



## **Enterprise Application Interoperability for H.323-to-SIP and SIP-to-SIP Configuration Guide, Cisco IOS XE Release 3S (Cisco ASR 1000)**

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## CHAPTER

# 1

# Enterprise Application Interoperability for H.323-to-SIP and SIP-to-SIP

This Cisco Unified Border Element (Enterprise) is a special Cisco IOS XE software image that runs on Cisco ASR1000. It provides a network-to-network interface point for billing, security, call admission control, quality of service, and signaling interworking. This chapter describes basic gateway functionality, software images, topology, and summarizes supported features.



### Note

Cisco Product Authorization Key (PAK)--A Product Authorization Key (PAK) is required to configure some of the features described in this guide. Before you start the configuration process, please register your products and activate your PAK at the following URL <http://www.cisco.com/go/license> .

- [Finding Feature Information, page 1](#)
- [Configuration of Enterprise Application Interoperability for H.323-to-SIP and SIP-to-SIP Features, page 1](#)

## Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see [Bug Search Tool](#) and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to [www.cisco.com/go/cfn](http://www.cisco.com/go/cfn). An account on Cisco.com is not required.

## Configuration of Enterprise Application Interoperability for H.323-to-SIP and SIP-to-SIP Features

This chapter contains the following configuration topics:

**Cisco UBE (Enterprise) Prerequisites and Restrictions**

- Prerequisites for Cisco Unified Border Element (Enterprise)
- Restrictions for Cisco Unified Border Element (Enterprise)

**CUCM Interworking**

- [Cisco Interoperability Portal](#)

[www.cisco.com/go/interoperability](http://www.cisco.com/go/interoperability)

**Third Party PBX Interworking**

- [Cisco Interoperability Portal](#)

[www.cisco.com/go/interoperability](http://www.cisco.com/go/interoperability)

**Application specific interworking notes**

- Support for SIP 181 "call is being forwarded" message
- Support for Expires timer reset on receiving or sending SIP 183 message



## Configuring SIP 181 Call is Being Forwarded Message

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You can configure support for SIP 181 Call is Being Forwarded messages either globally or on a specific dial-peer. Use the **block** command in voice service SIP configuration mode to globally configure Cisco IOS voice gateways and Cisco UBEs to drop specified SIP provisional response messages. To configure settings for an individual dial peer, use the **voice-class sip block** command in dial peer voice configuration mode. Both globally and at the dial peer level, you can also use the **sdp** keyword to further control when the specified SIP message is dropped based on either the absence or presence of SDP information.

Additionally, you can use commands introduced for this feature to configure a Cisco UBE, either globally or at the dial peer level, to map specific received SIP provisional response messages to a different SIP provisional response message on the outgoing SIP dial peer. To do so, use the **map resp-code** command in voice service SIP configuration mode for global configuration or, to configure a specific dial peer, use the **voice-class sip map resp-code** in dial peer voice configuration mode.

This section contains the following tasks:

- [Finding Feature Information, page 3](#)
- [Prerequisites for SIP 181 Call is Being Forwarded Message, page 4](#)
- [Configuring SIP 181 Call is Being Forwarded Message Globally, page 4](#)
- [Configuring SIP 181 Call is Being Forwarded Message at the Dial-Peer Level, page 5](#)
- [Configuring Mapping of SIP Provisional Response Messages Globally, page 6](#)
- [Configuring Mapping of SIP Provisional Response Messages at the Dial-Peer Level, page 8](#)
- [Feature Information for Configuring SIP 181 Call is Being Forwarded Message, page 9](#)

## Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see [Bug Search Tool](#) and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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## Prerequisites for SIP 181 Call is Being Forwarded Message

### Cisco Unified Border Element

Cisco IOS Release 15.0(1)XA or a later release must be installed and running on your Cisco Unified Border Element.

### Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.1S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

## Configuring SIP 181 Call is Being Forwarded Message Globally

Perform this task to configure support for SIP 181 messages at a global level in SIP configuration (conf-serv-sip) mode.

### SUMMARY STEPS

1. `enable`
2. `configure terminal`
3. `voice service voip`
4. `sip`
5. `block {180 | 181 | 183} [sdp {absent | present}]`
6. `exit`

### DETAILED STEPS

	Command or Action	Purpose
<b>Step 1</b>	<p><code>enable</code></p> <p><b>Example:</b></p> <pre>Router&gt; enable</pre>	<p>Enters privileged EXEC mode, or other security level set by a system administrator.</p> <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<p><code>configure terminal</code></p> <p><b>Example:</b></p> <pre>Router# configure terminal</pre>	<p>Enters global configuration mode.</p>

	Command or Action	Purpose
<b>Step 3</b>	<b>voice service voip</b>  <b>Example:</b> Router(config)# voice service voip	Enters voice service VoIP configuration mode.
<b>Step 4</b>	<b>sip</b>  <b>Example:</b> Router(conf-voi-serv)# sip	Enters SIP configuration mode.
<b>Step 5</b>	<b>block {180   181   183} [sdp {absent   present}]</b>  <b>Example:</b> Router(conf-serv-sip)# block 181 sdp present	Configures support of SIP 181 messages globally so that messages are passed as is. The sdp keyword is optional and allows for dropping or passing of SIP 181 messages based on the presence or absence of SDP.
<b>Step 6</b>	<b>exit</b>  <b>Example:</b> Router(conf-serv-sip)# exit	Exits the current mode.

## Configuring SIP 181 Call is Being Forwarded Message at the Dial-Peer Level

Perform this task to configure support for SIP 181 messages at the dial-peer level, in dial peer voice configuration (config-dial-peer) mode.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **voice-class sip block {180 | 181 | 183} [sdp {absent | present}]**
5. **exit**

## DETAILED STEPS

	Command or Action	Purpose
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Router> enable	Enters privileged EXEC mode, or other security level set by a system administrator. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
<b>Step 3</b>	<b>dial-peer voice tag voip</b>  <b>Example:</b> Router(config)# dial-peer voice 2 voip	Enters dial peer VoIP configuration mode.
<b>Step 4</b>	<b>voice-class sip block {180   181   183} [sdp {absent   present}]</b>  <b>Example:</b> Router(config-dial-peer)# voice-class sip block 181 sdp present	Configures support of SIP 181 messages on a specific dial peer so that messages are passed as is. The sdp keyword is optional and allows for dropping or passing of SIP 181 messages based on the presence or absence of SDP.
<b>Step 5</b>	<b>exit</b>  <b>Example:</b> Router(config-dial-peer)# exit	Exits the current mode.

## Configuring Mapping of SIP Provisional Response Messages Globally

Perform this task to configure mapping of specific received SIP provisional response messages at a global level in SIP configuration (conf-serv-sip) mode.

**SUMMARY STEPS**

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **map resp-code 181 to 183**
6. **exit**

**DETAILED STEPS**

	<b>Command or Action</b>	<b>Purpose</b>
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Router> enable	Enters privileged EXEC mode, or other security level set by a system administrator. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
<b>Step 3</b>	<b>voice service voip</b>  <b>Example:</b> Router(config)# voice service voip	Enters voice service VoIP configuration mode.
<b>Step 4</b>	<b>sip</b>  <b>Example:</b> Router(conf-voi-serv)# sip	Enters SIP configuration mode.
<b>Step 5</b>	<b>map resp-code 181 to 183</b>  <b>Example:</b> Router(conf-serv-sip)# map resp-code 181 to 183	Enables mapping globally of received SIP messages of a specified message type to a different SIP message type.
<b>Step 6</b>	<b>exit</b>  <b>Example:</b> Router(conf-serv-sip)# exit	Exits the current mode.

# Configuring Mapping of SIP Provisional Response Messages at the Dial-Peer Level

Perform this task to configure mapping of received SIP provisional response messages at the dial-peer level, in dial peer voice configuration (config-dial-peer) mode.

## SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **voice-class sip map resp-code 181 to 183**
5. **exit**

## DETAILED STEPS

	Command or Action	Purpose
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Router> enable	Enters privileged EXEC mode, or other security level set by a system administrator. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
<b>Step 3</b>	<b>dial-peer voice tag voip</b>  <b>Example:</b> Router(config)# dial-peer voice 2 voip	Enters dial peer VoIP configuration mode.
<b>Step 4</b>	<b>voice-class sip map resp-code 181 to 183</b>  <b>Example:</b> Router(config-dial-peer)# voice-class sip map resp-code 181 to 183	Enables mapping of received SIP messages of a specified SIP message type on a specific dial peer to a different SIP message type.
<b>Step 5</b>	<b>exit</b>  <b>Example:</b> Router(config-dial-peer)# exit	Exits the current mode.

## Feature Information for Configuring SIP 181 Call is Being Forwarded Message

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to [www.cisco.com/go/cfn](http://www.cisco.com/go/cfn). An account on Cisco.com is not required.

Feature History Table entry for the Cisco Unified Border Element.

**Table 1: Feature Information for SIP 181 Call is Being Forwarded Messages**

Feature Name	Releases	Feature Information
SIP 181 Call is Being Forwarded Message	12.2(13)T	This feature allows users to configure support for SIP 181 Call is Being Forwarded messages either globally or on a specific dial-peer.  This feature includes the following new or modified commands: <b>block</b> , <b>map resp-code</b> , <b>voice-class sip block</b> , <b>voice-class sip map resp-code</b> .

Feature History Table entry for the Cisco Unified Border Element (Enterprise).

**Table 2: Feature Information for SIP 181 Call is Being Forwarded Messages**

Feature Name	Releases	Feature Information
SIP 181 Call is Being Forwarded Message	Cisco IOS XE Release 3.1S	This feature allows users to configure support for SIP 181 Call is Being Forwarded messages either globally or on a specific dial-peer.  This feature includes the following new or modified commands: <b>block</b> , <b>map resp-code</b> , <b>voice-class sip block</b> , <b>voice-class sip map resp-code</b> .





## CHAPTER 3

# Expires Timer Reset on Receiving or Sending SIP 183 Message

This feature enables support for resetting the Expires timer when receiving or sending SIP 183 messages on Cisco Unified Communications Manager Express (Cisco Unified CME), a Cisco IOS voice gateway, or a Cisco Unified Border Element (Cisco UBE). When the terminating device lacks answer supervision or does not send the required SIP 200 OK message within the timer expiry, you can enable this feature to send periodic SIP 183 messages to reset the Expires timer and preserve the call until final response. This feature can be enabled globally or on a specific dial peer. Additionally, you can configure this feature based on the presence or absence of Session Description Protocol (SDP).

For details about enabling this feature, see the **reset timer expires** and **voice-class sip reset timer expires** commands in the Cisco IOS Voice Command Reference.

- [Finding Feature Information, page 11](#)
- [Prerequisites for Expires Timer Reset on Receiving or Sending SIP 183 Message, page 12](#)
- [How to Configure Expires Timer Reset on Receiving or Sending SIP 183 Message, page 12](#)
- [Feature Information for Configuring Support for Expires Timer Reset on Receiving or Sending SIP 183 Message, page 14](#)

## Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see [Bug Search Tool](#) and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to [www.cisco.com/go/cfn](http://www.cisco.com/go/cfn). An account on Cisco.com is not required.

# Prerequisites for Expires Timer Reset on Receiving or Sending SIP 183 Message

Before configuring support for Expires timer reset for SIP 183 on Cisco IOS SIP time-division multiplexing (TDM) gateways, Cisco UBEs, or Cisco Unified CME, verify the SIP configuration within the VoIP network for the appropriate originating and terminating gateways as described in the Cisco IOS SIP Configuration Guide.

## Cisco Unified Border Element

- Cisco IOS Release 15.0(1)XA or a later release must be installed and running on your Cisco Unified Border Element.

## Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.1S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

# How to Configure Expires Timer Reset on Receiving or Sending SIP 183 Message

To configure the Support for Expires Timer Reset on Receiving or Sending SIP 183 Message feature, complete the tasks in this section. You can enable this feature globally, using the **reset timer expires** command in voice service SIP configuration mode, or on a specific dial-peer using the **voice-class sip reset timer expires** command in dial peer voice configuration mode.

## Configuring Reset of Expires Timer Globally

Perform this task to enable resetting of the Expires timer at the global level in SIP configuration (conf-serv-sip) mode.

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **reset timer expires 183**
6. **exit**

## DETAILED STEPS

	Command or Action	Purpose
Step 1	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
Step 3	<b>voice service voip</b>  <b>Example:</b> Router(config)# voice service voip	Enters voice service VoIP configuration mode.
Step 4	<b>sip</b>  <b>Example:</b> Router(conf-voi-serv)# sip	Enters SIP configuration mode.
Step 5	<b>reset timer expires 183</b>  <b>Example:</b> Router(conf-serv-sip)# reset timer expires 183	Enables resetting of the Expires timer upon receipt of SIP 183 messages globally.
Step 6	<b>exit</b>  <b>Example:</b> Router(conf-serv-sip)# exit	Exits the current mode.

## Configuring Reset of Expires Timer at the Dial-Peer Level

Perform this task to enable resetting of the Expires timer at the dial-peer level in dial peer voice configuration (config-dial-peer) mode.

**SUMMARY STEPS**

1. **enable**
2. **configure terminal**
3. **dial-peer voice tag voip**
4. **voice-class sip reset timer expires 183**
5. **exit**

**DETAILED STEPS**

	<b>Command or Action</b>	<b>Purpose</b>
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Router> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Router# configure terminal	Enters global configuration mode.
<b>Step 3</b>	<b>dial-peer voice tag voip</b>  <b>Example:</b> Router(config)# dial-peer voice 2 voip	Enters dial peer VoIP configuration mode.
<b>Step 4</b>	<b>voice-class sip reset timer expires 183</b>  <b>Example:</b> Router(config-dial-peer)# voice-class sip reset timer expires 183	Enables resetting of the Expires timer upon receipt of SIP 183 messages on a specific dial peer.
<b>Step 5</b>	<b>exit</b>  <b>Example:</b> Router(config-dial-peer)# exit	Exits the current mode.

## Feature Information for Configuring Support for Expires Timer Reset on Receiving or Sending SIP 183 Message

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to [www.cisco.com/go/cfn](http://www.cisco.com/go/cfn). An account on Cisco.com is not required. Feature History Table entry for the Cisco Unified Border Element.

**Table 3: Feature Information for Support for Expires Timer Reset on Receiving or Sending SIP 183 Message**

Feature Name	Releases	Feature Information
Support for Expires Timer Reset on Receiving or Sending SIP 183 Message	15.0(1)XA 15.1(1)T	This feature enables support for resetting the Expires timer upon receipt of SIP 183 messages on Cisco Unified Communications Manager Express (Cisco Unified CME), a Cisco IOS voice gateway, or a Cisco Unified Border Element (Cisco UBE).  The following commands were introduced or modified: <b>reset timer expires</b> and <b>voice-class sip reset timer expires</b> .

Feature History Table entry for the Cisco Unified Border Element (Enterprise) .

**Table 4: Feature Information for Support for Expires Timer Reset on Receiving or Sending SIP 183 Message**

Feature Name	Releases	Feature Information
Support for Expires Timer Reset on Receiving or Sending SIP 183 Message	Cisco IOS XE Release 3.1S	This feature enables support for resetting the Expires timer upon receipt of SIP 183 messages on Cisco Unified Communications Manager Express (Cisco Unified CME), a Cisco IOS voice gateway, or a Cisco Unified Border Element (Cisco UBE).  The following commands were introduced or modified: <b>reset timer expires</b> and <b>voice-class sip reset timer expires</b> .





# Cisco Unified Border Element Intercluster Lookup Service

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The Cisco Unified Border Element (Cisco UBE) Intercluster Lookup Service feature enables Cisco Unified Communications Manager to establish calls using Uniform Resource Identifiers (URIs.) It provides a framework for sharing information about user-contact information between Cisco Unified Communications Manager clusters. All URIs being used within a cluster are grouped together and associated with a cluster identifier called a route string. To interoperate with Cisco Unified Communications Manager, Cisco UBE is enhanced to route the call based on the received destination route string. This feature works with Cisco Unified Communication Manager Version 9.5 and later.

- [Finding Feature Information, page 17](#)
- [Information About Cisco UBE Intercluster Lookup Service, page 18](#)
- [How to Configure Cisco UBE Intercluster Lookup Service, page 19](#)
- [Configuration Examples for Cisco UBE Intercluster Lookup Service, page 27](#)
- [Feature Information for Cisco UBE Intercluster Lookup Service, page 28](#)

## Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see [Bug Search Tool](#) and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table at the end of this module.

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# Information About Cisco UBE Intercluster Lookup Service

## Cisco UBE Intercluster Lookup Service Overview

A Uniform Resource Identifier (URI) is a device-independent user address. A subscriber can use a URI as a personal identity and move from one network to another without any change in the URI. You cannot summarize URIs within an enterprise network (for example, abc@company.com) the same way that directory number ranges are summarized.

The Intercluster Lookup Services is a dynamic mechanism to discover URIs. When it is enabled, Cisco Unified Communications Manager users can initiate calls using URIs. The Intercluster Lookup Service provides a framework for sharing user-contact information between Cisco Unified Communications Manager clusters. All URIs being used within a cluster are grouped together and associated with a cluster identifier called a route string. These URI groups and their associated route strings are shared between all other participating clusters.

While initiating a call, the URI uses the Intercluster Lookup Service to identify the target URI and associated route string to route the call between clusters. Cisco Unified Communications Manager uses a Session Initiation Protocol (SIP) route pattern to match the route string returned by Intercluster Lookup Service and route the call over a SIP trunk. If Intercluster Lookup Service is enabled, the Cisco Unified Communications Manager SIP trunk sends the SIP invite message with destination route string header information.

To interoperate with Cisco Unified Communications Manager, Cisco UBE is enhanced to route the call based on the received destination route string. Cisco UBE supports exact match and wildcard match for a route string and parses the received destination route string header and routes a call forward to the destination. The destination can be a Cisco Unified Communications Manager cluster, public switched telephone network (PSTN), or any third-party unified communications device.

The dial-peer module is enhanced to support the dial-peer matching based on the destination route string header. The destination route string is used to match an outbound dial peer. The match can be an exact match or wildcard match.

For example, consider London.UK.EU as the route string. The SIP dial-peer configuration is as follows:

- Dial-peer 1: London.UK.EU
- Dial-peer 2: \*.UK.EU
- Dial-peer 3: \*.EU

The destination route string header and route string match are not case-sensitive. In this scenario, London.UK.EU and london.uk.eu match dial-peer 1 and therefore, dial-peer 1 is selected for outbound process.

If call routing policies are enabled, call routing based on a destination route string takes precedence over any other routing configurations. For example, if call routing is configured on a destination route string globally or at the dial-peer level, the call is routed considering the destination route string. If no match is found, then the call is routed using other URLs and header configuration options.

## Cisco UBE Enterprise Support for URIs

Cisco UBE Enterprise does not support non-E164 URI number user-part in request line and header. For URI dialing from Cisco Unified Communications Manager phone, the URI in user@dest-route-string format is

used. By default, the integrated services router (ISR) converts this format to the session target IP address of the outbound dial-peer and delivers non-E164 numbers.

As a workaround, you can use a SIP profile to pass through the required URI format. You can configure the SIP profile on an outbound dial-peer to modify the URI to the desired format.

# How to Configure Cisco UBE Intercluster Lookup Service

## Configuring a Route String Pattern

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class route-string *tag***
4. **pattern *string***
5. **end**

### DETAILED STEPS

	Command or Action	Purpose
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Device> enable	Enables privileged EXEC mode.  • Enter your password if prompted.
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Device# configure terminal	Enters global configuration mode.
<b>Step 3</b>	<b>voice class route-string <i>tag</i></b>  <b>Example:</b> Device(config)# voice class route-string 2	Enters voice class configuration mode.
<b>Step 4</b>	<b>pattern <i>string</i></b>  <b>Example:</b> Device(config-class)# pattern london.uk.eu	Configures a pattern string in the specified route string.  <b>Note</b> Multiple patterns can be configured under one route string class and the same route string class can be configured under multiple dial-peers. You also can use an asterisk (*) as the wildcard match option while provisioning the pattern.

	Command or Action	Purpose
<b>Step 5</b>	<b>end</b>  <b>Example:</b> Device (config-class) # end	Exits voice class configuration mode and returns to privileged EXEC mode.

## Configuring a Call Route on a Destination Route String Globally

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **call-route dest-route-string**
6. **end**

### DETAILED STEPS

	Command or Action	Purpose
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Device# configure terminal	Enters global configuration mode.
<b>Step 3</b>	<b>voice service voip</b>  <b>Example:</b> Device(config)# voice service voip	Enters voice service configuration mode.
<b>Step 4</b>	<b>sip</b>  <b>Example:</b> Device (conf-voi-serv) # sip	Enters SIP configuration mode.

	Command or Action	Purpose
<b>Step 5</b>	<b>call-route dest-route-string</b>  <b>Example:</b> <pre>Device(conf-serv-sip)# call-route dest-route-string</pre>	Configures call routing globally on a destination route string. <b>Note</b> By default, call routing on a destination route string is disabled.
<b>Step 6</b>	<b>end</b>  <b>Example:</b> <pre>Device(conf-serv-sip)# end</pre>	Exits SIP configuration mode and returns to privileged EXEC mode.

## Configuring a Route String Passthrough List Header

### SUMMARY STEPS

1. enable
2. configure terminal
3. voice class sip-hdr-passthru-list *tag*
4. passthru-hdr *name*
5. end

### DETAILED STEPS

	Command or Action	Purpose
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> <pre>Device&gt; enable</pre>	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> <pre>Device# configure terminal</pre>	Enters global configuration mode.
<b>Step 3</b>	<b>voice class sip-hdr-passthru-list <i>tag</i></b>  <b>Example:</b> <pre>Device(config)# voice class sip-hdr-passthru-list 2</pre>	Enters voice class configuration mode.

	Command or Action	Purpose
<b>Step 4</b>	<b>passthru-hdr</b> <i>name</i>  <b>Example:</b> Device (config-class) # passthru-hdr x-cisco-dest-route-string	Configures header to be added to the route string passthrough list.
<b>Step 5</b>	<b>end</b>  <b>Example:</b> Device (config-class) # end	Exits voice class configuration mode and returns to privileged EXEC mode.

## Configuring a Destination Route String Call Route at the Dial-Peer Level

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **description** *string*
5. **destination route-string** *tag*
6. **session protocol sipv2**
7. **session target ipv4:***destination address*
8. **voice-class sip call-route dest-route-string**
9. **end**

### DETAILED STEPS

	Command or Action	Purpose
<b>Step 1</b>	<b>enable</b>  <b>Example:</b> Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
<b>Step 2</b>	<b>configure terminal</b>  <b>Example:</b> Device# configure terminal	Enters global configuration mode.

	Command or Action	Purpose
<b>Step 3</b>	<b>dial-peer voice tag voip</b>  <b>Example:</b> Device(config)# dial-peer voice 1 voip	Enters dial peer voice configuration mode.
<b>Step 4</b>	<b>description string</b>  <b>Example:</b> Device(config-dial-peer)# description outbound-dialpeer	Adds descriptive information about the dial peer.
<b>Step 5</b>	<b>destination route-string tag</b>  <b>Example:</b> Device(config-dial-peer)# destination route-string 2	Configures a destination route string for the dial peer. <b>Note</b> By default, the call route on a destination route string is disabled. The destination route string call route configuration at the dial-peer level takes precedence over the global configuration when routing a call.
<b>Step 6</b>	<b>session protocol sipv2</b>  <b>Example:</b> Device(config-dial-peer)# session protocol sipv2	Configures the IETF Session Initiation Protocol (SIP) for the dial peer.
<b>Step 7</b>	<b>session target ipv4:destination address</b>  <b>Example:</b> Device(config-dial-peer)# session target ipv4:192.0.2.6	Configures the session target IP address of the dial peer.
<b>Step 8</b>	<b>voice-class sip call-route dest-route-string</b>  <b>Example:</b> Device(config-dial-peer)# voice-class sip call-route dest-route-string	Configures call routing on the destination route string for a dial peer.
<b>Step 9</b>	<b>end</b>  <b>Example:</b> Device(config-dial-peer)# end	Exits dial peer voice configuration mode and returns to privileged EXEC mode.

## Configuring a Route String Header Passthrough at the Dial-Peer Level

### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice *tag* voip**
4. **description *string***
5. **session protocol sipv2**
6. **voice-class sip pass-thru headers *tag***
7. **end**

### DETAILED STEPS

	Command or Action	Purpose
Step 1	<b>enable</b>  <b>Example:</b> Device> enable	Enables privileged EXEC mode. <ul style="list-style-type: none"> <li>• Enter your password if prompted.</li> </ul>
Step 2	<b>configure terminal</b>  <b>Example:</b> Device# configure terminal	Enters global configuration mode.
Step 3	<b>dial-peer voice <i>tag</i> voip</b>  <b>Example:</b> Device(config)# dial-peer voice 1 voip	Enters dial peer voice configuration mode.
Step 4	<b>description <i>string</i></b>  <b>Example:</b> Device(config-dial-peer)# description inbound-dialpeer	Adds descriptive information about the dial peer.
Step 5	<b>session protocol sipv2</b>  <b>Example:</b> Device(config-dial-peer)# session protocol sipv2	Configures the IETF Session Initiation Protocol (SIP) for the dial peer.
Step 6	<b>voice-class sip pass-thru headers <i>tag</i></b>  <b>Example:</b> Device(config-dial-peer)# voice-class sip pass-thru headers hdr1	Enables call routing based on the destination route string for a dial peer.

	Command or Action	Purpose
<b>Step 7</b>	<b>end</b>  <b>Example:</b> Device(config-dial-peer)# end	Exits dial peer voice configuration mode and returns to privileged EXEC mode.

## Verifying Cisco UBE Intercluster Lookup Service Configuration

The **show** commands can be entered in any order.

### SUMMARY STEPS

1. **enable**
2. **show voice class route-string**
3. **show call active voice**
4. **show call history voice**
5. **show sip call**

### DETAILED STEPS

#### Step 1

**enable**

Enables privileged EXEC mode.

- Enter your password if prompted.

**Example:**

```
Device> enable
```

#### Step 2

**show voice class route-string**

Displays the call route-string status for voice ports.

**Example:**

```
Device# show voice class route-string
voice class route-string 2:
  pattern london.uk.eu
  configured in dial-peers: 7 4 6
```

#### Step 3

**show call active voice**

Displays call information for voice calls in progress. The sample output below shows the destination route string configuration.

**Example:**

```
Device# show call active voice
DestinationRouteStr=london.uk.eu
```

**Step 4****show call history voice**

Displays the call history table for voice calls. The sample output below shows the destination route string configuration.

**Example:**

```
Device# show call history voice | in Des
DestinationRouteStr=london.uk.eu
```

**Step 5****show sip call**

Displays active user agent client (UAC) and user agent server (UAS) information on SIP calls.

**Example:**

```
Device# show sip call
Total SIP call legs:2, User Agent Client:1, User Agent Server:1
SIP UAC CALL INFO
Call 1
SIP Call ID          : 5A4CAE55-E48D11E2-802BDD60-8693A1D1@192.0.2.1
State of the call    : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number       : 345111
Called Number        :
Bit Flags            : 0xC04018 0x10000100 0x80
CC Call ID           : 12
Source IP Address (Sig) : 192.0.2.1
Destn SIP Req Addr:Port : [192.0.2.6]:5060
Destn SIP Resp Addr:Port : [192.0.2.6]:5060
Destination Name     : 192.0.2.6
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object      : 0x0
Media Mode           : flow-through
Media Stream 1
State of the stream  : STREAM_ACTIVE
Stream Call ID       : 12
Stream Type          : voice-only (0)
Stream Media Addr Type : 1
Negotiated Codec     : g711ulaw (160 bytes)
Codec Payload Type   : 0
Negotiated Dtmf-relay : inband-voice
Dtmf-relay Payload Type : 0
QoS ID               : -1
Local QoS Strength   : BestEffort
Negotiated QoS Strength : BestEffort
Negotiated QoS Direction : None
Local QoS Status     : None
Media Source IP Addr:Port : [192.0.2.1]:16406
Media Dest IP Addr:Port  : [192.0.2.6]:6020

Options-Ping      ENABLED:NO      ACTIVE:NO
Number of SIP User Agent Client(UAC) calls: 1

SIP UAS CALL INFO
Call 1
SIP Call ID          : 1-27273@192.0.2.6
State of the call    : STATE_ACTIVE (7)
Substate of the call : SUBSTATE_NONE (0)
Calling Number       : 345111
Called Number        : alice
```

```

Bit Flags          : 0xC0401C 0x10000100 0x4
CC Call ID        : 11
Source IP Address (Sig) : 192.0.2.1
Destn SIP Req Addr:Port : [192.0.2.6]:5061
Destn SIP Resp Addr:Port: [192.0.2.6]:5061
Destination Name   : 192.0.2.6
Destination Route String: london.uk.eu //This is the configured dest-route-string pattern.//
Number of Media Streams : 1
Number of Active Streams: 1
RTP Fork Object    : 0x0
Media Mode         : flow-through
Media Stream 1
  State of the stream : STREAM_ACTIVE
  Stream Call ID      : 11
  Stream Type         : voice-only (0)
  Stream Media Addr Type : 1
  Negotiated Codec    : g711ulaw (160 bytes)
  Codec Payload Type  : 0
  Negotiated Dtmf-relay : inband-voice
  Dtmf-relay Payload Type : 0
  QoS ID              : -1
  Local QoS Strength  : BestEffort
  Negotiated QoS Strength : BestEffort
  Negotiated QoS Direction : None
  Local QoS Status    : None
  Media Source IP Addr:Port: [192.0.2.1]:16404
  Media Dest IP Addr:Port : [192.0.2.6]:6000

```

```

Options-Ping      ENABLED:NO      ACTIVE:NO
Number of SIP User Agent Server(UAS) calls: 1

```

## Configuration Examples for Cisco UBE Intercluster Lookup Service

### Example: Configuring a Route String Pattern

```

Device> enable
Device# configure terminal
Device(config)# voice class route-string 2
Device(config-class)# pattern london.uk.eu
Device(config-class)# pattern *.uk.eu
Device(config-class)# pattern *.eu
Device(config-class)# end

```

### Example: Configuring a Call Route on a Destination Route String Globally

```

Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# sip
Device(conf-serv-sip)# call-route dest-route-string
Device(conf-serv-sip)# end

```

## Example: Configuring a Route String Passthrough List Header

```
Device> enable
Device# configure terminal
Device(config)# voice class sip-hdr-passthru-list 2
Device(config-class)# passthru-hdr x-cisco-dest-route-string
```

## Example: Configuring a Destination Route String Call Route at the Dial-Peer Level

```
Device> enable
Device# configure terminal
Device# dial-peer voice 1 voip
Device(config-dial-peer)# description outbound-dialpeer
Device(config-dial-peer)# destination route-string 2
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# session target ipv4:192.0.2.6
Device(config-dial-peer)# voice-class sip call-route dest-route-string
```

## Example: Configuring a Route String Header Passthrough at the Dial-Peer Level

```
Device> enable
Device# configure terminal
Device(config)# dial-peer voice 1 voip
Device(config-dial-peer)# description inbound-dialpeer
Device(config-dial-peer)# session protocol sipv2
Device(config-dial-peer)# voice-class sip pass-thru headers hdr1
Device(config-dial-peer)# end
```

## Feature Information for Cisco UBE Intercluster Lookup Service

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to [www.cisco.com/go/cfn](http://www.cisco.com/go/cfn). An account on Cisco.com is not required.

**Table 5: Feature Information for CUBE Inter Cluster Look Up Service**

Feature Name	Releases	Feature Information
CUBE Intercluster Lookup Service (ILS)	15.3(3)M	<p>The Cisco UBE Inter Cluster Look up Service feature enables Cisco Unified Communications Manager to establish calls using Uniform Resource Identifiers (URIs.) It provides a framework for sharing information about user-contact information between Cisco Unified Communications Manager clusters. All URIs being used within a cluster are grouped together and associated with a cluster identifier called a route string. To interoperate with Cisco Unified Communications Manager, Cisco UBE is enhanced to route the call based on the received destination route string. This feature works with Cisco Unified Communication Manager Version 9.5 and later.</p> <p>The following commands were introduced or modified:</p> <p><b>call-route,destination route-string,pass thru-hdr,voice class route-string,voice class sip-hdr-pass thru list,voice-class sip call-route,show call active voice,show call history voice.</b></p>

Feature Name	Releases	Feature Information
CUBE Intercluster Lookup Service (ILS)	Cisco IOS XE Release 3.10S	<p>The Cisco UBE Inter Cluster Lookup Service feature enables Cisco Unified Communications Manager to establish calls using Uniform Resource Identifiers (URIs.) It provides a framework for sharing information about user-contact information between Cisco Unified Communications Manager clusters. All URIs being used within a cluster are grouped together and associated with a cluster identifier called a route string. To interoperate with Cisco Unified Communications Manager, Cisco UBE is enhanced to route the call based on the received destination route string. This feature works with Cisco Unified Communication Manager Version 9.5 and later.</p> <p>The following commands were introduced or modified:</p> <p><b>call-route,destination route-string, passthru-hdr, voice class route-string, voice class sip-hdr-passthru-list, voice-class sip call-route, show call active voice, show call history voice.</b></p>



## Additional References

The following sections provide references related to the Cisco Unified Border Element (Enterprise) Configuration Guide.

- [Related Documents](#), page 31
- [Standards](#), page 32
- [MIBs](#), page 33
- [RFCs](#), page 33
- [Technical Assistance](#), page 35

## Related Documents

Related Topic	Document Title
Cisco IOS commands	<a href="#">Cisco IOS Master Commands List, All Releases</a>
Cisco IOS Voice commands	<i>Cisco IOS Voice Command Reference</i>
Cisco IOS Voice Configuration Library	For more information about Cisco IOS voice features, including feature documents, and troubleshooting information--at <a href="http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm">http://www.cisco.com/en/US/docs/ios/12_3/vvf_c/cisco_ios_voice_configuration_library_glossary/vcl.htm</a>
Cisco IOS Release 15.0	Cisco IOS Release 15.0 Configuration Guides
Cisco IOS Release 12.2	Cisco IOS Voice, Video, and Fax Configuration Guide, Release 12.2

Related Topic	Document Title
internet Low Bitrate Codec (iLBC) Documents	<ul style="list-style-type: none"> <li>Codecs section of the Dial Peer Configuration on Voice Gateway Routers Guide</li> </ul> <a href="http://www.cisco.com/en/US/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-dp-overview.html">http://www.cisco.com/en/US/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-dp-overview.html</a> <ul style="list-style-type: none"> <li>Dial Peer Features and Configuration section of the Dial Peer Configuration on Voice Gateway Routers Guide</li> </ul> <a href="http://www.cisco.com/en/US/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-dp-feat-cfg.html">http://www.cisco.com/en/US/docs/ios-xml/ios/voice/dialpeer/configuration/15-mt/vd-dp-feat-cfg.html</a>
Related Application Guides	<ul style="list-style-type: none"> <li><i>Cisco Unified Communications Manager and Cisco IOS Interoperability Guide</i></li> <li><i>Cisco IOS SIP Configuration Guide</i></li> <li><a href="#">Cisco Unified Communications Manager (CallManager) Programming Guides</a></li> </ul>
Troubleshooting and Debugging guides	<ul style="list-style-type: none"> <li>Cisco IOS Debug Command Reference, Release 12.4.</li> <li><i>Troubleshooting and Debugging VoIP Call Basics</i> at <a href="http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml">http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a0080094045.shtml</a></li> <li><i>VoIP Debug Commands</i> at</li> </ul> <a href="http://www.cisco.com/en/US/docs/routers/access/1700/1750/software/configuration/guide/debug.html">http://www.cisco.com/en/US/docs/routers/access/1700/1750/software/configuration/guide/debug.html</a>

## Standards

Standard	Title
ITU-T G.711	--

## MIBs

MIB	MIBs Link
<ul style="list-style-type: none"> <li>• CISCO-PROCESS MIB</li> <li>• CISCO-MEMORY-POOL-MIB</li> <li>• CISCO-SIP-UA-MIB</li> <li>• DIAL-CONTROL-MIB</li> <li>• CISCO-VOICE-DIAL-CONTROL-MIB</li> <li>• CISCO-DSP-MGMT-MIB</li> <li>• IF-MIB</li> <li>• IP-TAP-MIB</li> <li>• TAP2-MIB</li> <li>• USER-CONNECTION-TAP-MIB</li> </ul>	<p>To locate and download MIBs for selected platforms, Cisco IOS XE software releases, and feature sets, use Cisco MIB Locator found at the following URL:</p> <p><a href="http://www.cisco.com/go/mibs">http://www.cisco.com/go/mibs</a></p>

## RFCs

RFC	Title
RFC 1889	<i>RTP: A Transport Protocol for Real-Time Applications</i>
RFC 2131	<i>Dynamic Host Configuration Protocol</i>
RFC 2132	<i>DHCP Options and BOOTP Vendor Extensions</i>
RFC 2198	<i>RTP Payload for Redundant Audio Data</i>
RFC 2327	<i>SDP: Session Description Protocol</i>
RFC 2543	<i>SIP: Session Initiation Protocol</i>
RFC 2543-bis-04	<i>SIP: Session Initiation Protocol, draft-ietf-sip-rfc2543bis-04.txt</i>
RFC 2782	<i>A DNS RR for Specifying the Location of Services (DNS SRV)</i>
RFC 2833	<i>RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals</i>

<b>RFC</b>	<b>Title</b>
RFC 3203	<i>DHCP reconfigure extension</i>
RFC 3261	<i>SIP: Session Initiation Protocol</i>
RFC 3262	<i>Reliability of Provisional Responses in Session Initiation Protocol (SIP)</i>
RFC 3323	<i>A Privacy Mechanism for the Session Initiation Protocol (SIP)</i>
RFC 3325	<i>Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks</i>
RFC 3515	<i>The Session Initiation Protocol (SIP) Refer Method</i>
RFC 3361	<i>Dynamic Host Configuration Protocol (DHCP-for-IPv4) Option for Session Initiation Protocol (SIP) Servers</i>
RFC 3455	<i>Private Header (P-Header) Extensions to the Session Initiation Protocol (SIP) for the 3rd-Generation Partnership Project (3GPP)</i>
RFC 3608	<i>Session Initiation Protocol (SIP) Extension Header Field for Service Route Discovery During Registration</i>
RFC 3711	<i>The Secure Real-time Transport Protocol (SRTP)</i>
RFC 3925	<i>Vendor-Identifying Vendor Options for Dynamic Host Configuration Protocol version 4 (DHCPv4)</i>

## Technical Assistance

Description	Link
<p>The Cisco Support website provides extensive online resources, including documentation and tools for troubleshooting and resolving technical issues with Cisco products and technologies.</p> <p>To receive security and technical information about your products, you can subscribe to various services, such as the Product Alert Tool (accessed from Field Notices), the Cisco Technical Services Newsletter, and Really Simple Syndication (RSS) Feeds.</p> <p>Access to most tools on the Cisco Support website requires a Cisco.com user ID and password.</p>	<p><a href="http://www.cisco.com/cisco/web/support/index.html">http://www.cisco.com/cisco/web/support/index.html</a></p>





## Glossary

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- [Glossary, page 37](#)

## Glossary

**AMR-NB** —Adaptive Multi Rate codec - Narrow Band.

**Allow header** —Lists the set of methods supported by the UA generating the message.

**bind** — In SIP, configuring the source address for signaling and media packets to the IP address of a specific interface.

**call** —In SIP, a call consists of all participants in a conference invited by a common source. A SIP call is identified by a globally unique call identifier. A point-to-point IP telephony conversation maps into a single SIP call.

**call leg** —A logical connection between the router and another endpoint.

**CLI** —command-line interface.

**Content-Type header** —Specifies the media type of the message body.

**CSeq header** —Serves as a way to identify and order transactions. It consists of a sequence number and a method. It uniquely identifies transactions and differentiates between new requests and request retransmissions.

**delta** —An incremental value. In this case, the delta is the difference between the current time and the time when the response occurred.

**dial peer** —An addressable call endpoint.

**DNS** —Domain Name System. Used to translate H.323 IDs, URLs, or e-mail IDs to IP addresses. DNS is also used to assist in locating remote gatekeepers and to reverse-map raw IP addresses to host names of administrative domains.

**DNS SRV** —Domain Name System Server. Used to locate servers for a given service.

**DSP** —Digital Signal Processor.

**DTMF** —dual-tone multifrequency. Use of two simultaneous voice-band tones for dialing (such as touch-tone).

**EFXS** —IP phone virtual voice ports.

**FQDN** —fully qualified domain name. Complete domain name including the host portion; for example, *serverA.companyA.com* .

**FXS** —analog telephone voice ports.

**gateway** —A gateway allows SIP or H.323 terminals to communicate with terminals configured to other protocols by converting protocols. A gateway is the point where a circuit-switched call is encoded and repackaged into IP packets.

**H.323** —An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing. H.323 is an umbrella standard that describes the architecture of the conferencing system and refers to a set of other standards (H.245, H.225.0, and Q.931) to describe its actual protocol.

**iLBC** —internet Low Bitrate Codec.

**INVITE**—A SIP message that initiates a SIP session. It indicates that a user is invited to participate, provides a session description, indicates the type of media, and provides insight regarding the capabilities of the called and calling parties.

**IP**—Internet Protocol. A connectionless protocol that operates at the network layer (Layer 3) of the OSI model. IP provides features for addressing, type-of-service specification, fragmentation and reassemble, and security. Defined in RFC 791. This protocol works with TCP and is usually identified as TCP/IP. See TCP/IP.

**ISDN** —Integrated Services Digital Network.

**Minimum Timer** —Configured minimum value for session interval accepted by SIP elements (proxy, UAC, UAS). This value helps minimize the processing load from numerous INVITE requests.

**Min-SE** —Minimum Session Expiration. The minimum value for session expiration.

**multicast** —A process of transmitting PDUs from one source to many destinations. The actual mechanism (that is, IP multicast, multi-unicast, and so forth) for this process might be different for LAN technologies.

**originator** —User agent that initiates the transfer or Refer request with the recipient.

**PDU** —protocol data units. Used by bridges to transfer connectivity information.

**PER** —Packed Encoding Rule.

**proxy** —A SIP UAC or UAS that forwards requests and responses on behalf of another SIP UAC or UAS.

**proxy server** —An intermediary program that acts as both a server and a client for the purpose of making requests on behalf of other clients. Requests are serviced internally or by passing them on, possibly after translation, to other servers. A proxy interprets and, if necessary, rewrites a request message before forwarding it.

**recipient** —User agent that receives the Refer request from the originator and is transferred to the final recipient.

**redirect server** —A server that accepts a SIP request, maps the address into zero or more new addresses, and returns these addresses to the client. It does not initiate its own SIP request or accept calls.

**re-INVITE** —An INVITE request sent during an active call leg.

**Request URI** —Request Uniform Resource Identifier. It can be a SIP or general URL and indicates the user or service to which the request is being addressed.

**RFC** —Request For Comments.

**RTP** —Real-Time Transport Protocol (RFC 1889)

**SCCP** —Skinny Client Control Protocol.

**SDP**—Session Description Protocol. Messages containing capabilities information that are exchanged between gateways.

**session** —A SIP session is a set of multimedia senders and receivers and the data streams flowing between the senders and receivers. A SIP multimedia conference is an example of a session. The called party can be invited several times by different calls to the same session.

**session expiration** —The time at which an element considers the call timed out if no successful INVITE transaction occurs first.

**session interval** —The largest amount of time that can occur between INVITE requests in a call before a call is timed out. The session interval is conveyed in the Session-Expires header. The UAS obtains this value from the Session-Expires header of a 2xx INVITE response that it sends. Proxies and UACs determine this value from the Session-Expires header in a 2xx INVITE response they receive.

**SIP** —Session Initiation Protocol. An application-layer protocol originally developed by the Multiparty Multimedia Session Control (MMUSIC) working group of the Internet Engineering Task Force (IETF). Their goal was to equip platforms to signal the setup of voice and multimedia calls over IP networks. SIP features are compliant with IETF RFC 2543, published in March 1999.

**SIP URL** —Session Initiation Protocol Uniform Resource Locator. Used in SIP messages to indicate the originator, recipient, and destination of the SIP request. Takes the basic form of *user@host*, where *user* is a name or telephone number, and *host* is a domain name or network address.

**SPI** —service provider interface.

**socket listener** —Software provided by a socket client to receives datagrams addressed to the socket.

**stateful proxy** —A proxy in keepalive mode that remembers incoming and outgoing requests.

**TCP** —Transmission Control Protocol. Connection-oriented transport layer protocol that provides reliable full-duplex data transmissions. TCP is part of the TCP/IP protocol stack. See also TCP/IP and IP.

**TDM** —time-division multiplexing.

**UA** —user agent. A combination of UAS and UAC that initiates and receives calls. See **UAS** and **UAC**.

**UAC** —user agent client. A client application that initiates a SIP request.

**UAS** —user agent server. A server application that contacts the user when a SIP request is received and then returns a response on behalf of the user. The response accepts, rejects, or redirects the request.

**UDP** —User Datagram Protocol. Connectionless transport layer protocol in the TCP/IP protocol stack. UDP is a simple protocol that exchanges datagrams without acknowledgments or guaranteed delivery, requiring that error processing and retransmission be handled by other protocols. UDP is defined in RFC-768.

**URI** —Uniform Resource Identifier. Takes a form similar to an e-mail address. It indicates the user's SIP identity and is used for redirection of SIP messages.

**URL** —Universal Resource Locator. Standard address of any resource on the Internet that is part of the World Wide Web (WWW).

**User Agent** —A combination of UAS and UAC that initiates and receives calls. See **UAS** and **UAC**.

**VFC** —Voice Feature Card.

**VoIP** —Voice over IP. The ability to carry normal telephone-style voice over an IP-based Internet with POTS-like functionality, reliability, and voice quality. VoIP is a blanket term that generally refers to the Cisco standards-based approach (for example, H.323) to IP voice traffic.

