is the node handling this call because the SIP trunk is registered to it. CCM3 sends a DNS SRV lookup for cluster2.foo.com
3. The DNS server replies with two records: CCM-A.cluster2.foo.com and CCM-B.cluster2.foo.com. Because CCM-A.cluster2.foo.com has a higher priority, the call is attempted to this Unified CM. Before sending the SIP Invite, another DNS lookup is done for CCM-A.cluster2.foo.com.
4. CCM3 sends a SIP Invite to 87522001@cluster2.foo.com, with destination address set to the IP address of CCM-A.
5. Unified CM interprets this call as a local call because the host portion of the uniform resource identifier (URI) matches the Cluster FQDN enterprise parameter. Cluster2 does not have any SIP trunk configured with a destination of CCM3, so it does a DNS SRV lookup for all domains configured under the SIP trunks with DNS SRV. In this case, the example shows a single trunk with a DNS SRV destination of cluster1.foo.com

6. The DNS server returns two entries, and one of them matches the source IP address of the Invite. The cluster accepts the call and extends it to extension 87522001.

High Availability for SIP Trunks

A variety of Unified CM options is available for configuring high availability with SIP trunks, all of which can be combined to provide redundancy and resiliency for both the source and destination servers of SIP trunks. These options can be categorized as follows:

- [Multiple Source Unified CM Servers for Originating SIP Trunk Calls, page 6-19](#)
- [Multiple Destination IP Addresses per SIP Trunk, page 6-20](#)
- [Multiple SIP Trunks Using Route Lists and Route Groups, page 6-20](#)
- [SIP OPTIONS Ping, page 6-20](#)

Multiple Source Unified CM Servers for Originating SIP Trunk Calls

Using Standard Unified CM Groups

The nodes defined in the Unified CM Group associated with an individual trunk make up the set of servers that can place or receive calls over that trunk. Up to three nodes can be defined in a Unified CM Group, thus ensuring high availability of the trunk itself.

Using Run on All Active Unified CM Nodes

The **Run on all Active Unified CM Nodes** feature creates and enables a SIP trunk instance on each call processing subscriber within the cluster, thus allowing these nodes to place or receive calls over the trunk.

The Unified CM Route Local Feature And Its Effect on Subscriber Selection for Outbound SIP Trunk Calls

The Route Local feature in Unified CM is designed to reduce intra-cluster traffic. The feature operates as illustrated by the following example:

When a device such as a phone is making an outbound call over SIP Trunk 1, if an instance of SIP Trunk 1 is active on the same node as the one to which the phone is registered, then always use this co-located SIP Trunk 1 instance rather than internally routing the call to another SIP Trunk 1 instance on another node within the cluster.

The effect of the Route Local feature on node selection depends on whether Unified CM Groups or **Run on all Active Unified CM Nodes** is configured on the trunk. For trunks with **Run on all Active Unified CM Nodes** configured, the node to which the calling device is registered is used to make the outbound SIP trunk call. When Unified CM Groups are used on the trunk, if the calling device is registered to one of the nodes in the trunk's Unified CM Group, then the Route Local rule applies. If the calling device is not registered to one of the nodes in the trunk's Unified CM Group, then Unified CM will randomly distribute the call over the nodes in the trunk's Unified CM Group.

Using **Run on all Active Unified CM Nodes** is the recommended approach for SIP trunks because it allows call distribution across nodes to be determined by the calling device and it minimizes intra-cluster traffic.

Multiple Destination IP Addresses per SIP Trunk

A single SIP trunk can be configured with up to 16 destination IP addresses. Unified CM uses random distribution to the configured destination IP addresses when placing calls over a SIP trunk. Using multiple IP addresses on a SIP trunk can help to reduce the need to deploy multiple trunks with route lists and route groups.

Design Considerations When Using Run on All Active Unified CM Nodes

When using **Run on All Active Unified CM Nodes** in conjunction with multiple destination addresses, be aware that to accept inbound calls, the inbound source IP address received on the SIP trunk must match with a configured destination IP address on the inbound trunk. For example, if **Run on all Active Unified CM Nodes** is configured on the SIP intercluster trunk in each cluster, then each trunk must be configured with the corresponding destination address of every active node in the destination cluster. Where clustering over the WAN designs are deployed and geographic call distribution and failover are required, use standard Unified CM Groups on multiple intercluster trunks (each with up to three destination IP addresses) in conjunction with route lists and route groups.

Multiple SIP Trunks Using Route Lists and Route Groups

Multiple prioritized SIP trunks are often required to address failure scenarios in Unified Communications designs. These trunks should be configured in route groups in a single route list and associated with a route pattern. If Unified CM is not able to place a call over the selected trunk in the list, it will try the next trunk in the list. As a general recommendation, enable **Run on all Active Unified CM Nodes** for all route lists.

SIP OPTIONS Ping

SIP OPTIONS Ping can be enabled on the SIP Profile associated with a SIP trunk to dynamically track the state of the trunk's destination(s). When this option is enabled, each node running the trunk's SIP daemon will periodically send an OPTIONS Request to each of the trunk's destination IP addresses to determine its reachability. Enabling SIP OPTIONS Ping is recommended for all SIP trunks that require high availability because it allows Unified CM to dynamically track trunk state rather than determining trunk state on a per-call and timeout basis.

Load Balancing for SIP Trunks

When designing load balancing for SIP trunks, consider both the node that sources the call and its destination. With Unified CM SIP trunks, the node used to originate the call is determined by the Route Local rule, the number of nodes on which the outbound trunk is active, and whether a route list is used in conjunction with multiple outbound trunks. These considerations are discussed in the following sections.

Outbound Calls over a Single SIP Trunk

A single SIP trunk can run on up to three Unified CM nodes in a Unified CM Group, or it can run on all active Unified CM nodes in the cluster. To select the source node for outbound calls, Unified CM applies the following decision processes:

- Where an instance of the trunk runs on all nodes, the Route Local rule applies and the node used for each outbound call is determined by the node on which the call arrives (for example, the node to which the calling phone is registered or the node on which the inbound trunk call arrives).
- Where Unified CM Groups are used, the Route Local rule still applies for those calling devices that are registered to the same node as the nodes in the trunk's Unified CM Group. For calling devices that are registered to other servers within the cluster, Unified CM will randomly distribute calls across the nodes in the trunk's Unified CM Group. For Unified CM SIP trunks, calls are randomly distributed across the trunk's configured destination addresses. SIP trunks may be configured with up to 16 destination IP addresses.

Outbound Calls over Multiple SIP Trunks

Because SIP trunks can run on all active Unified CM nodes and have up to 16 destination addresses, multiple SIP trunks typically do not need to be used to provide even call distribution between two Unified Communications systems. Where multiple trunks are used with route lists and route groups, route lists should be enabled to run on all active Unified CM nodes. Multiple SIP trunks are often used in conjunction with route lists to provide failover to the PSTN or to a group of Unified CM servers in a different site as part of a cluster deployed over the WAN. The selection of the Unified CM node used to initiate an outbound SIP trunk call, and the distribution of calls over the trunk's configured destination IP addresses, are determined in the same way as described for single trunks.

SIP OPTIONS Ping

Use OPTIONS Ping to dynamically track the state of each destination IP address on each SIP trunk and the collective state of the trunk as a whole. If a destination address is unreachable, Unified CM will not extend calls to this device. When all destinations are unreachable, the SIP trunk is considered to be out-of-service.

SIP Delayed Offer and Early Offer

Cisco Unified CM uses the SIP Offer/Answer model for establishing SIP sessions, as defined in RFC 3264. In this context, an Offer is contained in the Session Description Protocol (SDP) fields sent in the body of a SIP message. The Offer typically defines the media characteristics supported by the device (media streams, codecs, directional attributes, IP address, and ports to use). The device receiving the Offer sends an Answer in the SDP fields of its SIP response, with its corresponding matching media streams and codec, whether accepted or not, and the IP address and port on which it wants to receive the media streams. Unified CM uses this Offer/Answer model to establish SIP sessions as defined in the key SIP standard, RFC 3261.

RFC 3261 defines two ways that SDP messages can be sent in the Offer and Answer. These methods are commonly known as Delayed Offer and Early Offer, and support for both methods by User Agent Client/Servers is a mandatory requirement of 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 without SDP in the message body defines a Delayed Offer.

In an Early Offer, the session initiator (calling device) sends its capabilities (for example, codecs supported) in the SDP contained in the initial Invite (thus allowing the called device to choose its preferred codec for the session). In a Delayed Offer, the session initiator does not send its capabilities in the initial Invite but waits for the called device to send its capabilities first (for example, the list of codecs supported by the called device, thus allowing the calling device to choose the codec to be used for the session).

Delayed Offer and Early Offer are the two options available to all standards-based SIP switches for media capabilities exchange. Most vendors have a preference for either Delayed Offer or Early Offer, each of which has its own set of benefits and limitations. For Unified CM SIP trunks, both SIP Delayed Offer and SIP Early Offer are supported. Unified CM Delayed Offer trunks do not require an MTP for the SIP Offer/Answer exchange. For Unified CM Early Offer trunks, **SIP Early Offer for voice and video (insert MTP if needed)** is the preferred configuration option rather than SIP Early Offer using **MTP Required** because MTP resources are typically not required to establish calls over an Early Offer trunk.

**Note**

Unified CM can support Delayed Offer in one direction and Early Offer in the other direction over a SIP trunk. This capability can be useful in situations where a SIP switch connected to Unified CM by a SIP trunk wishes to control the codecs offered and selected for both inbound and outbound calls.

Early Media

In certain circumstances, a SIP session might require that a media path be set up prior to the finalization of the media capabilities exchange between the two SIP endpoints. To this end, the SIP protocol allows the establishment of Early Media after the initial Offer has been received by an endpoint. Some reasons for using Early Media include:

- The called device might want to 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 want to establish an Early Media RTP path to access a DTMF or voice-driven interactive voice response (IVR) system.

The requirement to send one-way Early Media is one reason why service providers offering IP PSTN services accept only Early Offer SIP calls. With Early Offer, the service provider can stream one-way media to the caller after receipt of the initial INVITE with its SDP body containing the caller's media characteristics. An example of the need to stream Early Media to a caller would be where a caller calls a non-existent number and the service provider needs to send a network announcement to the caller without billing them for the attempted call (typically billing commences after two-way media has been established and when the final ACK is received). Unified CM supports the receipt of one-way media with SIP Early Offer calls.

Early two-way media cut-through can also be achieved by enabling Provisional Reliable Acknowledgement (PRACK) on each SIP Unified Communications system. PRACK allows the SIP Offer and Answer to be sent reliably in provisional responses (for example, 1XX responses), thus reducing the number of messages that need to be exchanged before two-way media can be established.

Unified CM supports PRACK-based Early Media for both Early Offer and Delayed Offer calls.

For a SIP trunk to support Early Media cut-through, you must enable PRACK through the **SIP Rel1XX Options** feature in the SIP Profile associate with the trunk.

**Note**

The terms Early Offer and Early Media are often confused, but they are not the same.

Media Termination Points

MTPs are used by Unified CM for the following purposes:

- To deliver a SIP Early Offer over SIP trunks
- To address DTMF transport mismatches
- To act as an RSVP agent
- To act as a Trusted Relay Point (TRP)
- To provide conversion between IPv4 and IPv6 for voice RTP streams

Either of the following methods can be used to enable Early Offer on SIP trunks:

- Check the **MTP Required** checkbox on the SIP trunk.

In this case an MTP is used for every inbound and outbound call, and only voice calls are supported using a single codec as defined on the SIP trunk.

- Check the **Early Offer support for voice and video calls (insert MTP if needed)** checkbox on the SIP Profile associated with the SIP trunk.

With this method an MTP is inserted only if the calling device or trunk cannot send all of the information about its media capabilities in the initial SIP Invite (for example, an inbound call to Unified CM from a SIP Delayed Offer or H.323 Slow Start trunk). In this case, when an MTP is used, additional voice codecs can be supported in the initial call setup by using the MTP's pass-through codec. Once established, this audio call can be escalated to support video and encryption if the call's media is renegotiated (for example, after hold/resume). When an MTP is not needed, all calls support voice, video, and encrypted media.

Unified CM SIP Delayed Offer and Early Offer Recommendations

Cisco Unified CM SIP trunks support Delayed Offer (Invite without SDP) by default. Media termination points (MTPs) are generally not required for Delayed Offer calls from Unified CM SIP trunks and therefore voice, video, and encrypted calls are all supported.

In cases where SIP Early Offer is required on Unified CM SIP trunks, Cisco recommends **Early Offer support for voice and video calls (insert MTP if needed)** because fewer MTP resources are required in comparison with **MTP Required**. When MTPs are used with **Early Offer support for voice and video calls (insert MTP if needed)**, they can provide support for voice, video, and encrypted media.

For IP PSTN SIP trunk connections, SIP Early Offer is generally required by the Service Provider. For designs where the IP PSTN needs to support large numbers of concurrent calls, the Cisco Unified Border Element's SIP Delayed Offer to Early Offer feature can be used as an alternative to using Early Offer on the Unified CM SIP trunk, thereby potentially eliminating MTP usage.

For calls inbound and outbound from Unified CM, endpoints can negotiate the use of RFC 2833 or an out-of-band DTMF method (for example, KPML) end-to-end. If a common DTMF method cannot be negotiated between the endpoints, Unified CM will insert an MTP dynamically.

MTPs are available in three forms:

- Software MTPs in Cisco IOS gateways — Available with any Cisco IOS T-train software release and scaling up to 5,000 sessions (calls) on the Cisco ASR 1000 Series Aggregation Services Routers with Route Processor RP2.
- Hardware MTPs in Cisco IOS gateways — Available with any Cisco IOS T-train software release, hardware MTPs use on-board DSP resources and scale calls according to the number of DSPs supported on the Cisco router platform.
- Cisco Unified CM software MTPs using the Cisco IP Voice Media Streaming Application on a Cisco Media Convergence Server (MCS).

In general, Cisco IOS MTPs are recommended over Unified CM MTPs because Cisco IOS MTPs provide additional functionality such as support for additional codec types and the pass-through codec. (For details, see [Media Termination Point \(MTP\)](#), page 7-8.)

The following example configuration is for a Cisco IOS software-based MTP:

```
!
sccp local Vlan5
sccp ccm 10.10.5.1 identifier 5 version 5.0.1
! Communications Manager IP address (10.10.5.1)
sccp
!
sccp ccm group 5
  bind interface Vlan5
  associate ccm 5 priority 1
  associate profile 5 register MTP000E83783C50
! MTP name (MTP000E83783C50) ... must match the Unified CM MTP name.
!
dspfarm profile 5 mtp
  description software MTP
  codec g711ulaw
  codec pass-through
  maximum sessions software 500
  associate application SCCP
```

DTMF Transport

There are several methods of transporting DTMF information between SIP endpoints. In general terms, these methods can be classified as out-of-band (OOB) and in-band signaling. In-band DTMF transport methods send either raw or signaled DTMF tones within the RTP stream, and they need to be handled and interpreted by the endpoints that generate and/or receive them. Out-of-band signaling methods transport DTMF tones outside of the RTP path, either directly to and from the endpoints or through a call agent such as Cisco Unified CM, which interprets and/or forwards these tones as required.

Out-of-band (OOB) SIP DTMF signaling methods include Unsolicited Notify (UN), Information (INFO), and Key Press Markup Language (KPML). While KPML (RFC 4730) is the OOB signaling method preferred by Cisco, KPML is not widely used in the market place at this time. Currently, the only known products supporting KPML are Cisco Unified CM, Cisco IOS Gateways (Release 12.4 and later), and some models of Cisco IP Phones. INFO is not supported by Unified CM.

In-band 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. Among 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 in-band 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. If the two endpoints are using a back-to-back user agent (B2BUA) server for the call control (for example, Cisco Unified CM) and the endpoints negotiate different DTMF methods between each device and call control box, then the call agent determines how to handle the DTMF differences, either through MTP insertion or by OOB methods. With Unified CM, a DTMF transport mismatch (for example, in-band to out-of-band DTMF) is resolved by inserting a media termination point (MTP), which terminates the RTP stream with in-band DTMF signaling (RFC 2833), extracts the DTMF tones from the RTP stream, and forwards these tones out-of-band to Unified CM, where they are then forwarded to the endpoint supporting out-of-band signaling. In this case, the MTP is always in the media path between the two endpoints because there is no MTP codec dependency for DTMF translation.

In-band DTMF tones can also be transported as raw (audible) tones in the RTP media stream. This transport method is not widely supported by Cisco products and, in general, is not recommended as an end-to-end DTMF transport mechanism. In-band audio DTMF tones can generally be reproduced reliably only when using G.711 a-law or mu-law codecs, and they are not suitable for use with low-bandwidth codecs. In cases where in-band audio is the only available DTMF transport mechanism, the Cisco Unified Border Element can be used to translate the in-band audio DTMF signaling into RFC 2833 signaling.

Over Unified CM SIP trunks, Cisco recommends configuring the DTMF Signaling Method to **No Preference**. This setting allows Unified CM to make an optimal decision for DTMF and to minimize MTP allocation.

SIP Trunk Transport Protocols

SIP trunks can use TCP, TLS (which runs over TCP), or UDP as a message transport protocol. TCP is recommended within Cisco Enterprise Unified Communications networks because it is a reliable and connection-orientated protocol with the capability to fragment and re-assemble large SIP messages. UDP is not connection-orientated or reliable (message delivery is not guaranteed), and it relies on the SIP Invite Retry count and SIP Trying timers to detect and respond to far-end device failures. Use SIP OPTIONS Ping to dynamically track the state of each destination IP address on each SIP trunk and the collective state of the trunk as a whole.

For more information on SIP trunk timer tuning, refer to the configuration example and technical notes at:

http://www.cisco.com/en/US/products/sw/voicesw/ps556/products_configuration_example09186a008082d76a.shtml



Note

Although TCP is the recommended transport protocol within a Cisco Enterprise Unified Communications network, most service providers prefer to use UDP because it has a lower processing overhead than TCP. Cisco Unified Border Element can be used to provide TCP-based SIP trunk connections to the Enterprise Unified Communications network and UDP-based SIP trunk connections to service provider networks.

Secure SIP Trunks

Securing SIP trunks involves two processes:

- Configuring the trunk to encrypt media (see [Media Encryption, page 6-26](#))
- Configuring the trunk to encrypt signaling (see [Signaling Encryption, page 6-26](#))

Media Encryption

Media encryption can be configured on SIP trunks by checking the trunk's **SRTP allowed** check box. It is important to understand that enabling **SRTP allowed** causes the media for calls to be encrypted, but the trunk signaling will not be encrypted and therefore the session keys used to establish the secure media stream will be sent in the clear. It is therefore important that you ensure that signaling between Unified CM and its destination SIP trunk device is also encrypted so that keys and other security-related information do not get exposed during call negotiations.

Signaling Encryption

SIP trunks use TLS for signaling encryption. TLS is configured on the SIP Security Profile associated with the SIP trunk, and it uses X.509 certificate exchanges to authenticate trunk devices and to enable signaling encryption.

Certificates can be either of the following:

- Imported to each Unified CM node from every device that wishes to establish a TLS connection to that node's SIP trunk daemon
- Signed by a Certificate Authority (CA), in which case there is no need to import the certificates of the remote devices; only the CA certificate needs to be imported.

Unified CM provides a bulk certificate import and export facility. However, for SIP trunks using **Run on all Active Unified CM Nodes** and up to 16 destination addresses, using a Certificate Authority provides a centralized and less administratively burdensome approach to setting up signaling encryption on SIP trunks.

For more information on TLS for SIP trunks, refer to the latest version of the *Cisco Unified Communications Manager Security Guide*, available at

http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html

For information on certificate authorities, refer to the Certificate Authority (CA) information in the latest version of the *Cisco Unified Communications Operating System Administration Guide*, available at

http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_maintenance_guides_list.html

If the system can establish a secure media or signaling path and if the end devices support SRTP, the system uses a SRTP connection. If the system cannot establish a secure media or signaling path or if at least one device does not support SRTP, the system uses an RTP connection. SRTP-to-RTP fallback (and vice versa) may occur for transfers from a secure device to a non-secure device or for conferencing, transcoding, music on hold, and so on.

For SRTP-configured devices, Unified CM classifies a call as encrypted if the **SRTP Allowed** check box is checked for the device and if the SRTP capabilities for the devices are successfully negotiated for the call. If these criteria are not met, Unified CM classifies the call as non-secure. If the device is connected to a phone that can display security icons, the phone displays the lock icon when the call is encrypted.

**Note**

MTPs that are statically assigned to a SIP trunk by means of the **MTP Required** checkbox do not support SRTP because they do not support the pass-through codec.

To ensure that SRTP is supported for all calls, configure the SIP trunk for Delayed Offer.

Where **Early Offer support for voice and video calls (insert MTP if needed)** is configured, for devices that support encryption, all calls that do not need to use MTPs can support SRTP. When an MTP is inserted into the call path, this dynamically inserted MTP supports the pass-through codec, and encrypted calls are supported in the following cases:

- If the calling device is an older SCCP-based phone registered to Unified CM, SRTP can be negotiated in the initial call setup.
- If the call arrives inbound to Unified CM on a Delayed Offer SIP trunk or an H.323 Slow Start trunk, SRTP will not be negotiated in the initial call setup because no security keys are available, but the call can be escalated mid-call to support SRTP if the call media is renegotiated (for example, after hold/resume).

If Unified CM dynamically inserts an MTP for reasons other than Early Offer, such as for a Trusted Relay Point or as an RSVP agent, then SRTP will be supported with an MTP that supports the pass-through codec.

Note that **dtmf-relay** using an MTP (where the MTP needs to convert between in-band and out-of-band DTMF signals) will not function for SRTP because it will be unable to decrypt the DTMF packets in the media stream.

**Note**

SRTP is not supported over SAF-enabled SIP trunks.

Calling Party Number Transformation and SIP Trunks

Unified CM provides the capability to transform calling party numbers of calls inbound over gateways and trunks to a normalized format. Typically, you would want this format to be the globally routable international representation of the number according to E.164 specifications.

The process of normalization relies on receiving the number and the associated number-type of the incoming call. The number-type parameter can be used to select the appropriate digits to prefix to the calling number. Number-types can be one of four types: Unknown, Subscriber, National, or International. For more details and examples on how these number-types are used, refer to the chapter on [Dial Plan, page 14-1](#).

You can specify the prefix digits for each of the four number types in the H.323 trunk and H.323 gateway configuration pages in Unified CM. H.323 can transport these number types in its signaling. SIP, on the other hand, is unable to transport the number-type information in its signaling. Thus, a call coming in through a SIP gateway across a SIP trunk to Unified CM will not have any indication of whether the calling-party number is local, national, or international. Without the number-type information, Unified CM is unable to apply the correct prefix to the calling-party number.

The inability of the SIP trunk to transport the number type implies that the normalization of the calling number must be performed before the call is presented to Unified CM. One place where the transformation can be performed is on the ingress SIP gateway. The following example configuration shows the translation rules that can be defined on a Cisco IOS gateway to accomplish this transformation:

```
voice translation-rule 1
```

```
rule 1 // /+4940/ type subscriber subscriber
rule 2 // /+49/ type national national
rule 3 // /+/ type international international
...
voice translation-profile 1
  translate calling 1
...
dial-peer voice 300 voip
  translation-profile outgoing 1
  destination-pattern .T
  session protocol sipv2
  session target ipv4:9.6.3.12
...
```

When configured as in the example above, a Cisco IOS gateway using SIP to communicate with Unified CM will send calling party information digits normalized to the E.164 format, including the + sign. The Unified CM configuration will receive all calls from this gateway with a numbering type of "unknown" and would not need to add any prefixes.

For more details on configuring translation rules, refer to the document *Voice Translation Rules*, available at

http://www.cisco.com/en/US/tech/tk652/tk90/technologies_tech_note09186a0080325e8e.shtml

Unified CM can set the calling party number of outgoing calls to the normalized global format. The number-type in outgoing calls from the SIP trunk will be "unknown," and the Cisco IOS gateway should change it to International if no stripping is done, or perform a combination of stripping and numbering type change if required by the connected service provider.

SIP Trunk Service Types

Most SIP trunks are general-purpose trunks capable of connecting to a wide variety of SIP servers such as other Cisco Unified CMs, Cisco Unified Border Elements, Cisco Unified Gateways, and so forth. In addition to these all-purpose trunks, Unified CM provides SIP trunks dedicated for specific services. These special-purpose trunks enable technologies such as the following:

- Cisco Intercompany Media Engine (IME)
See [Cisco Intercompany Media Engine, page 10-50](#).
- Cisco Unified Communications Call Control Discovery (CCD) through the Cisco IOS Service Advertisement Framework (SAF)
See [Service Advertisement Framework \(SAF\), page 3-69](#).
- Cisco Extension Mobility Cross Cluster (EMCC)
See [Extension Mobility Cross Cluster \(EMCC\), page 18-10](#).

Design Considerations for SIP Trunks

This section presents key design considerations and guidelines for deploying SIP trunks.

Considerations for SIP Intercluster Trunks

For intercluster trunk connections, the SIP trunk configured in each cluster may be using standard Unified CM Groups or the **Run on all Active Unified CM Nodes** feature. The reasons for using each type of feature will typically be determined by the Unified CM version used in the cluster, or if clustering over the WAN has been deployed and geographically based call distribution is required.

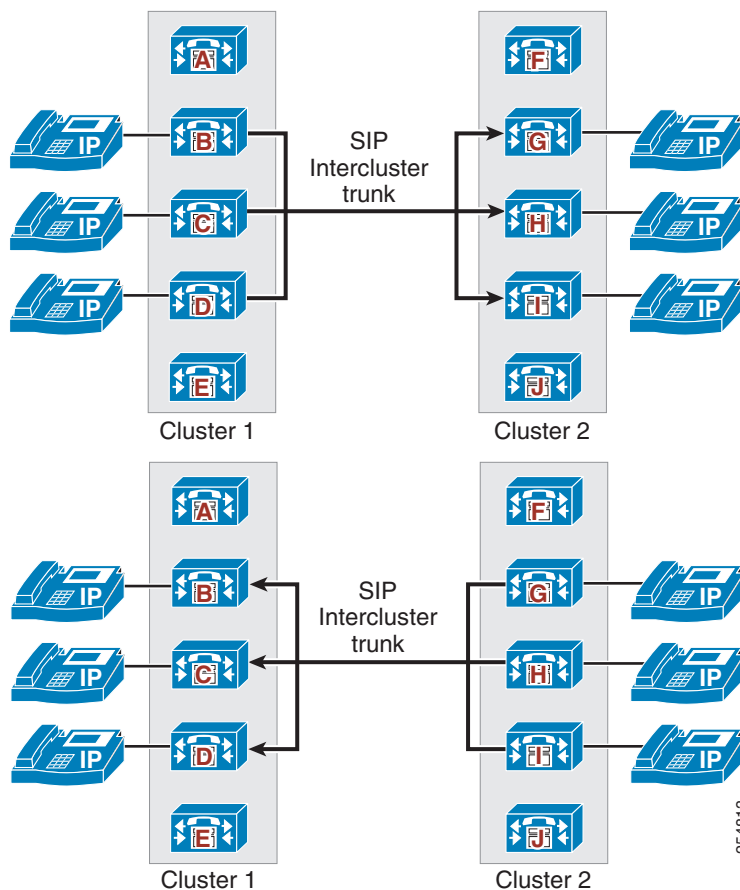
Using Standard Unified CM Groups with SIP Intercluster Trunks

In this type of deployment standard Unified CM Groups are used by SIP intercluster trunks in each cluster. When defining this type of trunk with standard Unified CM Groups, you should define a maximum of three remote Unified CM servers as destination IP addresses in the remote cluster. The trunk will automatically load-balance across all defined remote Unified CM servers. In the remote cluster, it is important to configure a corresponding SIP intercluster trunk that has the same Unified CM nodes in its Unified CM Group as those defined as remote destination Unified CM servers in the first cluster.

For example, if Cluster 1 has a SIP trunk to Cluster 2 and Cluster 2 has a SIP trunk to Cluster 1, the following configurations would be needed (see [Figure 6-10](#)):

- Cluster 1
 - Servers B, C, and D are configured as members of the Unified CM Group defined in the device pool associated with the SIP trunk to Cluster 2.
 - The SIP trunk has Cluster 2's remote servers G, H, and I configured as destinations.
- Cluster 2
 - Servers G, H, and I are configured as members of the Unified CM Group defined in the device pool associated with the SIP trunk to Cluster 1.
 - The SIP trunk has Cluster 1's remote servers B, C, and D configured as destinations.

Figure 6-10 SIP Intercluster Trunks with Unified CM Groups



Using Run on All Active Unified CM Nodes with SIP Intercluster Trunks

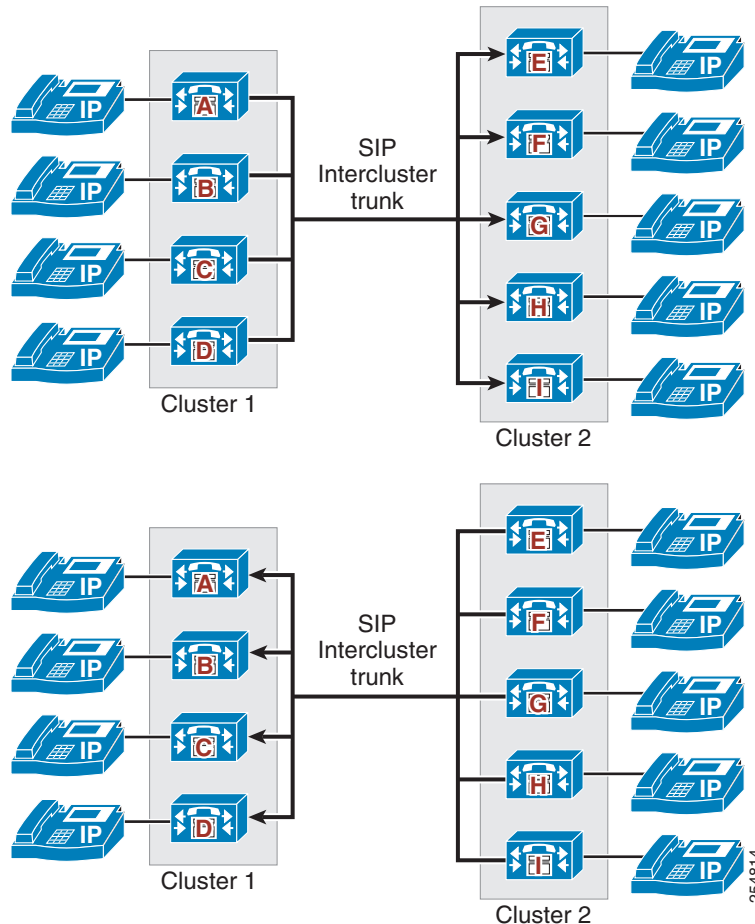
In this type of deployment, **Run on all Active Unified CM Nodes** is used by SIP intercluster trunks in each cluster. When defining this type of trunk you may define up to 16 remote Unified CM servers in the destination cluster. (The number of remote servers that you need to define will depend on the number of active Unified CM nodes in the destination cluster.) The trunk will automatically load-balance calls across all defined remote destination servers. In the remote cluster, it is important to configure a corresponding SIP intercluster trunk that has **Run on all Active Unified CM Nodes** configured, where these nodes are defined as the remote destination Unified CM servers in the first cluster.

For example, if Cluster 1 (with four active nodes) has a SIP trunk to Cluster 2, and Cluster 2 (with five active nodes) has a SIP trunk to Cluster 1, the following configurations would be needed (see [Figure 6-11](#)):

- Cluster 1 has four active Unified CM nodes (A, B, C, and D).
 - Enabling **Run on all Active Unified CM Nodes** causes servers A, B, C, and D to have active SIP trunk daemons associated with the SIP trunk to Cluster 2.
 - The SIP trunk has Cluster 2's remote servers E, F, G, H, and I configured as destinations.

- Cluster 2 has five active Unified CM nodes (E, F, G, H, and I).
 - Enabling **Run on all Active Unified CM Nodes** causes servers E, F, G, H, and I to have active SIP trunk daemons associated with the SIP trunk to Cluster 1.
 - The SIP trunk has Cluster 1's remote servers A, B, C, and D configured.

Figure 6-11 *SIP Intercluster Trunks Running on All Active Unified CM Nodes*



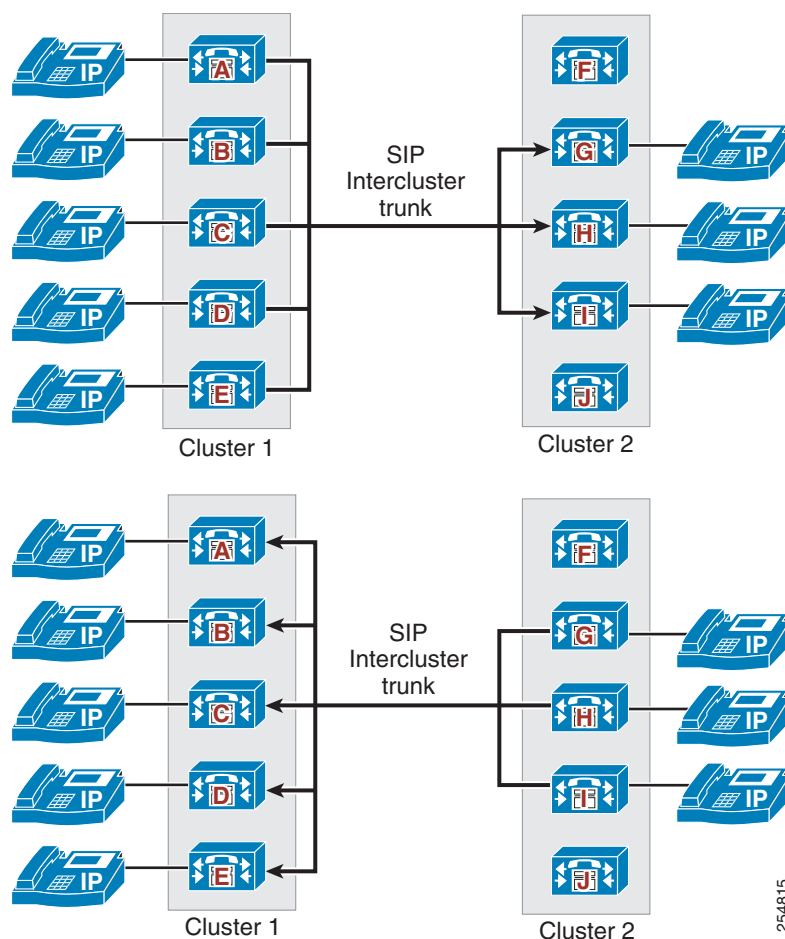
Using Standard Unified CM Groups and Run on All Active Unified CM Nodes with SIP Intercluster Trunks

In this type of deployment, **Run on all Active Unified CM Nodes** is used by the SIP intercluster trunk in one cluster and standard Unified CM Groups are used by the SIP intercluster trunk in the other cluster. When configuring these trunks, the number of remote Unified CM server destinations that you define should match the number of active Unified CM nodes associated with the corresponding trunk in the destination cluster. The trunk will automatically load-balance calls across all defined remote destination Unified CM servers. In the remote cluster, it is important to configure a corresponding SIP intercluster trunk that has Unified CM nodes with active SIP daemons where these nodes are defined as remote destination Unified CM servers in the first cluster.

For example, if Cluster 1 has a trunk to Cluster 2, and Cluster 2 has a trunk to Cluster 1, the following configurations would be needed (see Figure 6-12):

- Cluster 1 has five active Unified CM nodes (A, B, C, D, and E).
 - Enabling **Run on all Active Unified CM Nodes** causes servers A, B, C, D, and E to have active SIP trunk daemons associated with the SIP trunk to Cluster 2.
 - The SIP trunk has Cluster 2's remote servers G, H, and I configured as destinations.
- Cluster 2 has five active Unified CM nodes and uses an intercluster trunk with a Unified CM Group containing nodes G, H, and I.
 - Servers G, H, and I are configured as members of the Unified CM Group defined in the device pool associated with the SIP trunk to Cluster 1.
 - The SIP trunk has Cluster 1's remote servers A, B, C, D, and E configured as destinations.

Figure 6-12 SIP Intercluster Trunks Using Unified CM Groups and Run on All Active Unified CM Nodes



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Trunk Type and Feature Recommendations for Multi-Cluster Deployments

This section provides some trunk recommendations that vary with different releases of Cisco Unified Communications Manager (Unified CM).

Multiple Clusters All Running Unified CM 8.5 or Later Releases

Where all clusters are running Unified CM 8.5 or later releases, the following SIP trunk features should be used where applicable (see [Figure 6-13](#)):

- SIP OPTIONS Ping
- SIP Delayed Offer
- Early Offer support for Voice and Video (insert MTP if needed)
- Run on All Active Unified CM Nodes
- Multiple destination IP addresses
- Audio codec preference lists
- QSIG over SIP
- SIP Normalization and Transparency

Deploying these features reduces MTP usage and provides high availability, even call distribution, and dynamic SIP trunk failure detection. For Unified CM SIP trunks, both SIP Delayed Offer and SIP Early Offer can be used. Unified CM Delayed Offer trunks do not require an MTP for the SIP Offer/Answer exchange. For Unified CM Early Offer trunks, **SIP Early Offer for voice and video (insert MTP if needed)** is the preferred configuration option rather than SIP Early Offer using **MTP Required** because MTP resources are typically not required to establish calls over an Early Offer trunk.

In general, for outbound calls from Unified CM SIP trunks, Delayed Offer is preferred because MTPs are not required to establish a Delayed Offer call. For inbound calls to Unified CM SIP trunks, Early Offer or Delayed Offer (or a mixture of both Early Offer and Delayed Offer) can be used.

SIP intercluster trunks support voice; video, and encrypted media between Unified CM clusters, and all of the above features can be used. If multiple trunks are used with route lists, enable the **Run on All Active Unified CM Nodes** feature on the route lists.

For SIP trunks to an IP PSTN, SIP Early Offer is typically required by the service provider, and most providers support voice calls only. However, if required, video calls and encrypted media are also supported. Where SIP Early Offer is required by the Service Provider, the Cisco Unified Border Element (SIP Delayed Offer to Early Offer feature) can be used as an alternative to configuring Early Offer on the Unified CM SIP trunk. For inbound calls to Unified CM SIP trunks, Early Offer or Delayed Offer (or a mixture of both Early Offer and Delayed Offer) can be used. When connecting to a service provider's IP PSTN network, Cisco strongly recommends the use of the Cisco Unified Border Element as an enterprise edge Session Border Controller to provide a controlled demarcation and security point between your enterprise and the service provider's network.

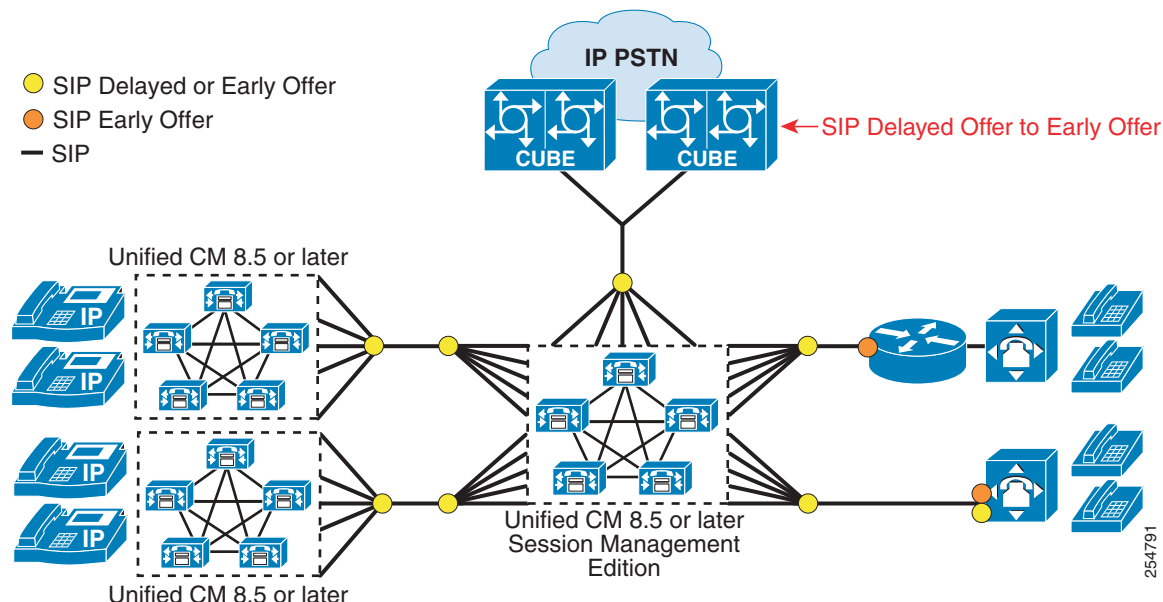
SIP trunks to third-party unified communications systems may support voice, video, and encrypted media. Check the capabilities of the end system to determine the SIP trunk features and media capabilities that it supports. For outbound calls from Unified CM SIP trunks, Delayed Offer or Early Offer can be used. For inbound calls to Unified CM SIP trunks, Early Offer or Delayed Offer (or a mixture of both Early Offer and Delayed Offer) can be used.

**Note**

SIP trunks on Cisco IOS gateways always send Early Offer.

For SIP trunk connections to the IP PSTN and third-party unified communications systems, normalization and transparency scripts can be used to address SIP interoperability issues.

Figure 6-13 Multi-Cluster Deployments with Unified CM 8.5 and Later Releases



Multiple Clusters Running Unified CM 8.5 and Prior Releases

When the leaf clusters are running Unified CM 8.5 or a later release in combination with leaf clusters running prior releases of Unified CM, the following trunk types and features should be used (see Figure 6-14):

When the leaf cluster is running an earlier version (pre-8.5) of Unified CM and voice, video, and encryption are required, use H.323 Slow Start intercluster trunks and Annex M1 (QSIG) if desired. Deploy one or more H.323 Slow Start intercluster trunks using standard Unified CM Groups and up to three destination IP addresses. If multiple trunks are used with route lists, to avoid the Route Local rule (described earlier) ensure that the primary server in the route list's Unified CM Group does not reside on the same node as an associated outbound H.323 trunk.

For leaf clusters running Unified CM 8.5 and later releases, use a SIP Delayed Offer or Early Offer (for voice and video (insert MTP if needed)) intercluster trunk, enable **Run on All Active Unified CM Nodes**, and use multiple destination IP addresses and SIP OPTIONS Ping for high availability and even call distribution. If multiple trunks are used with route lists, enable the **Run on All Active Unified CM Nodes** feature on the route lists.

Using SIP Delayed Offer or Early Offer (for voice and video (insert MTP if needed)) intercluster trunks on Unified CM 8.5 (or later release) leaf clusters and H.323 Slow Start intercluster trunks on leaf clusters using earlier versions of Unified CM, allows voice, video, and encrypted calls to be made between clusters and reduces the number MTPs required.

For SIP trunks to an IP PSTN, SIP Early Offer is typically required by the service provider, and most providers support voice calls only. However, if required, video calls and encrypted media are also supported. Where SIP Early Offer is required by the service provider, the Cisco Unified Border Element (SIP Delayed Offer to Early Offer feature) can be used as an alternative to configuring Early Offer on

the Unified CM SIP trunk. For inbound calls to Unified CM SIP trunks, Early Offer or Delayed Offer (or a mixture of both Early Offer and Delayed Offer) can be used. When connecting to a service provider's IP PSTN network, Cisco strongly recommends the use of the Cisco Unified Border Element as an enterprise edge Session Border Controller to provide a controlled demarcation and security point between your enterprise and the service provider's network.

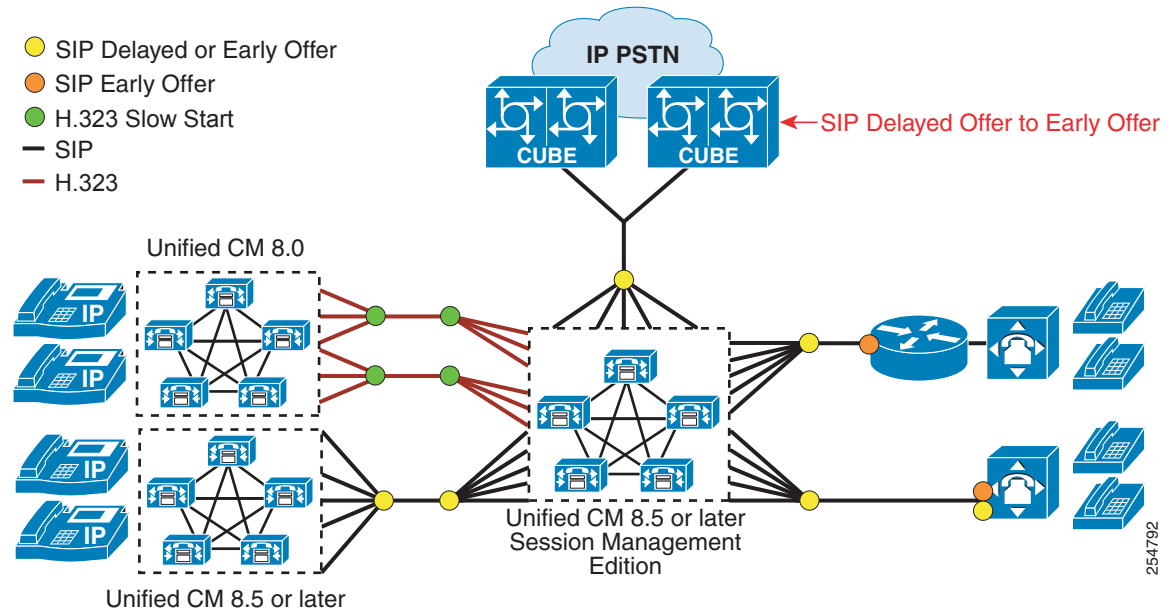
SIP trunks to third-party unified communications systems may support voice, video, and encrypted media. Check the capabilities of the end system to determine the SIP trunk features and media capabilities that it supports. For outbound calls from Unified CM SIP trunks, Delayed Offer or Early Offer can be used. For inbound calls to Unified CM SIP trunks, Early Offer or Delayed Offer (or a mixture of both Early Offer and Delayed Offer) can be used.

**Note**

SIP trunks on Cisco IOS gateways always send Early Offer.

For Unified CM SIP trunk connections to the IP PSTN and third-party unified communications systems, normalization and transparency scripts can be used to address SIP interoperability issues.

Figure 6-14 Multi-Cluster Deployments with Unified CM 8.5 and Prior Releases



Trunk Design Considerations for Clustering over the WAN

When deploying clustering over the WAN for spatial resilience and redundancy, SIP trunk features such as OPTIONS Ping and QSIG can be used as required and appropriate. SIP and H.323 trunk features such as **Run on all Unified CM Nodes** and multiple destination addresses should be used with consideration, primarily because of the mechanism that trunks use to identify and accept inbound calls. (A trunk will accept a call if the incoming source IP address matches one of the addresses defined as its destination IP address.)

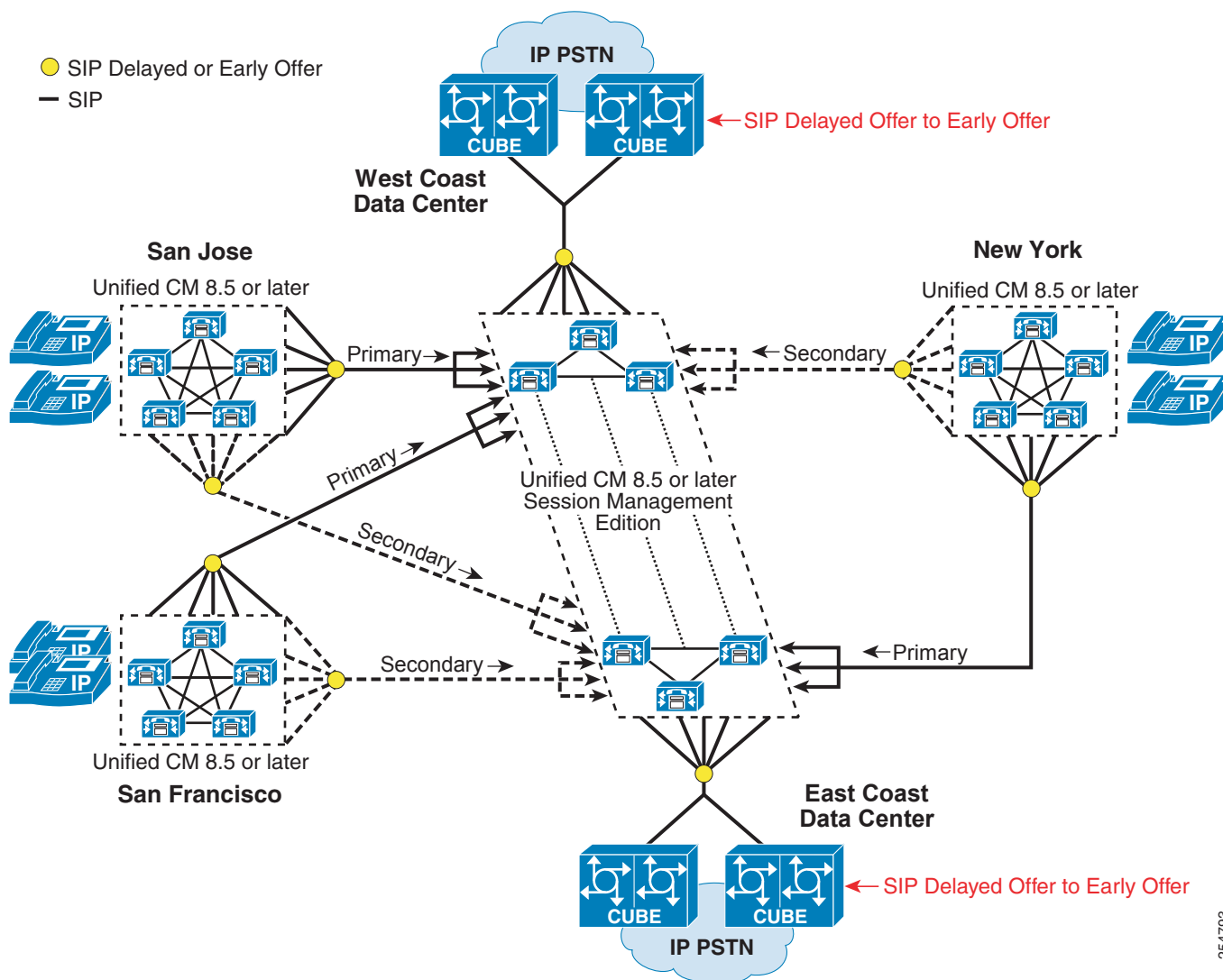
For clustering over the WAN deployments where calls need to be routed to different groups of Unified CM nodes based on their geographic location, consideration should be given to the trunk configuration for both inbound and outbound calls. This is described in the following section, using a Unified CM Session Management Edition cluster that is clustered over the WAN as an example.

Design Guidance for Clustering over the WAN with Leaf Cluster Trunks

Create and prioritize multiple SIP trunks in route lists in each leaf cluster to distribute calls to each group of Unified CM Session Management Edition nodes in each data center, and run route lists on all nodes. (See Figure 6-15.)

Enable **Run on all Nodes** on each leaf cluster SIP trunk (each SIP trunk must use a unique incoming port number). Define destination IP addresses per trunk for geographic call distribution.

Figure 6-15 Calls from Leaf Clusters to Unified CM Session Management Edition

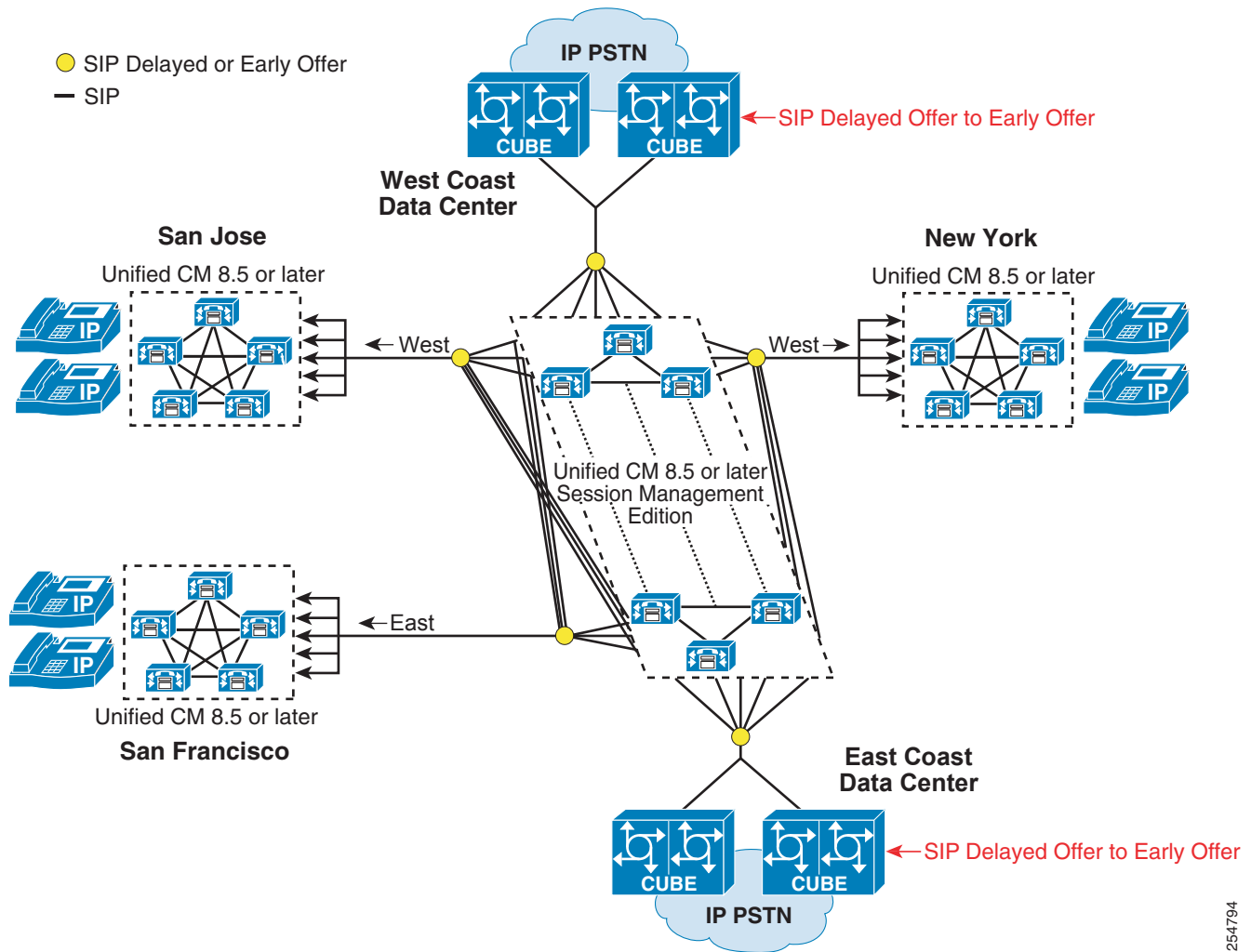


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Design Guidance for Clustering over the WAN with Unified CM Session Management Edition Cluster Trunks

For each leaf cluster, create a single SIP trunk on the Unified CM Session Management Edition cluster. Enable **Run on all Unified CM Nodes** on this SME trunk and configure destination IP addresses for each call processing node in the leaf cluster. (See [Figure 6-16](#).)

Figure 6-16 Calls from Unified CM Session Management Edition to Leaf Clusters



254794

Other SIP Trunk Deployment Considerations

Voice clipping, if observed, can be minimized or eliminated by enabling PRACK on the trunk. The PRACK feature can be enabled via the **SIP Rel1XX Options** of the Trunk Specific Configuration section in the SIP trunk's SIP Profile Configuration. Note that PRACK must be supported by both SIP Unified Communications systems.

Other operating parameters for security settings and the types of messages accepted over a SIP trunk can be enabled in the SIP Trunk Security Profile. Here you can set parameters not only for TLS and Digest Authentication, but also for whether or not the trunk will accept Presence Subscription, an out-of-dialog REFER message, Replaces header, or an Unsolicited Notify message.

SIP trunks support topology-aware RSVP call admission control using SIP Preconditions and locations-based call admission control which is unaware of the underlying WAN topology.

For connection to service provider networks, Cisco recommends the use of the Cisco Unified Border Element. In addition to providing a demarcation point between the enterprise and service provider networks, the Cisco Unified Border Element can also be used for address hiding and enhancing SIP signaling interoperability between the two networks.

For more information on the Cisco Unified Border Element, refer to the documentation available at

<http://www.cisco.com/en/US/products/sw/voicesw/ps5640/index.html>

H.323 Trunks Overview

H.323 trunks provide connectivity to other H.323 devices such as gateways, Unified CM Session Management Edition, gatekeepers, Unified Communications applications, and other Unified CM clusters. Cisco Unified CM provides the following call routing enhancement for all H.323 trunk types:

- Run route lists on all Unified CM nodes

In addition to this, H.323 non-gatekeeper controlled intercluster trunks also support the following features:

- Run on all Unified CM nodes
- Up to 16 destination IP addresses per trunk

These two features improve outbound call distribution from Unified CM clusters and reduce the number of H.323 non-gatekeeper controlled intercluster trunks required between clusters.

These features and their operation are discussed in detail later in this section.

For the complete list of new enhancements for H.323 trunks, refer to the latest Cisco Unified Communications Manager product release notes available at

http://www.cisco.com/en/US/products/sw/voicesw/ps556/prod_release_notes_list.html

General H.323 Intercluster Trunk Deployment Considerations

Prior to Unified CM 8.5, H.323 Annex M1 trunks were the preferred choice for connections between Unified CM clusters. Unified CM SIP trunks now offer a greater set of features in comparison with H.323 intercluster trunks, thus making SIP the protocol of choice for intercluster trunk connections. However, the majority of Unified CM clusters using earlier software versions are likely to be deployed with H.323 Annex M1 intercluster trunks, and this may determine the intercluster trunk type that you use to these clusters.

Basic Operation of H.323 Trunks

H.323 trunks provide connectivity to other Unified CM clusters and other H.323 devices such as gateways. H.323 trunks support most of the audio and video codecs that Unified CM supports for intra-cluster communications, with the exception of wideband audio and wideband video.

H.323 trunks use the Empty Capabilities Set (ECS) to provide supplementary call services such as hold/resume and transfer. This method is a standard H.245 mechanism to stop or close a media stream (or channel) and start or open it to the same or a different endpoint address. This method allows Unified CM to keep a call active while still being able to control the source and destination of the media streams on the fly.

For example, consider a call between two clusters (A and B) using the H.323 trunk. When a user in cluster A places a user in cluster B on hold, the media streams between the two users are closed and the user in cluster B is connected to a music on hold (MoH) server in cluster A. The MoH server is instructed to send media (the music file) to the user. When the user in cluster A resumes the call, the MoH stream is closed and the two-way media streams are reopened between the two users. (Unified CM does not support H.450 for supplementary call services.) In this case, MoH is an example of an ECS operation. H.323 trunks support multicast MoH, therefore the media resource group list (MRGL) for the H.323 trunks can contain both unicast and multicast MoH sources. (For details, see [Music on Hold, page 7-18](#).)

The bandwidth used for calls on H.323 trunks can be controlled by the use of regions configured in Unified CM and assigned to each trunk. A region limits the amount of bandwidth allocated for calls by specifying the inter-region Max Audio Bit Rate for audio and the inter-region Max Video Call Bit Rate setting for video (that includes audio). Calls between one region and another region must be within the specified bandwidth limit. If the device making the call over the H.323 trunk is in a more restrictive region or does not support a particular codec such as video, then it is a subset of codecs that are allowed for that call.

H.323 Trunk Types

The following major types of H.323 trunks can be configured in a Unified CM:

- [Intercluster Trunk \(Non-Gatekeeper Controlled\), page 6-39](#)
- [Intercluster Trunk \(Gatekeeper Controlled\), page 6-46](#)
- [H.225 Trunk \(Gatekeeper Controlled\), page 6-47](#)

Each of these H.323 trunk types and their specific design considerations are discussed in the following sections.

Intercluster Trunk (Non-Gatekeeper Controlled)

This trunk is the simplest H.323 trunk type and is used for connecting to other Unified CM clusters in either a multi-cluster single campus or a distributed call processing deployment. This trunk does not use a gatekeeper for call admission control, although it may use locations configured in Unified CM if bandwidth control is required.

Cisco Unified CM supports the following trunk features and call routing enhancements for H.323 non-gatekeeper controlled intercluster trunks:

- Run on all active Unified CM nodes
- Up to 16 destination IP addresses per trunk
- Run route lists on all Unified CM nodes

These features are discussed in the following sections.

H.323 Non-Gatekeeper Intercluster Trunks Running on all Active Unified CM Nodes

When the **Run on all Active Unified CM Nodes** option is checked on a H.323 non-gatekeeper intercluster trunk, Unified CM creates an instance of the H.323 trunk daemon on every call processing Unified CM Groups.) This allows H.323 non-gatekeeper intercluster trunk calls to be made or received on any call processing subscriber. With **Run on all Active Unified CM Nodes** enabled, outbound H.323 non-gatekeeper intercluster trunk calls originate from the same server on which the inbound call (for example, from a phone or trunk) is received. As with all Unified CM H.323 non-gatekeeper intercluster trunks, the H.323 daemons associated with the trunk will accept only inbound calls from end systems with IP addresses that are defined in the trunk's destination address fields. Running the H.323 non-gatekeeper intercluster trunk on all nodes is recommended where the H.323 a non-gatekeeper intercluster trunk is required to process a large number of calls, so that outbound and inbound call distribution can be evenly spread across all call processing subscribers within a cluster. Bear in mind that (unlike SIP trunks) H.323 non-gatekeeper intercluster trunks use a fixed destination port and an ephemeral source, and therefore H. 323 non-gatekeeper intercluster trunks cannot be differentiated using port numbers. When configuring H.323 non-gatekeeper intercluster trunks, make sure that each trunk uses different destination IP addresses when **Run on all Active Unified CM Nodes** is enabled.

Up to 16 Destination IP Addresses per H.323 Non-Gatekeeper Intercluster Trunk

An H.323 non-gatekeeper intercluster trunk can be configured with up to 16 destination IP addresses. Support for additional destination IP addresses reduces the need to create multiple trunks associated with route lists and route groups for call distribution between two Unified Communications systems, thus simplifying Unified CM trunk design. This feature can be used in conjunction with the **Run on all Active Unified CM Nodes** feature. Bear in mind, however, that the H.323 daemons associated with a Unified CM H.323 non-gatekeeper intercluster trunk will accept only inbound calls from end systems with IP addresses that are defined in the trunk's destination address fields.

Route Lists Running on All Active Unified CM Nodes

Although this is not specifically an H.323 non-gatekeeper intercluster trunk feature, running route lists on all nodes provides benefits for trunks in route lists and route groups. Running route lists on all nodes improves outbound call distribution by using the Route Local rule to avoid unnecessary intra-cluster traffic.

For route lists, the Route Local rule operates as follows:

For outbound calls that use route lists and associated route groups and trunks, when a call from a registered phone or inbound trunk arrives at the node with the route list instance associated with the trunk selected for the outbound call, Unified CM checks to see if an instance of the selected outbound trunk exists on the same node as the route list. If so, Unified CM will use this node to establish the outbound trunk call.

If both the route list and the selected outbound trunk have **Run on all Active Unified CM Nodes** enabled, outbound call distribution will be determined by the node on which the inbound call arrives. When the selected outbound trunk uses Unified Groups instead of running on all nodes, Unified CM will

apply the Route Local rule if an instance of the selected outbound trunk exists on the same node on which the inbound call arrived. If an instance of the trunk does not exist on this node, then Unified CM will forward the call (within the cluster) to a node where the trunk is active.

If the route list does not have **Run on all Active Unified CM Nodes** enabled, the route list will be active on one node within the cluster (the primary node in the route list's Unified Group) and the Route Local rule will be applied on this node.

As a general recommendation, **Run on all Active Unified CM Nodes** should be enabled for all route lists.

Design Considerations for H.323 Non-Gatekeeper Intercluster Trunks

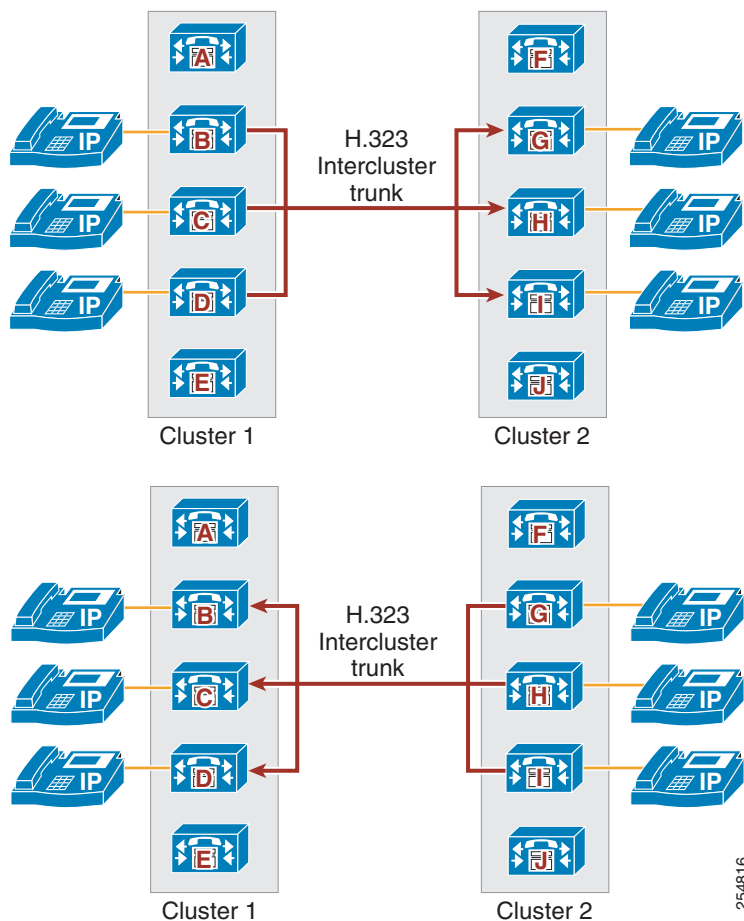
For intercluster trunk connections, the H.323 non-gatekeeper intercluster trunk configured in each cluster may be using standard Unified CM Groups or the **Run on all Active Unified CM Nodes** feature. The reasons for using each type of feature will typically be determined by the Unified CM version used by a cluster, or if clustering over the WAN has been deployed and geographically based call distribution is required.

Using Standard Unified CM Groups with H.323 Non-Gatekeeper Intercluster Trunks

In this type of deployment, standard Unified CM Groups are used by H.323 non-gatekeeper intercluster trunks in each cluster. When defining this type of trunk with standard Unified CM Groups, you should define a maximum of three remote Unified CM servers in the destination cluster. The trunk will automatically load-balance calls across all servers defined as remote destination addresses. In the remote cluster, it is important to configure a corresponding intercluster trunk (non-gatekeeper controlled) that has the same Unified CM nodes in its Unified CM Group as those defined as remote destination Unified CM servers in the first cluster.

For example, if Cluster 1 has a trunk to Cluster 2, and Cluster 2 has a trunk to Cluster 1, the following configurations would be needed (see [Figure 6-17](#)):

- Cluster 1
 - Servers B, C, and D are configured as members of the Unified CM Group defined in the device pool associated with the non-gatekeeper controlled trunk to Cluster 2.
 - The non-gatekeeper controlled trunk has Cluster 2's remote servers G, H, and I configured as destinations.
- Cluster 2
 - Servers G, H, and I are configured as members of the Unified CM Group defined in the device pool associated with the non-gatekeeper controlled trunk to Cluster 1.
 - The non-gatekeeper controlled trunk has Cluster 1's remote servers B, C, and D configured as destinations.

Figure 6-17 H.323 Non-Gatekeeper Intercluster Trunks Using Standard Unified CM Groups

Using Run on All Active Unified Nodes with H.323 Non-Gatekeeper Intercluster Trunks

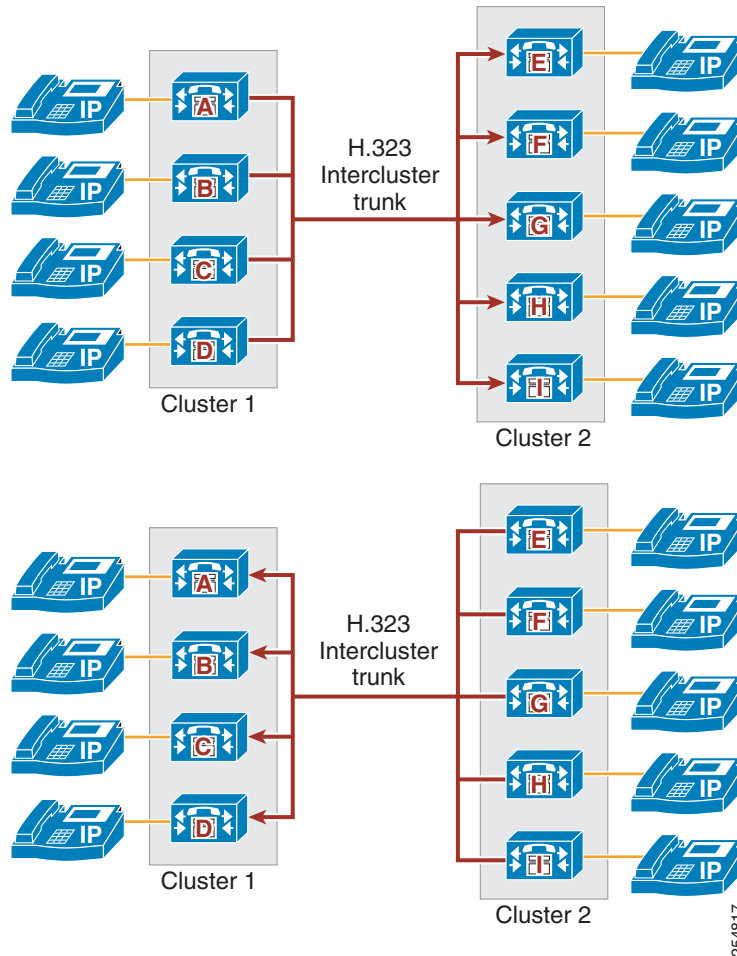
In this type of deployment, **Run on all Active Unified CM Nodes** is used by the H.323 non-gatekeeper intercluster trunks in each cluster. When defining this type of trunk, you may define up to 16 remote Unified CM servers in the destination cluster. (The number of remote servers that you need will depend on the number of active Unified CM nodes in the destination cluster.) The trunk will automatically load-balance calls across all defined remote destination Unified CM servers. In the remote cluster, it is important to configure a corresponding intercluster trunk (non-gatekeeper controlled) that has **Run on all Active Unified CM Nodes** configured, where these nodes are defined as the remote destination Unified CM servers in the first cluster.

For example, if Cluster 1 (four nodes) has a trunk to Cluster 2, and Cluster 2 (five nodes) has a trunk to Cluster 1, the following configurations would be needed (see Figure 6-18):

- Cluster 1 has four active Unified CM nodes (A, B, C, and D).
 - Enabling **Run on all active Unified CM Nodes** causes servers A, B, C, and D to have active H.323 trunk daemons associated with the non-gatekeeper controlled trunk to Cluster 2.
 - The non-gatekeeper controlled trunk has Cluster 2's remote servers E, F, G, H, and I configured as destinations.

- Cluster 2 has five active Unified CM nodes (E, F, G, H, and I).
 - Enabling Run on all active Unified CM Nodes causes servers E, F, G, H, and I to have active H.323 trunk daemons associated with the non-gatekeeper controlled trunk to Cluster 2.
 - The non-gatekeeper controlled trunk has Cluster 1's remote servers A, B, C, and D configured.

Figure 6-18 H.323 Non-Gatekeeper Intercluster Trunks Using Run on All Active Unified Nodes



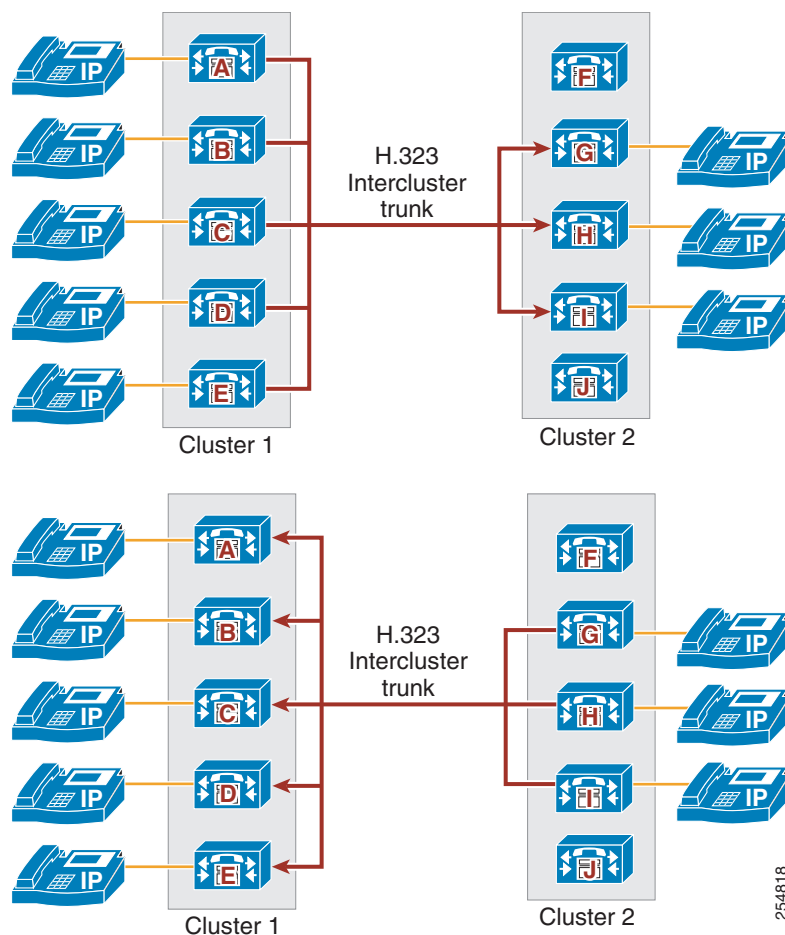
Using Standard Unified CM Groups and Run on All Active Unified CM Nodes with H.323 Non-Gatekeeper Intercluster Trunks

In this type of deployment, **Run on all Active Unified CM Nodes** is used by the H.323 non-gatekeeper intercluster trunk in one cluster, and standard Unified CM Groups are used by the H.323 non-gatekeeper intercluster trunk in the other cluster. When configuring these trunks, the number of remote Unified CM server destinations that you define should match the number of active Unified CM nodes for the corresponding trunk in the destination cluster. The trunk will automatically load-balance calls across all defined remote destination Unified CM servers. In the remote cluster, it is important to configure a corresponding intercluster trunk (non-gatekeeper controlled) that has Unified CM nodes with active H.323 daemons where these nodes are defined as remote destination Unified CM servers in the first cluster.

For example, if Cluster 1 has a trunk to Cluster 2, and Cluster 2 has a trunk to Cluster 1, the following configurations would be needed (see Figure 6-19):

- Cluster 1 has five active Unified CM nodes (A, B, C, D, and E).
 - Enabling **Run on all Active Unified CM Nodes** causes servers A, B, C, D, and E to have active H.323 trunk daemons associated with the non-gatekeeper controlled trunk to Cluster 2.
 - The non-gatekeeper controlled trunk has Cluster 2's remote servers G, H, and I configured as destinations.
- Cluster 2 has five active Unified CM nodes and uses an intercluster trunk with a Unified CM Group containing nodes G, H, and I.
 - Servers G, H, and I are configured as members of the Unified CM Group defined in the device pool associated with the non-gatekeeper controlled trunk to Cluster 1.
 - The non-gatekeeper controlled trunk has Cluster 1's remote servers A, B, C, D, and E configured as destinations.

Figure 6-19 *H.323 Non-Gatekeeper Intercluster Trunks Using Standard Unified CM Groups and Run on All Active Unified CM Nodes*



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High Availability for Non-Gatekeeper Controlled Intercluster Trunks

High availability and redundancy for H.323 non-gatekeeper intercluster trunks can be provided by using multiple source Unified CM servers for originating calls and multiple destination IP addresses per trunk.

Multiple Source Unified CM Servers for Originating H.323 Non-Gatekeeper Intercluster Trunk Calls

- Using standard Unified CM Groups

The nodes defined in the Unified CM Group associated with an individual trunk make up the set of servers that can place or receive calls over that trunk. Up to three nodes can be defined in a Unified CM Group, thus ensuring high availability of the trunk itself.

- Using **Run on all Active Unified CM Nodes**

The **Run on all Active Unified CM Nodes** feature creates and enables an H.323 trunk instance on each call processing subscriber within the cluster, thus allowing these nodes to place or receive calls over that trunk.

- The Unified CM Route Local feature and its effect of subscriber selection for outbound H.323 non-gatekeeper intercluster trunk calls

The Route Local feature in Unified CM is designed to reduce intra-cluster traffic. The feature operates as follows: When a device such as a phone is making an outbound call over H.323 intercluster trunk ICT 1, if an instance of H.323 ICT 1 is active on the same node as the one to which the phone is registered, then always use this co-located H.323 ICT 1 instance rather than internally route the call to another H.323 ICT 1 instance on another node within the cluster.

The effect of the Route Local feature on node selection depends on whether Unified CM Groups or **Run on all Active Unified CM Nodes** is configured on the trunk. For trunks with **Run on all Active Unified CM Nodes** configured, the node to which the calling device is registered is used to make the outbound H.323 intercluster trunk call. When Unified CM Groups are used on the trunk, if the calling device is registered to one of the nodes in the trunk's Unified CM Group, then the Route Local rule applies. If the calling device is not registered to one of the nodes in the trunk's Unified CM Group, then Unified CM will randomly distribute the call over the nodes in the trunk's Unified CM Group.

In general, using **Run on all Active Unified CM Nodes** is recommended for H.323 intercluster trunks because call distribution across nodes is determined by the calling device and intra-cluster traffic is minimized.

Multiple Destination IP Addresses per H.323 Non-Gatekeeper Intercluster Trunks

A single H.323 non-gatekeeper intercluster trunk can be configured with up to 16 destination IP addresses. Unified CM uses round-robin distribution to the configured destination IP addresses when placing calls over an H.323 non-gatekeeper intercluster trunk.

Design Considerations When Using Run on All Active Unified CM Nodes

When using **Run on All Active Unified CM Nodes** in conjunction with multiple destination addresses, be aware that to accept inbound calls, the inbound source IP address received on the H.323 trunk must match with a configured destination IP address on the inbound trunk. Where clustering over the WAN designs are deployed and geographic call distribution and failover are required, use standard Unified CM Groups on multiple intercluster trunks (each with up to three destination IP addresses) in conjunction with route lists and route groups.

Load Balancing for H.323 Non-Gatekeeper Intercluster Trunks

When designing load balancing for H.323 non-gatekeeper intercluster trunks, consider both the node that sources the call and its destination. With H.323 non-gatekeeper intercluster trunks, the node that originates the call is determined by the Route Local rule, the number of nodes on which the outbound trunk is active, and whether a route list is used in conjunction with multiple outbound trunks. These considerations are discussed below.

Outbound Calls over a Single H.323 Non-Gatekeeper Intercluster Trunk

A single H.323 non-gatekeeper intercluster trunk can run on up to three Unified CM nodes in a Unified CM Group or on all active Unified CM nodes in the cluster. To select the source node for outbound calls, Unified CM applies the following decision processes:

Where an instance of the trunk runs on all nodes, the Route Local rule applies and the node used for each outbound call is determined by the node on which the call arrives (for example, the node to which the calling phone is registered or the node on which the inbound trunk call arrives). Where Unified CM Groups are used, the route local rule still applies for those calling devices that are registered to the same node as the nodes in the trunk's Unified CM Group. For calling devices that are registered to other servers within the cluster, Unified CM will randomly distribute calls across the nodes in the trunk's Unified CM Group. Unified CM uses round-robin call distribution across the trunk's configured destination addresses. H.323 non-gatekeeper intercluster trunks may be configured with up to 16 destination IP addresses.

Outbound Calls over Multiple H.323 Non-Gatekeeper Intercluster Trunks

Because H.323 non-gatekeeper intercluster trunks can run on all active Unified CM nodes and have up to 16 destination addresses, you typically do not have to use multiple H.323 non-gatekeeper intercluster trunks to provide even call distribution between two Unified Communications clusters. Where multiple trunks are used with route lists and route groups, route lists should be enabled to run on all active Unified CM nodes. Multiple H.323 trunks are often used in conjunction with route lists to provide failover to the PSTN or to a group of Unified CM servers in a different site as part of a clustering over the WAN deployment. The selection of the Unified CM node used to initiate an outbound trunk call and the distribution of calls over the trunk's configured destination IP addresses is determined in the same way as described for single trunks. Where clustering over the WAN designs are deployed and geographic call distribution and failover are required, use multiple intercluster trunks (each with up to three destination IP addresses) with standard Unified CM Groups in conjunction with route lists and route groups.

Intercluster Trunk (Gatekeeper Controlled)

The intercluster gatekeeper controlled trunk can be used instead of the non-gatekeeper controlled trunk to interconnect a large number of Unified CM clusters. The advantages of using the gatekeeper controlled trunk are mainly the overall administration of the cluster and failover times. With non-gatekeeper controlled trunks, if a subscriber server in a cluster becomes unreachable, there will be a 5-second (default) timeout while the call is attempted. If an entire cluster is unreachable, the number of attempts before either call failure or rerouting over the PSTN will depend on the number of remote servers defined for the trunk and on the number of trunks in the route list or route group (if any). If there are many remote servers and many non-gatekeeper controlled trunks, the call delay can become excessive.

With a H.323 gatekeeper controlled trunk, you configure only one trunk that can then communicate by means of the gatekeeper with all other clusters registered to the gatekeeper. If a cluster or subscriber becomes unreachable, the gatekeeper automatically directs the call to another subscriber in the cluster or rejects the call if no other possibilities exist, thus allowing the call to be rerouted over the PSTN (if

required) with little incurred delay. With a single Cisco gatekeeper, it is possible to have 100 clusters all registering a single trunk each, with all clusters being able to call each other. The gatekeeper controlled intercluster trunk should be used for communicating only with other Unified CMs because the use of this trunk with other H.323 devices might cause problems with supplementary services.

**Note**

Gatekeeper controlled trunks do not support the **Run on All Active Unified CM Nodes** feature, and only standard Unified CM Groups are supported. Destination addresses are returned to Unified CM by the gatekeeper. Where gatekeeper controlled trunks are used in route lists, Cisco recommends enabling the **Run on All Active Unified CM Nodes** feature on the route list.

H.225 Trunk (Gatekeeper Controlled)

The H.225 gatekeeper controlled trunk is essentially the same as the intercluster gatekeeper controlled trunk except that it has the capability of working with Unified CM clusters as well as other H.323 devices such as gateways, conferencing systems, and clients. This capability is achieved through a discovery mechanism on a call-by-call basis. (See [H.323 Operation in Unified CM, page 6-53](#), for details of this discovery process.)

**Note**

Gatekeeper controlled trunks do not support the **Run on All Active Unified CM Nodes** feature, and only standard Unified CM Groups are supported. Destination addresses are returned to Unified CM by the gatekeeper. Where gatekeeper controlled trunks are used in route lists, Cisco recommends enabling the **Run on All Active Unified CM Nodes** feature on the route list.

High Availability for Gatekeeper Controlled Trunks

Redundancy can be achieved in several ways, depending on the requirements of the design. The simplest method is to configure a gatekeeper controlled trunk and assign up to three subscribers in the Unified CM Group associated with the device pool assigned to that trunk. This configuration will cause all servers to register with the same gatekeeper in the same zone with the same technology prefix. However, the H.323 trunk name that is used for the h323_id will have a suffix of "_n" where *n* is the node number in the cluster. This ID is automatically generated and cannot be changed. You configure a single trunk, but the gatekeeper registers multiple trunks, one from each subscriber in the Unified CM Group.

If you have additional redundancy requirements, it is possible to configure another gatekeeper controlled trunk with a different name and different subscribers in the Unified CM Group, but with all the other parameters identical to the first trunk. This second trunk will cause additional subscribers to register with the gatekeeper.

Cisco recommends assigning device pools that contain a Unified CM Group consisting of the two servers that make up the standard subscriber pair. (See [Call Processing Subscriber Redundancy, page 9-20](#), for more information on subscriber redundancy.) For complete redundancy in each full cluster, four trunks would be needed, using four different device pools and resulting in eight subscribers registering with the gatekeeper. (The same result could be achieved with three trunks and larger Unified CM Groups.)

During registration, several parameters are passed between Unified CM and the gatekeeper. Unified CM uses an ephemeral User Datagram Protocol (UDP) port for gatekeeper Registration Admission Status (RAS) messages. This port would normally be UDP 1719. However, Unified CM must be able to identify precisely which H.323 daemon is the originator of a RAS message from a particular server; therefore it uses a range of UDP ports and assigns them dynamically.

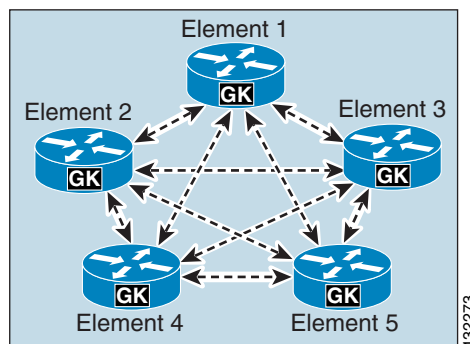
During the registration process, a trunk registers the following information for the other subscribers in its Unified CM Group:

- H.225 call signaling port
- h323_id
- CanMapAlias support
- Technology prefix
- H.225 call signaling address

If the recommended clustered gatekeepers are used, the gatekeeper will return a list of alternate gatekeeper addresses that may be used if the primary gatekeeper fails or does not have sufficient available resources.

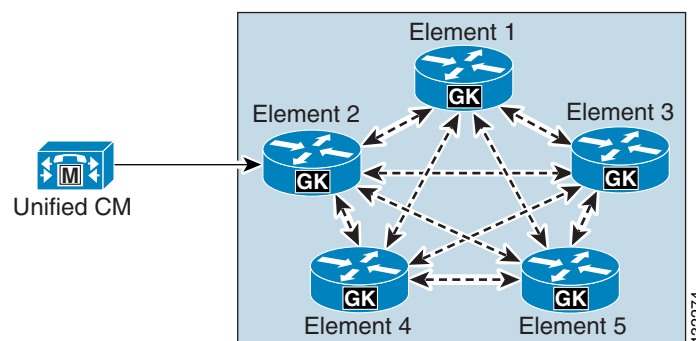
Figure 6-20 shows a cluster of gatekeepers that use Gatekeeper Update Protocol (GUP) to communicate. (See the chapter on [Call Processing](#), page 9-1, for more information on gatekeepers.)

Figure 6-20 Gatekeeper Cluster

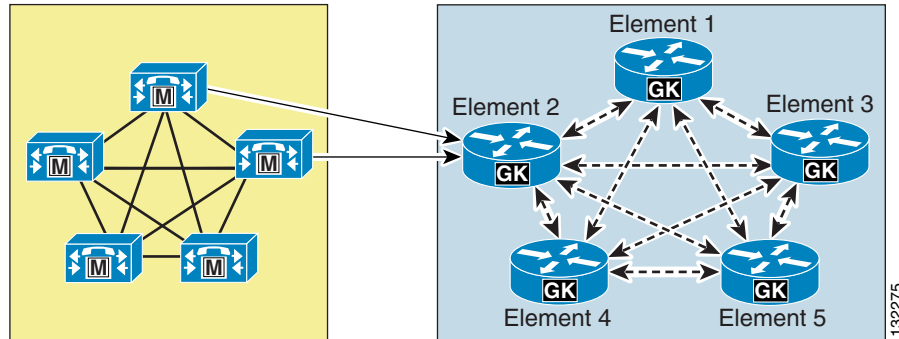


If an H.323 trunk has only a single subscriber in its Unified CM Group, there will be only one connection between the configured gatekeeper in Unified CM and the gatekeeper cluster, as illustrated in Figure 6-21.

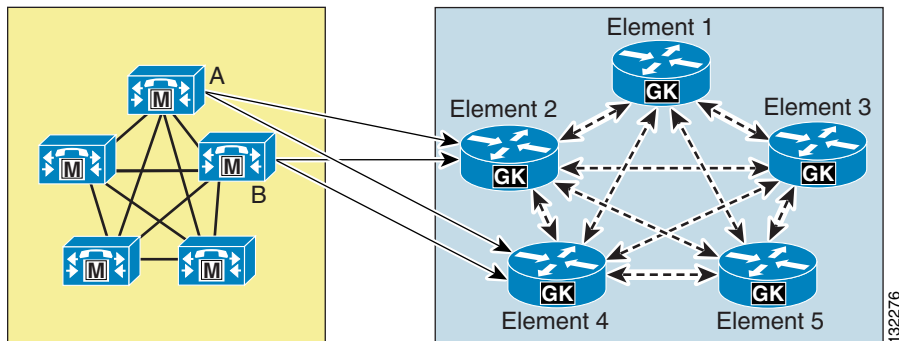
Figure 6-21 H.323 Trunk with a Single Unified CM Subscriber



If there are multiple subscribers in the Unified CM Group associated with the trunk, additional connections will be established between the Unified CM cluster and the gatekeeper cluster, as illustrated in Figure 6-22.

Figure 6-22 H.323 Trunk with Multiple Unified CM Subscribers

This approach provides redundancy for subscriber failures as well as gatekeeper failures after registration because the alternate gatekeeper is communicated when the trunk registers. This approach does not, however, provide redundancy if the configured gatekeeper is unavailable at initial registration or following a reset because the list of alternate gatekeepers is dynamic and not stored in the database. To provide an additional level of redundancy as well as load balancing, an additional gatekeeper from the gatekeeper cluster is configured in Unified CM. For example, if the original trunk is registered with Element 2, the additional gatekeeper could be configured as Element 4, as illustrated in [Figure 6-23](#).

Figure 6-23 Additional Gatekeeper Configured for Load Balancing and Additional Redundancy

The Unified CM configuration for the example in [Figure 6-23](#) would contain the following components:

- Two gatekeepers for Element 2 and Element 4
- Two H.323 trunks defined with a Unified CM Group containing subscriber servers A and B

Using this approach, the Unified CM cluster will still be able to register when either Element 2 or Element 4 is not reachable during initial registration (that is, during power-up or trunk reset).

Load balancing of calls inbound to the Unified CM cluster is done automatic by default because the gatekeeper randomly selects one of the subscribers registered within the zone. If this is not the desired behavior, you can use the **gw-priority** configuration command in the gatekeeper to modify this default behavior, as illustrated in [Example 6-3](#).

Example 6-3 Using the gw-priority Command to Direct Calls to a Particular Trunk

```

gatekeeper
zone local SJC cisco.com 10.0.1.10
zone prefix SJC 1408..... gw-priority 10 sjc-trunk_2
zone prefix SJC 1408..... gw-priority 9 sjc-trunk_3
zone prefix SJC 1408..... gw-default-priority 0
gw-type-prefix 1#* default-technology
arg reject-unknown-prefix
no shutdown
endpoint ttl 60

```

In [Example 6-3](#), the H.323 trunk was configured as sjc-trunk in Unified CM, and the “_2” and “_3” suffixes are appended automatically by the Unified CM subscribers to indicate which node number they are in the cluster. Therefore, this example uses node 2 as the first choice, which should be the highest-priority Unified CM in the Unified CM Group for this trunk. Node 3 is the second choice in this case.

The use of **gw-default-priority 0** is optional. It was used in this example to disable the use of any other trunk that might accidentally be configured to register in this zone.

Load Balancing Outbound Calls over H.323 Gatekeeper Controlled Trunks

With Unified CM H.323 gatekeeper controlled trunks, the node from which the call originates is determined by the Route Local rule, the number of nodes on which the outbound trunk is active, and whether a route list is used in conjunction with multiple outbound trunks. These considerations are discussed below.

Outbound Call Load Balancing When Deploying a Single H.323 Gatekeeper Controlled Trunk

For the initiation of outbound calls over a single H.323 gatekeeper controlled trunk, the route local rule applies and the following factors within a Unified CM cluster determine which server is selected:

- Which Unified CM servers have an active H.323 daemon for the selected trunk
- Whether the phone originating the call is registered to a Unified CM server with an active H.323 daemon for the selected trunk

For a single H.323 gatekeeper controlled trunk, the Route Local process of server selection for outgoing calls operates as follows:

- If there is an active H.323 daemon for the selected trunk on the Unified CM server to which the phone or device originating the call is registered (that is, if the server is one of those listed in the trunk's Unified CM Group), then use this Unified CM server to originate the H.323 call.
- If there is no active H.323 daemon for the selected trunk on the Unified CM server to which the phone or device originating the call is registered, then select a server on a round-robin basis from the Unified CM Group of the selected trunk.

Outbound Call Load Balancing When Deploying Route Lists in Conjunction with H.323 Gatekeeper Controlled Trunks

In configurations where route lists are employed to select a trunk for outbound calls, enable **Run on all Active Unified CM Nodes** for all route lists. Running route lists on all nodes improves outbound call distribution by using the Route Local rule to avoid unnecessary intra-cluster traffic. For route lists, the Route Local rule operates as follows:

For outbound calls that use route lists (and associated route groups and trunks), when a call (from a registered phone or inbound trunk) arrives at the node with the route list instance associated with the outbound trunk call, Unified CM checks to see if an instance of the selected outbound trunk call exists on the same node as the route list. If so, Unified CM will use this node to establish the outbound trunk call.

If the route list has **Run on all Active Unified CM Nodes** enabled: For gatekeeper controlled trunks using Unified CM Groups, Unified CM will apply the route local rule if an instance of the selected outbound trunk exists on the same node on which the inbound call arrived. If an instance of the trunk does not exist on this node, then Unified CM will forward the call (within the cluster) to a node where the trunk is active.

If the route list does not have **Run on all Active Unified CM Nodes** enabled, the route list will be active on one node within the cluster (the primary node in the route list's Unified CM Group) and the Route Local rule will be applied on this node.

H.323 Outbound Fast Start Call Connections

Calls that are placed from IP phones over large WAN topologies with long delays can experience voice clipping when the called party goes off-hook to answer the call. When H.323 trunks or gateways are separated from the Unified CM server, significant delays can occur because of the many H.245 messages that are exchanged when a call is set up.

With the Fast Start feature, information that is required to complete a media connection between two parties gets exchanged during the H.225 portion of call setup, and this exchange eliminates the need for H.245 messages. The connection experiences one round-trip WAN delay during call setup, and the calling party does not experience voice clipping when the called party answers the call.

Unified CM uses media termination points (MTPs) for making an H.323 outbound Fast Start call. Unified CM starts an outbound Fast Start call by allocating an MTP and opening the receive channel. Next, the H.323 Fast Connect procedure sends the SETUP message with a Fast Start element to the called endpoint. The Fast Start element includes information about the receiving channel for the MTP.

By default, H.323 Fast Start is disabled. To enable H.323 Fast Start, check the **MTP Required** and **Enable Outbound FastStart** checkboxes on the H.323 trunk, and select the desired Codec For Outbound Fast Start. Also note that Inbound Fast Start is enabled separately with check box **Enable Inbound FastStart**. (Inbound Fast Start does not require an MTP or a codec selection.)



Note

When H.323 Fast Start is enabled, an MTP is assigned for every outbound H.323 trunk call. MTPs used for H.323 Fast Start support a single voice codec only, and therefore video and encrypted calls are not supported. H.323 Fast Start is disabled by default on H.323 trunks, and MTPs are not required for outbound or inbound calls. As a general rule, this default H.323 (Slow Start) trunk configuration is preferred so that voice, video, and encrypted calls are supported over H.323 trunk connections.

H.323 Trunks with Media Termination Points

Media termination points (MTPs) are generally not required for normal operation of the H.323 trunk. They are, however, required for communication with devices that are H.323 Version 1, that do not support the Empty Capabilities Set (ECS) for supplementary services, or that require H.323 Fast Start.

To test whether or not an MTP is required, use the following simple procedure:

1. Place a call from a phone through the H.323 trunk to the other device. This call should work normally.
2. Place the call on hold, then resume it. If the call drops, then it is highly likely that an MTP is required to ensure interoperability between Unified CM and the other device.

DTMF Transport

The H.323 trunk supports DTMF signaling for both out-of-band DTMF using H.245 and in-band DTMF using RTP Named Telephone Events (RFC 2833). There are no configuration options. An MTP may be allocated dynamically to convert between out-of-band DTMF relay to in-band DTMF relay, if required. For an explanation of when the H.323 trunk uses which method and when MTPs are required, refer to the chapter on [Media Resources](#), page 7-1.

H.323 Trunk Transport Protocols

H.323 trunks use TCP for H.225 call control and H.245 media control signaling, and UDP for gatekeeper H.225 Registration Admission Status (RAS) signaling.

Secure H.323 Trunks

Securing H.323 trunks involves two processes: configuring the trunk to encrypt media and configuring the trunk to encrypt signaling.

Media Encryption

Media encryption can be configured on H.323 trunks by checking the trunk's **SRTP allowed** check box. It is important to understand that checking the **SRTP allowed** checkbox will cause the media for calls to be encrypted but the trunk signaling will not be encrypted, therefore the session keys used to establish the secure media stream will be sent in the clear. It is therefore important to ensure that signaling between Unified CM and its destination H.323 trunk device is also encrypted so that keys and other security-related information do not get exposed during call negotiations.

Signaling Encryption

H.323 trunks use IPSec for signaling encryption. You may configure IPSec in the network infrastructure, or you may configure IPSec between Cisco Unified Communications Manager (Unified CM) and the remote gateway or trunk. If you implement one method to set up IPSec, you do not need to implement the other method. Using IPSec on Unified CM servers can incur a significant impact on server performance, therefore Cisco recommends that you provision IPSec in the network infrastructure rather than in Unified CM itself.

If the system can establish a secure media or signaling path and if the end devices support SRTP, the system uses an SRTP connection. If the system cannot establish a secure media or signaling path or if at least one device does not support SRTP, the system uses an RTP connection. SRTP-to-RTP fallback (and vice versa) may occur for transfers from a secure device to a non-secure device, conferencing, transcoding, music on hold, and so on.

For SRTP-configured devices, Unified CM classifies a call as encrypted if the **SRTP Allowed** check box is checked for the device and if the SRTP capabilities for the devices are successfully negotiated for the call. If the preceding criteria are not met, Unified CM classifies the call as non-secure. If the device is connected to a phone that can display security icons, the phone displays the lock icon when the call is encrypted.

MTPs that are statically assigned to an H.323 trunk using the **MTP Required** checkbox do not support SRTP because they do not support the pass-through codec. To ensure that SRTP is supported for all calls, do not configure the H.323 trunk for H.323 Outbound Fast Start (that is, do not select **MTP Required**). SRTP is supported for Inbound Fast Start. (Inbound Fast Start does not require an MTP or a codec selection.)

H.323 Operation in Unified CM

This section provides information on how the H.323 protocol is used and implemented in Unified CM, and it explains how and why certain features work the way they do.

The most important point to understand is which subscribers run the call signaling daemons. These daemons are pieces of code that make and receive H.323 calls. They are usually referred to as H.323 or H.225 daemons (H.323Ds or H.225Ds). H.225 is part of the H.323 protocol and is mainly responsible for call control. H.245 is the other major component of H.323 that is responsible for the media control of a call.

For the majority of H.323 devices, the subscribers listed in the Unified CM Group for a particular H.323 device determine which subscribers run the daemons and when. For H.323 non-gatekeeper controlled intercluster trunks, standard Unified CM Groups can be used or the **Run on All Active Unified CM Nodes** can be enabled, in which case the daemon will run on all active nodes.

For devices using Unified CM Groups, it is important to know which nodes will run H.225 daemons because calls sent to an incorrect subscriber might be rejected. For example, this situation would occur if a Cisco IOS H.323 gateway is configured with dial peers that send calls to subscriber C in a Unified CM cluster but the Unified CM Group for that gateway has only subscribers A and B in its list. In such a case, the call will fail or be handled by an H.323 trunk daemon if one happens to be configured on the subscriber.

The following scenarios describe where and when H.225Ds are created on subscribers:

- H.323 client

The H.225D is active on only the highest-priority subscriber available in the Unified CM Group associated with the H.323 client.

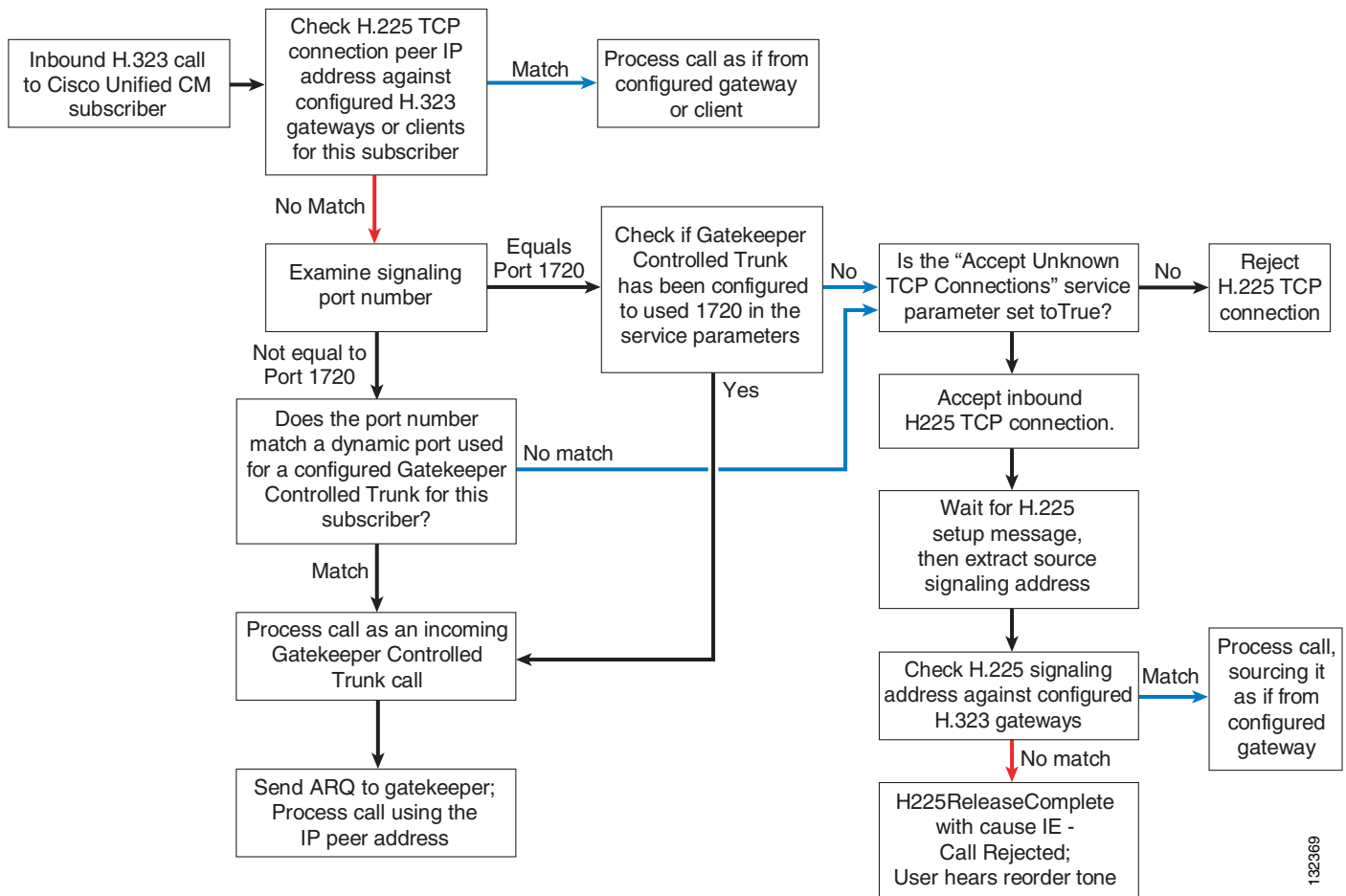
If the H.323 client is gatekeeper controlled, the RasAggregator device registers from only the highest-priority subscriber available in the Unified CM Group associated with the gatekeeper controlled H.323 client.

The RasAggregator is a special device that registers in gatekeeper zones for the purpose of providing two specific features:

- If H.323 clients use DHCP, they cannot be used with a Unified CM using DNS unless they support Dynamic DNS. With the RasAggregator, Unified CM can obtain the IP address of a specific H.323 client that is registered with the gatekeeper whenever a call is placed. The gatekeeper registration is done using standard RAS ARQ messages that contain the E.164 address of the H.323 client. The gatekeeper resolves the E.164 address and provides the IP address back to Unified CM in an ACF message.

- The RasAggregator also ensures that all calls by the H.323 clients are made through Unified CM and not directly between the clients themselves, thus ensuring that dialing rules and codec restrictions are enforced.
- H.323 gateway
The H.225D is active on all subscribers in the Unified CM Group associated with the H.323 gateway.
- H.323 gatekeeper controlled trunks
The H.225D is active on all subscribers in the Unified CM Group associated with the H.323 trunk. A RAS daemon registers the trunk with the gatekeeper from all subscribers in the associated Unified CM Group.
- H.323 non-gatekeeper controlled trunks using Unified CM Groups
The H.225D is active on all subscribers in the Unified CM Group associated with the H.323 trunk.
- H.323 non-gatekeeper controlled trunks using **Run on all Active Unified CM Nodes**
The H.225D is active on all active Unified CM subscribers in the cluster.

When an incoming H.323 call is made to a subscriber in a Unified CM cluster, various decisions are made to determine if the call is accepted or rejected and which H.225D will receive the call if it is accepted. [Figure 6-24](#) shows how this process works.

Figure 6-24 Process for Determining if an H.323 Call is Accepted or Rejected

Unified CM H.323 protocol includes the following additional features:

- Protocol Auto Detect

This feature provides the ability to determine, on a call-by-call basis, if the calling device is from Cisco Unified CM. Whenever a call is received, Unified CM looks for an H.225 User-to-User Information Element (UUIE) that indicates if the other end is another Unified CM. If it is, it will always use the Intercluster Trunk Protocol. If no UUIE is found, it will use the configured protocol for that device. This feature enables an H.225 gatekeeper controlled trunk to switch between Intercluster Trunk Protocol and H.225 on a call-by-call basis, allowing a mixture of Unified CM clusters and other H.323 devices to use the gatekeeper. Intercluster Trunk Protocol is the same as H.225 except for several differences that enable specific features to work correctly between Unified CM clusters.

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- Tunneled QSIG or H.323 Annex M1 (ISO and ECMA variants supported per trunk)

This feature can be enabled on all H.323 trunks. It allows specific H.323 Annex M1 features to be implemented between Unified CM clusters and other verified systems that also support H.323 Annex M1. These features include:

- Path replacement
- Message waiting indication (MWI)
- Callback

- Alternate Endpoints

When registering with a gatekeeper that supports this feature, such as a Cisco Multimedia Conference Manager (MCM) Gatekeeper, Unified CM can inform the gatekeeper of alternate destinations for calls to the H.323 trunk. These alternate endpoints or destinations are sent to the calling device by the gatekeeper when this H.323 trunk is called. They are the other subscribers listed in the Unified CM Group associated with the H.323 trunk that registers with the gatekeeper.

- Alternate Gatekeeper

When an H.323 trunk registers with a gatekeeper that supports this feature (for example, a Cisco gatekeeper cluster), Unified CM is dynamically informed about other gatekeepers that can process registrations, call admission requests, and other RAS functions in the event that this gatekeeper fails or exhausts its own resources.

- CanMapAlias

When an H.323 trunk sends an admission request (ARQ) to the gatekeeper, it might receive a different E.164 number in the admission confirmation message (ACF), indicating that the original called number should be replaced with this new one. This feature requires a route server using Gatekeeper Transaction Message Protocol (GKTMP) to communicate with Cisco gatekeepers.



Note CanMapAlias is supported for the called number only.

- Bandwidth Requests

H.323 trunks can update the gatekeeper with bandwidth information to indicate a change in the requested bandwidth allocated to a specific call. This feature is disabled by default and is controlled by setting the Unified CM service parameter **BRQ Enabled** to **True**, under the H.323 section. This feature is especially important when video is used on an H.323 trunk because the original bandwidth request is for the maximum amount allowed. Enabling this feature ensures that call admission control uses the actual bandwidth negotiated during call setup.

Other Design Considerations for H.323 Trunks

Unified CM SIP trunks now offer a greater set of features in comparison with H.323 intercluster trunk, making SIP the protocol of choice for intercluster trunk connections, although H.323 Annex M1 may still be preferred for intercluster trunk connections to Unified CM clusters using earlier software versions. For more information on deploying intercluster trunks in multi-cluster and clustering over the WAN environments, see [Design Considerations for SIP Trunks, page 6-29](#).

General SIP and H.323 Trunk Design Considerations

This section covers the following general design considerations:

- [Deterministic Outbound Call Load Balancing over Unified CM Trunks, page 6-57](#)
- [Codec Selection Over IP Trunks, page 6-58](#)
- [Other MTP Uses, page 6-62](#)

Deterministic Outbound Call Load Balancing over Unified CM Trunks

In the majority of cases, using **Run on all Active Unified CM Nodes** or assigning Unified CM Groups to devices is sufficient to handle the call distribution of outbound calls over trunks from call processing subscribers. Due to the Route Local rule, trunk calls might appear to originate randomly from call processing subscribers, but the trade-off for this random call origination is reduced call processing and reduced Intra-Cluster Communication Signaling (ICCS) traffic within the cluster.

Deterministic load balancing of outbound IP trunk calls across call processing servers is possible but can be counter-productive because the advantages gained from predictable call origination within the cluster can be outweighed by the increase in ICCS traffic created by calls from phones registered to one subscriber extending their communication to another server within the cluster to originate the outgoing IP trunk call.

Predictable and deterministic subscriber-based load balancing of outgoing IP trunk calls can be achieved as follows:

- To deterministically load-balance outbound trunk calls across a subset of the call processing servers in the cluster, define multiple trunks and assign only a single subscriber to the Unified CM Group of each trunk. Place these trunks into a route group and use circular call distribution.

For example, to spread outbound trunk calls across four subscribers in the cluster, perform the following tasks:

- Configure four H.323 trunks or four SIP trunks with individual Unified CM Groups, all contained within a route group with circular call distribution.
- Define Unified CM Groups as follows:
 - Group A: Subscriber A
 - Group B: Subscriber B
 - Group C: Subscriber C
 - Group D: Subscriber D

With no backup subscribers defined, if the primary subscriber for the specified trunk fails, Unified CM will re-route outgoing calls to the next trunk in the route group.

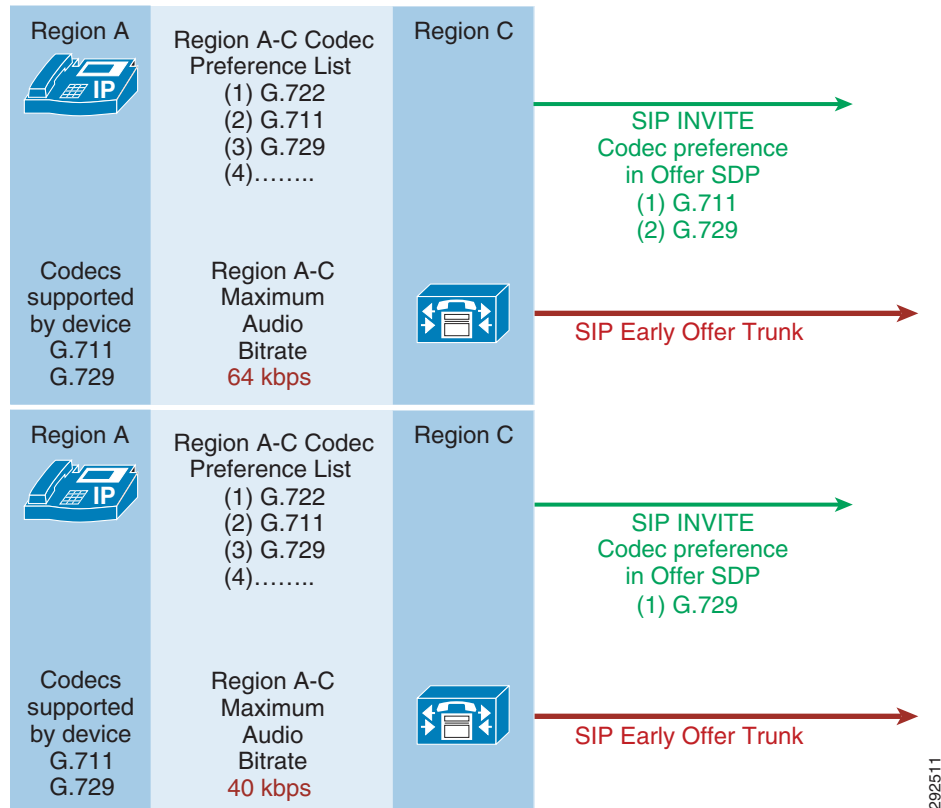
- To spread outbound trunk calls across all eight subscribers in a cluster, perform the following tasks:
 - Configure eight H.323 trunks or eight SIP trunks with individual Unified CM Groups, each containing only one subscriber and all contained within a circular route group.
 - Define Unified CM Groups as follows:
 - Group A: Subscriber A
 - Group B: Subscriber B
 - Group C: Subscriber C
 - Group D: Subscriber D
 - Group E: Subscriber E
 - Group F: Subscriber F
 - Group G: Subscriber G
 - Group H: Subscriber H

Codec Selection Over IP Trunks

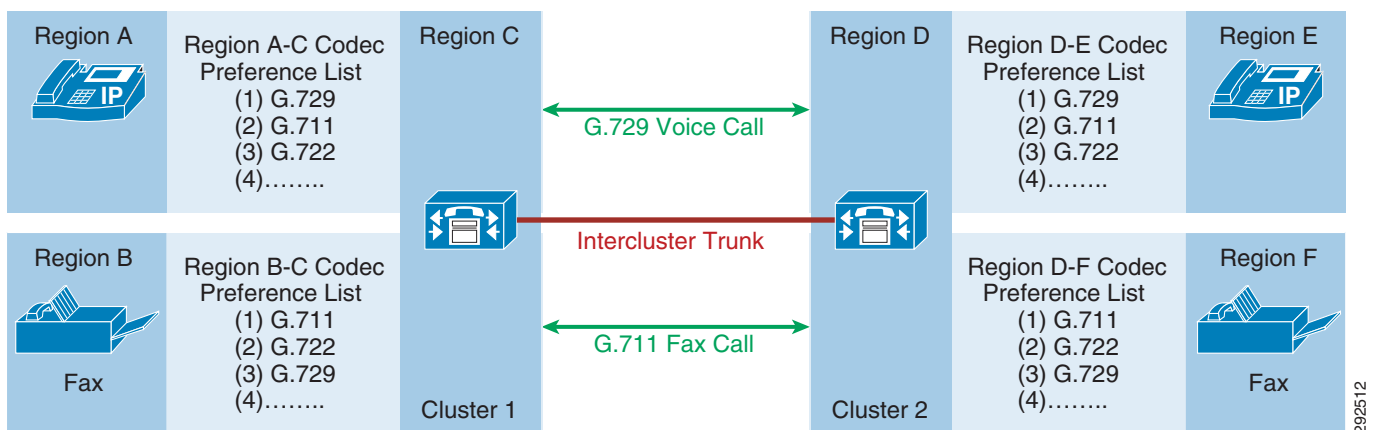
Before media can be established between communicating entities, both the entities must agree on the codec(s) that they want to use. This codec (or codecs if both audio and video are involved) is derived from the intersection of codecs supported by communicating entities involved and the configured policy in Unified CM. Policy in Unified CM is configured by region settings.

The region settings in Unified CM 9.x provide for configurable audio codec preference lists. In addition to the default Lossy and Low Loss audio codec preference lists that can be selected via a region's Link Loss Type, multiple custom audio codec preference lists can also be configured. Audio codec preference lists can be used for codec selection for calls within a region and between regions. The **Maximum Audio Bit rate** is still applied for calls within a region and between regions; but rather than using the highest audio quality codec (as in earlier Unified CM releases) based on the maximum bit rate setting, the codec selection is made based on the codec order in the audio codec preference list and the codecs that the endpoints support. (See [Figure 6-25](#).)

The Audio Codec Preference List is a list of all the codec types supported by Unified CM. The preference order of this list of codecs can be modified and saved as a custom preference list. (Note that codecs cannot be removed from the Audio Codec Preference List). The list of codecs used for codec negotiation during call setup is the subset of codecs supported by the device and those in the codec preference list, limited by the maximum audio bit rate for the region or region pair.

Figure 6-25 Examples of How Codecs Are Selected for Codec Negotiation During Call Setup

For calls between two Unified CM clusters via SIP or H.323 intercluster trunks, audio codec preference lists allow the codec to be selected for a call based upon the codec preferences of the calling and called devices. By grouping devices in each cluster into regions based on their codec preferences, a single intercluster trunk can be used to support multiple calls, each using a specific codec. (See [Figure 6-26](#).)

Figure 6-26 Audio Codec Preference Lists for Voice and Fax Calls Between Two Unified CM Clusters

**Note**

Equivalent audio codec preference lists for each device type region should be configured in each cluster to ensure that a common codec is selected for each device type, irrespective of call direction or trunk configuration. If the audio codec preference lists in each cluster are not equivalent, the codecs used per call may vary based on call direction and trunk configuration. (Ordinarily, the codec preference order is not honored by the cluster receiving the codec preference list.)

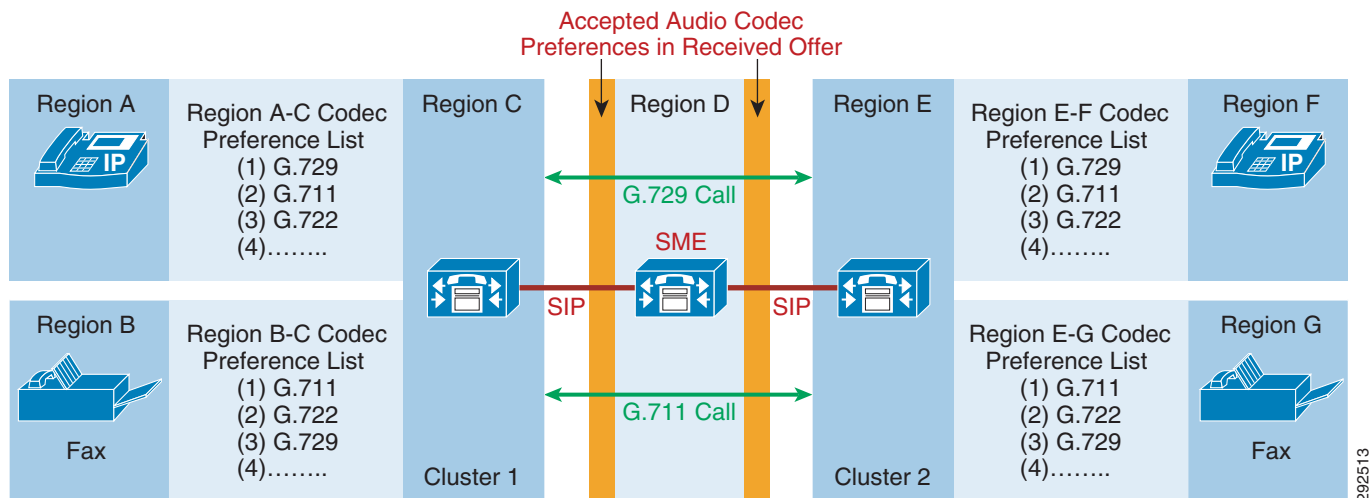
**Note**

Avoid using SIP trunks configured for Early Offer using **MTP Required** or H.323 trunks with Fast Start enabled. These trunk configurations insert an MTP for outbound calls, which limits calls to a single audio codec only, thereby overriding codec preference and selection. In these cases, the maximum audio bit rate setting must also be configured appropriately to allow this codec to be used.

Accept Audio Codec Preferences in Received Offer

In deployments where calls can pass through more than one Unified CM cluster (for example, SME deployments), the inter-region audio codec preference list of the intermediary Unified CM cluster can override the preferred codec selection between the calling and called device. To ensure that the endpoints' codec preferences are honored as calls pass through SME, enable the SIP Profile feature **Accept Audio Codec Preferences in Received Offer** on all SME SIP trunks. (See [Figure 6-27](#).)

Figure 6-27 SME Deployment Using "Accept Audio Codec Preferences in Received Offer" on SIP Trunks

**Note**

The **Accept Audio Codec Preferences in Received Offer** feature is available only on SIP trunks (a SIP Profile feature). This feature does not offer consistent results if used in an SME deployment where the SME cluster uses a combination of SIP, H.323, and/or MGCP trunks. Therefore, the **Accept Audio Codec Preferences in Received Offer** feature should be used when the SME cluster is deployed using only SIP trunks.

**Note**

For Unified CM 8.5 and later releases, Cisco recommends the use of SIP trunks instead of H.323 and MGCP trunks because SIP offers the richest feature set. (See [Table 6-2](#) for a comparison of SIP and H.323 intercluster trunks. For a comparison of features for SIP, H.323, and MGCP trunks, see the *Cisco Unified Communications Manager Session Management Edition Deployment Guide*, available at http://www.cisco.com/en/US/products/ps10661/products_implementation_design_guides_list.html.)

Cisco Unified CM and Cisco Unified Border Element SIP Trunk Codec Preference

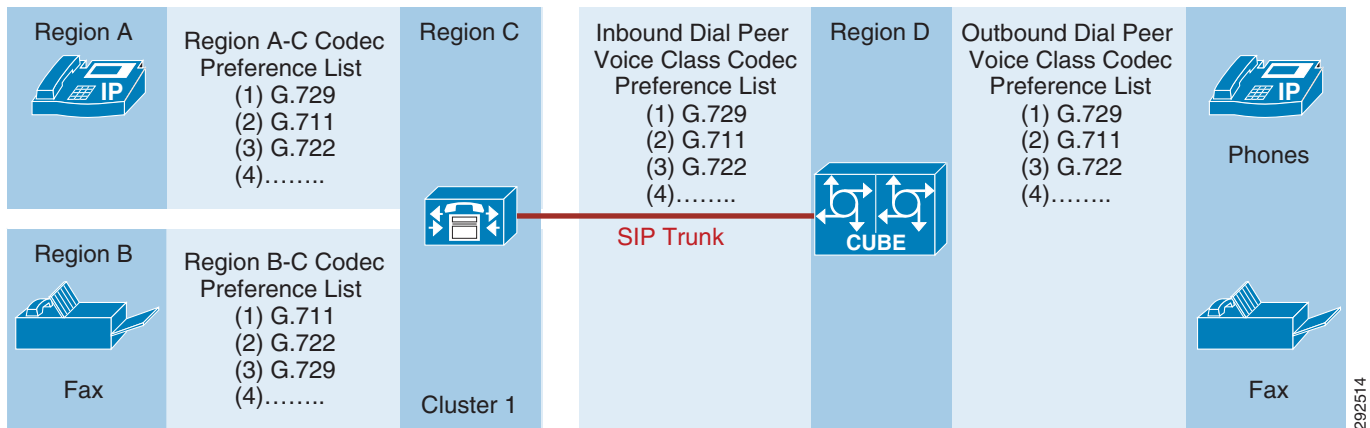
Unified CM audio codec preference lists can be used in Unified Communications deployments with Cisco Unified Border Element (CUBE) to simplify Unified CM-to-CUBE SIP trunk configuration. For example, instead of using dedicated SIP trunks to CUBE for voice and fax calls, a single Unified CM SIP trunk can be used where the codec preference for each device type is honored as calls pass through CUBE.

In [Figure 6-28](#), the Voice Class Codec Preference lists defined on CUBE's inbound and outbound dial peers do not change the preference of the listed codecs in the received Offer. CUBE does codec filtering on the received Offer, both on the inbound and outbound dial-peer, and passes across the common codecs in the same preference order as received in the inbound Offer to the peer leg.

If codecs, in addition to those received in an Offer, are defined in the voice class codec list, then these codecs will be appended to those received in the ordered list and sent out in the outbound Offer.

Thus, a single inbound and outbound dial-peer can be configured on CUBE for all device types. Cisco recommends using the same voice class codec preference list for both the inbound and outbound dial-peer, with that list containing the codecs that you want to negotiate with the service provider. As mentioned above, the order of the codecs will be dictated first by the order received in the inbound Offer and then by the order defined in the voice class codec preference list.

Figure 6-28 Cisco Unified CM and Cisco Unified Border Element SIP Trunk Codec Preference



Other MTP Uses

MTPs are very useful for terminating media streams from other devices that make calls over trunks and for re-originating the media streams with the same voice payload; however, in such cases the IP address is changed to that of the MTP. With this fact in mind, you can utilize MTPs in the following scenarios:

- If the phones, gateways, and other devices within your enterprise all use RFC 1918 private addresses, you might still want to connect to other systems on a public network without using Network Address Translation (NAT) for all your voice and video devices. If the Unified CM subscriber that communicates to the public network is using a public IP address, the signaling will be routed. If all MTPs are also using public addresses, the media from the devices with RFC 1918 addresses will be terminated on the MTP and then originated again, but this time with a public address that is routable on the public network. This approach allows tens of thousands of devices with RFC 1918 addresses to communicate with the public network. This same method can be used to conceal the real IP addresses of devices in an enterprise network when communicating with other enterprises or service providers.
- Trust boundaries can be established to traverse firewalls or to allow access through an access control list (ACL). Normally, for media to traverse a firewall, you could either use an Application Layer Gateway (ALG) or fix-up to provide access dynamically for the media streams or you could allocate a wide range of addresses and ports for use by all voice devices that need to communicate across the firewall. All calls that use a trunk and traverse a firewall or ACL will have media that is sourced from the MTP(s), which may use either a single IP address or a small range of IP addresses.

With both of these methods, if the **MTP Required** box is checked, the default behavior is to allow calls on SIP and H.323 trunks even if MTP resources are unavailable or exhausted. This default behavior might result in no voice path for the call, but the behavior can be changed by setting the Unified CM service parameter **Fail Call if MTP allocation fails** under the SIP and H.323 sections to **True**.

Cisco Unified CM Trunks and Emergency Services

IP trunks might be unable to deliver emergency 911 calls or, like centralized PSTN trunks, might be unable to deliver such calls to the appropriate Public Safety Answering Point (PSAP) for the caller's location. Customers must investigate carefully the capabilities of the IP trunk service provider to deliver emergency 911 calls and caller locations to the appropriate PSAP. Cisco Emergency Responder may be used to provide the location-specific calling party number to the IP trunk service provider for emergency 911 calls.

Centralized IP or PSTN trunks might also temporarily become unavailable for emergency 911 calls from remote locations due to WAN congestion or failure. For this reason, remote locations should always have local gateways to the PSTN that are capable of delivering emergency 911 calls. For more information, see [Emergency Services, page 15-1](#).

Capacity Planning for Unified CM IP Trunks

Cisco 7800 Series Media Convergence Servers support the following trunk capacities:

- An MCS-7845 cluster or Cisco Unified Computing System (UCS) equivalent cluster can support up to 2100 trunks.
- An MCS-7835 cluster can support up to 1100 trunks.
- An MCS-7825 cluster can support up to 1100 trunks.
- An MCS-7816 cluster can support up to 200 trunks.

While the above values represent the nominal maximum capacities, actual trunk scalability and performance ultimately depend on several factors including all other applications and tasks that the individual subscribers are processing, the busy hour call attempts (BHCA) across the trunks, and so forth. To determine the overall system capacity, use the Cisco Unified Communications Sizing Tool (Unified CST), which is available to Cisco employees and partners with proper login authentication at

<http://tools.cisco.com/cucst>

To obtain the most trunk throughput from a cluster, ensure that the trunk load for both incoming and outgoing calls is distributed uniformly over all of the subscribers in the cluster as much as possible.

Interconnecting Unified CM and Cisco VCS

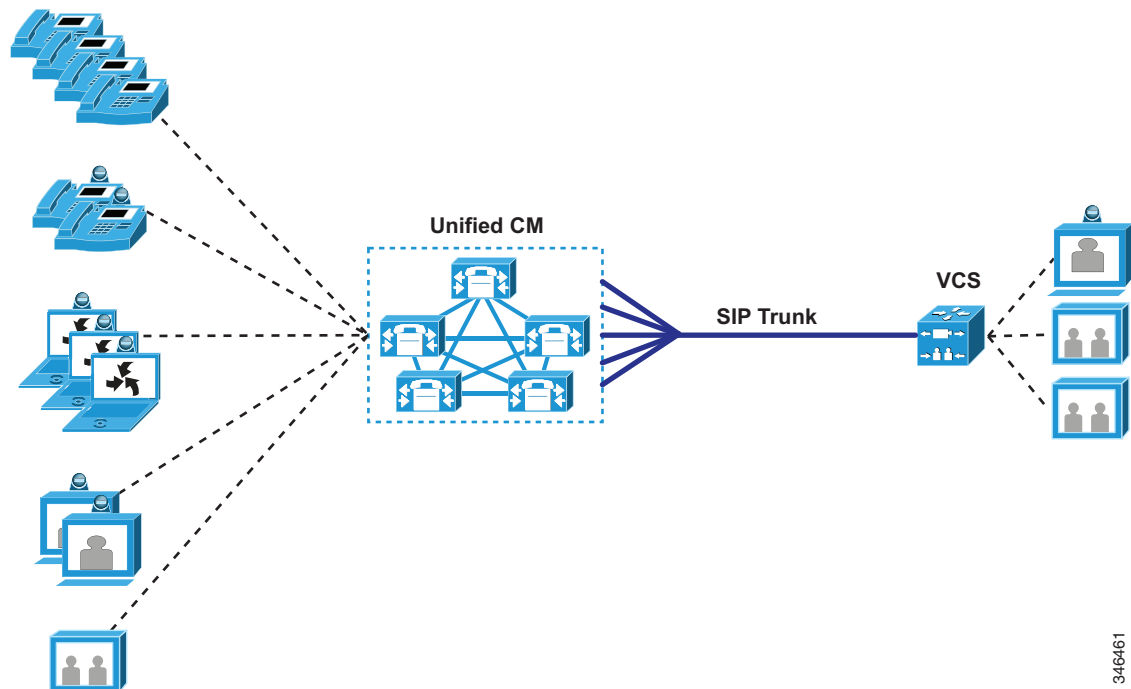
Integration of Cisco Unified Communications Manager (Unified CM) and Cisco TelePresence Video Server (VCS) supports the following features:

- Voice calls
- Video calls
- Encrypted calls
- Desktop and presentation sharing using Binary Floor Control Protocol (BFCP)

Use a SIP trunk to interconnect the Unified CM cluster with the VCS server(s), as illustrated in [Figure 6-29](#). As a general recommendation, SIP Delayed Offer is preferred because no MTPs are required during call setup. SIP Early Offer using **SIP Early Offer for Voice and Video (Insert MTP if needed)** can also be used, but bear in mind that if an MTP is inserted, then only voice is supported during the initial call setup. (For more information on SIP Early Offer, see the section on [Early Offer Support for Voice and Video Calls \(Insert MTP If Needed\)](#), page 6-9.

For detailed information on Cisco VCS and Unified CM trunk configuration, refer to the latest version of the *Cisco Unified Communications Manager with Cisco VCS Deployment Guide*, available at

http://www.cisco.com/en/US/docs/telepresence/infrastructure/vcs/config_guide/Cisco_VCS_Cisco_Unified_Communications_Manager_Deployment_Guide_CUCM_8_9_and_X7-2.pdf

Figure 6-29 Interconnecting Unified CM and VCS

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IP PSTN and IP Trunks to Service Provider Networks

Service providers are increasing their offerings of non-TDM PSTN connections to enterprise customers. Apart from the key benefit of the cost savings from deploying non-TDM interfaces, these IP-based PSTN connections might also offer additional voice features compared to traditional PSTN interfaces.

SIP-based services dominate the available offerings, and although earlier H.323 services were available in select geographies, they are being phased out. This is mainly due to the increasing popularity of SIP within the enterprise and the promise of additional capabilities such as Presence and support for many rich media applications (such as instant messaging). SIP will probably become the most widely deployed Unified Communications protocol in the long term.

When connecting to a service provider's IP PSTN network, Cisco strongly recommends the use of the Cisco Unified Border Element as an enterprise edge Session Border Controller to provide a controlled demarcation and security point between your enterprise and the service provider's network.

Cisco Unified Border Element

The Cisco Unified Border Element provides a wide range of signaling and media functionality between the enterprise and service-provider Cisco Unified Communications networks. Cisco Unified Border Element provides a Session Border Controller network-to-network interface point for:

- Address and port translations (privacy and Level 7 topology hiding)
- SIP Delayed Offer to Early Offer conversion
- Protocol interworking (H.323 and SIP) and normalization
- Media interworking (DTMF, fax, codec transcoding, and transrating volume and gain control)
- Call admission control (based on total calls, CPU, memory, call arrival spike detection, or maximum calls per destination)
- Security (including SIP malformed packet detection, non-dialog RTP packet drops, SIP listening port configuration, digest authentication, simultaneous call limits, call rate limits, toll fraud protection, and a number of signaling and media encryption options)
- PPI/PAI/Privacy and RPID Interworking with service providers
- QoS and bandwidth management (QoS marking using ToS, DSCP, and bandwidth enforcement using RSVP and codec filtering)
- Simultaneous connectivity to SIP trunks from multiple service providers

High availability with in-box or box-to-box failover options (depending on platform and release)

- Billing statistics and CDR collection

The Cisco Unified Border Element is a licensed Cisco IOS application available on the Cisco Integrated Service Routers Generation 2 (ISR G2), the Cisco AS5000XM Media Gateways, and the Cisco 1000 Series Aggregation Services Routers (ASR). Depending on your choice of hardware platform, the Cisco Unified Border Element can provide session scalability for up to 16,000 concurrent voice calls with in-box or box-to-box failover options.

For more information on the Cisco Unified Border Element, refer to the documentation at

<http://www.cisco.com/go/cube>

Trunk Aggregation Platforms

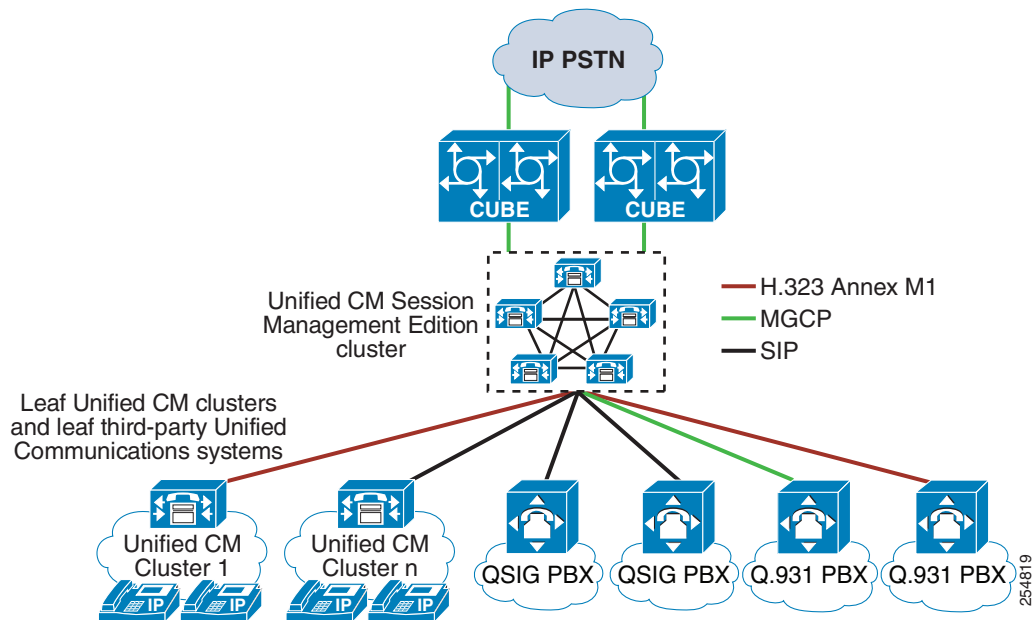
Large-scale IP PSTN deployments often involve the requirement to aggregate trunks from many Unified Communications systems prior to connecting them by means of the Cisco Unified Border Element to the service provider's IP PSTN. In most cases, the choice of aggregation platform depends on the protocol(s) the end systems are using. Cisco Unified CM Session Management Edition and Cisco Unified SIP Proxy are two commonly used aggregation platforms and are discussed in this section. Cisco's H.323 gatekeeper is also an option but tends to be less widely used today since SIP has become the preferred choice of protocol for IP PSTN connections. (Cisco H.323 gatekeeper options are discussed in the section on [H.323 Trunk Types, page 6-39](#).)

Unified CM Session Management Edition

Unified Communications deployments using Unified CM Session Management Edition are a variation on the multisite distributed call processing deployment model and are typically employed to interconnect large numbers of Unified Communications systems and also to provide connections to an IP PSTN.

Cisco Unified CM Session Management Edition is essentially a Unified CM cluster with trunk interfaces only and no IP endpoints. It enables aggregation of multiple unified communications systems, referred to as leaf systems. (See [Figure 6-30](#).)

Figure 6-30 Cisco Unified CM Session Management Edition



Unified CM Session Management Edition deployments can be used to migrate a deployment of multiple PBXs and associated phones to a Unified CM cluster with IP phones and relatively few trunks. The Unified CM Session Management Edition cluster may start with a large number of trunks interconnecting third-party PBXs and migrate over time to a Unified CM cluster deployment with thousands of IP phones. Unified CM Session Management Edition may also be used to connect to third-party unified communications systems such as IP PSTN connections and centralized unified communications applications.

Unified CM Session Management Edition supports the following features:

- SIP trunks
- H.323 trunks
- MGCP trunks
- Voice calls
- Video calls
- Encrypted calls
- Fax calls

Because Unified CM Session Management Edition uses the same software as Unified CM, all Unified CM features such as scalability, availability, load balancing, SIP message normalization, call routing, and number modification are available to the Unified CM Session Management Edition cluster.

For more information on Cisco Unified CM Session Management Edition, see the chapter on [Collaboration Deployment Models](#), page 10-1.

Cisco Unified SIP Proxy

The Cisco Unified SIP Proxy provides SIP proxy functionality on a Cisco NME-522 network module that can be plugged into a network module slot on the Cisco 3800 Series Integrated Services Routers (ISRs). This ISR does not have to be dedicated to hosting the network module and running the proxy alone, but can be used simultaneously for other network functions such as to run the Cisco Unified Border Element described above.

The Cisco Unified SIP Proxy brings the following benefits to a network using Unified CM SIP trunks:

- Aggregation and routing

The Unified SIP Proxy is capable of connecting several SIP servers to each other without each of the servers connecting to every other one in a full-mesh configuration

- Scalability

The Unified SIP Proxy can be used to terminate calls to and from the enterprise and IP PSTN service providers. The proxy, in turn, distributes the calls across a pool of Unified Border Elements. More Unified Border Elements may be added to increase capacity.

- Availability and load balancing

The Unified SIP Proxy distributes calls over the pool of available Unified Border Elements and monitors the status of each Unified Border Element to ensure reliable call completion.

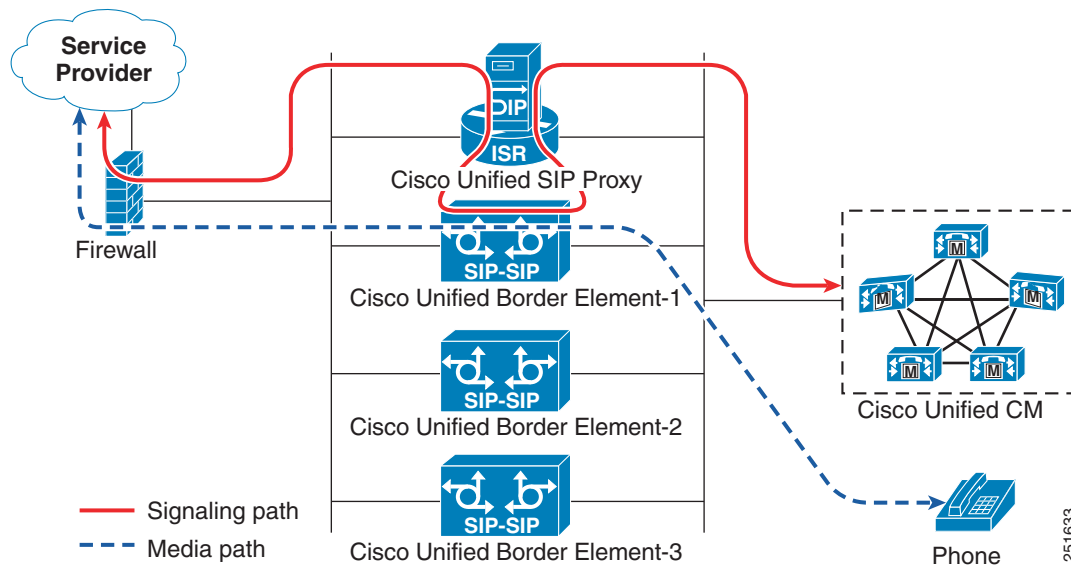
- Message normalization

The Unified SIP Proxy serves to hide differences in SIP protocol messaging by providing the means to manipulate headers and contents of the messages as they pass through the Unified SIP Proxy.

The following design considerations should be taken into account when deploying the Cisco Unified SIP Proxy with the Cisco Unified Border Element:

- Configure a load balancing scheme on the Unified SIP Proxy so that none of the attached Unified Border Elements is overloaded.
- Set up trunk monitoring in the Unified SIP Proxy so that it can detect and act on any failure of the Unified CM or Unified Border Element.

Figure 6-31 shows the call flow using the Cisco Unified SIP Proxy with Cisco Unified Border Element.

Figure 6-31 Call Flow for Cisco Unified SIP Proxy with Cisco Unified Border Element

To originate a call to the service provider, Unified CM sends the call to the Unified SIP Proxy. The Unified SIP Proxy determines that the request came from Unified CM and forwards the call to a Unified Border Element. The Unified Border Element terminates and re-originates the call, and sends it back to the Unified SIP Proxy. The Unified SIP Proxy determines that the call came from a Unified Border Element, and this time forwards it to the service provider. Media is then established directly from the originating phone to the service provider through the Unified Border Element.

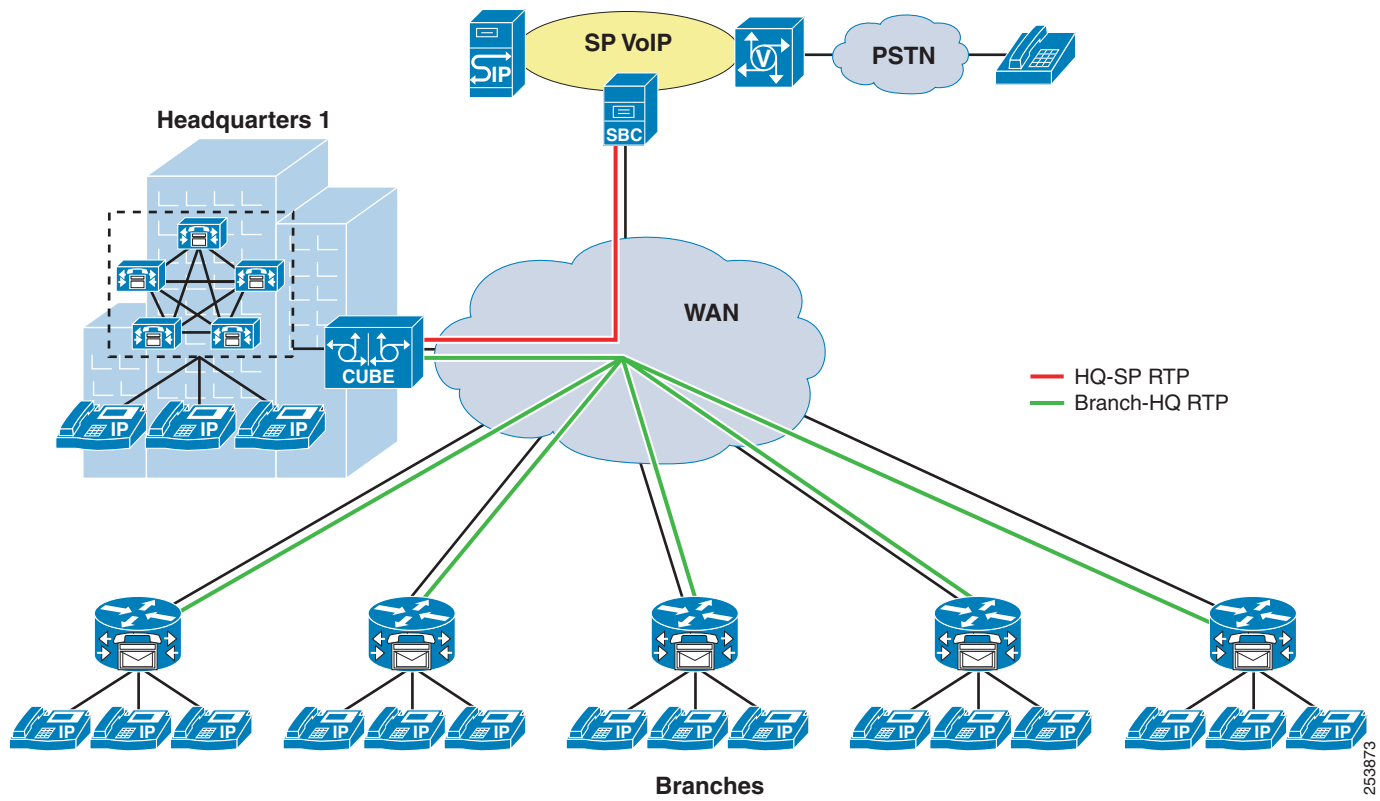
Large-Scale Session Border Controller

Depending on hardware platform, Cisco Unified Border Element on a single hardware chassis can aggregate up to 16,000 simultaneous calls on SIP trunks from one or more providers, and if desired, also with stateful failover between redundant chassis.

Trunk IP-PSTN Connection Models

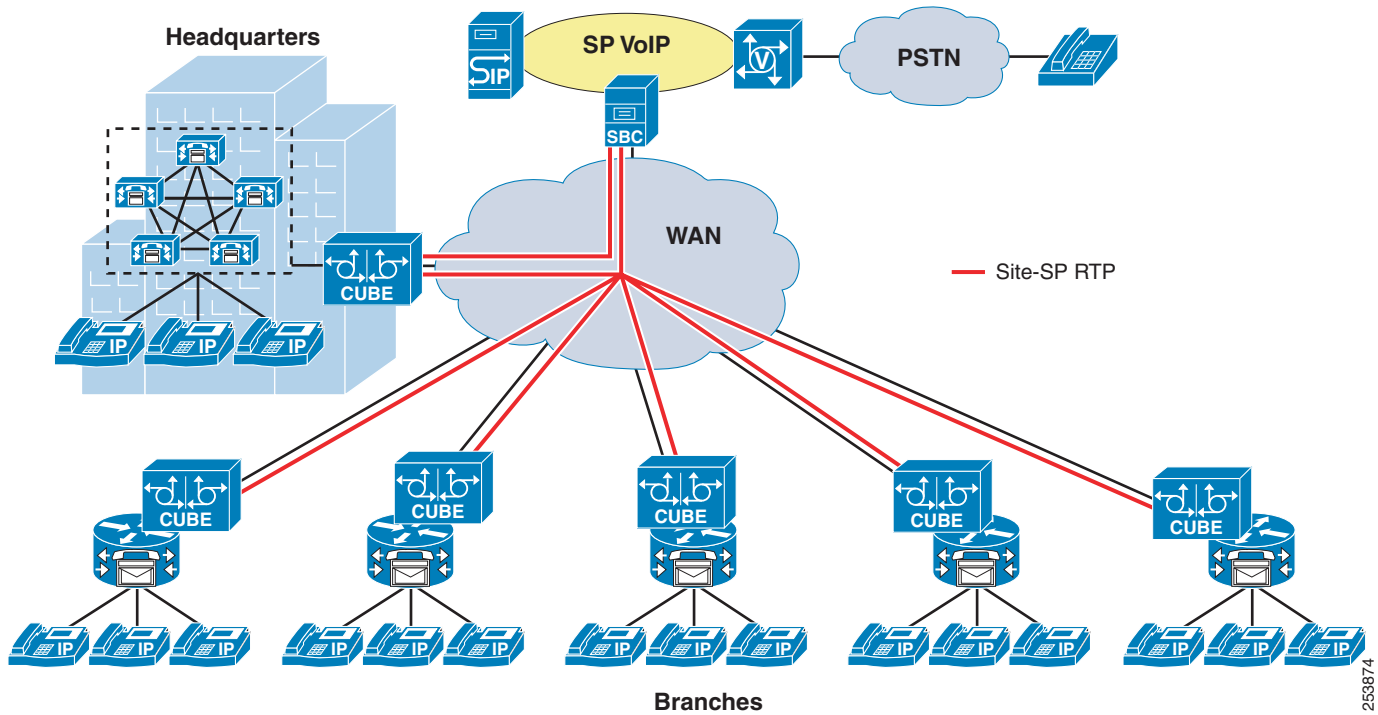
Trunks may be connected to IP PSTN service providers in several different ways, depending on the desired architecture. The two most common architectures for this connectivity are centralized trunks and distributed trunks.

Centralized trunks connect to the service provider (SP) through one logical connection (although there may be more than one physical connection for redundancy) with Session Border Controllers (SBCs) such as the Cisco Unified Border Element. (See [Figure 6-32](#).) All calls to and from the enterprise use this set of trunks. If the enterprise hosts a single central Unified CM cluster at its headquarters, with remote branches connected to the headquarters through a WAN, then media and signaling for PSTN calls to and from each of the sites traverse the WAN.

Figure 6-32 Centralized or Aggregated SIP Trunk Model

Distributed trunks connect to the service provider through several logical connections. (See [Figure 6-33](#).) Each branch of an enterprise may have its own local trunk to the service provider. Media from branches no longer needs to traverse the WAN but flows to the service provider interface through a local SBC.

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Figure 6-33 *Distributed SIP Trunk Model*

Each connectivity model has its own advantages and disadvantages. Centralized trunks are generally easier to deploy in terms of both physical equipment and configuration complexity. Distributed trunks have the advantage of local hand-off of media and better number portability from local providers. As illustrated in [Figure 6-34](#), a hybrid connectivity model that groups some of the branches together for connectivity, or that provides trunks from each Unified CM cluster of a multi-cluster deployment, captures the advantages of both forms of deployment.

Figure 6-34 Hybrid SIP Trunk Model with Regional Aggregation