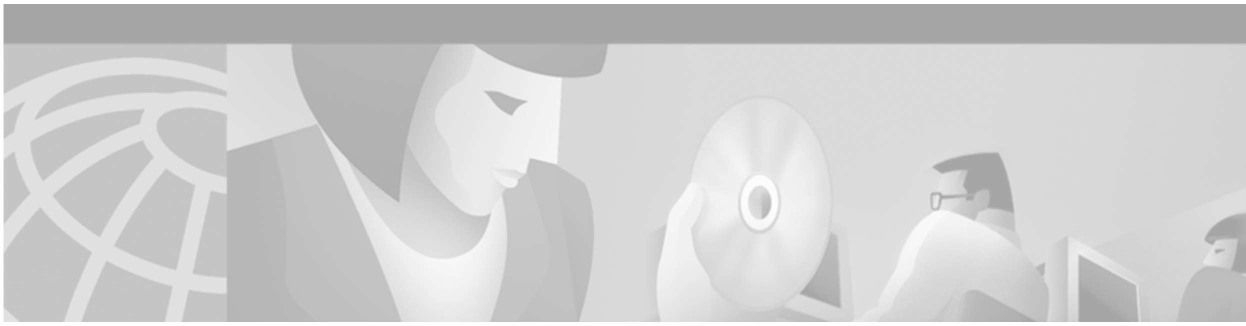

SIP Gateway Support for the Bind Command

Document Update Alert

This document was originally produced for Cisco IOS Release 12.2(11)T. This feature has been updated in subsequent releases, and more recent documentation is available.

If you are using Cisco IOS Release 12.2(11)T or higher, refer to the following section in the Configuring Additional SIP Features chapter of the *Cisco IOS SIP Configuration Guide*, Cisco IOS Voice Configuration Library, Release 12.3:

- *SIP Gateway Support for the bind Command*

Feature History

Release	Modification
12.2(2)XB	This feature was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms.
12.2(2)XB2	This feature was implemented on the Cisco AS5850 platform.
12.2(8)T	This feature was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco 3700 series. The Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms were not supported in this release.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms.

This document describes the SIP Gateway Support for the Bind Command feature. Currently, Session Initiation Protocol (SIP) signaling and media paths use an IP address that is provided by the IP layer as the source address. However, with the addition of the **bind** command, you can now configure the source IP address of signaling packets, or both signaling and media packets.

This document includes the following sections:

Feature Overview

In previous releases of Cisco IOS software, the source address of a packet going out of the gateway was never deterministic. That is, the session protocols and VoIP layers always depended on the IP layer to give the *best local address*. The best local address was then used as the source address (the address showing where the SIP request came from) for signaling and media packets. Using this nondeterministic address occasionally caused confusion for firewall applications, as a firewall could not be configured with an exact address and would take action on several different source address packets.

However, the **bind** interface command allows you to configure the source IP address of signaling and media packets to a specific interface's IP address. Thus, the address that goes out on the packet is bound to the IP address of the interface specified with the **bind** command. Packets that are not destined to the bound address are discarded.

When you do not want to specify a bind address, or if the interface is down, the IP layer still provides the best local address.

The bind command performs different functions based on the state of the interface:

Interface State	Result Using Bind Command
Shutdown **With or without active calls**	The Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) socket listeners are initially closed. (Socket listeners receive datagrams addressed to the socket.) Then the sockets are opened to listen to any IP address. If the outgoing gateway has the **bind** command enabled and has an active call, the call becomes a one-way call with media flowing from the outgoing gateway to the terminating gateway.
No Shutdown **No Active Calls**	The TCP and UDP socket listeners are initially closed. (Socket listeners receive datagrams addressed to the socket.) Then the sockets are opened and bound to the IP address set by the **bind** command. The sockets accept packets destined for the bound address only.
No Shutdown **Active Calls**	The TCP and UDP socket listeners are initially closed. Then the sockets are opened to listen to any IP address.
Bound interface's IP address is **removed**	The TCP and UDP socket listeners are initially closed. Then the sockets are opened to listen to any address, because the IP address has been removed. A message stating that the IP address has been deleted from SIP bound interface is printed. If the outgoing gateway has **bind** enabled and has an active call, the call becomes a one-way call with media flowing from the outgoing gateway to the terminating gateway.

Interface State	Result Using Bind Command
The **physical cable is pulled** on the bound port, or the **Interface layer goes down**	The TCP and UDP socket listeners are initially closed. Then the sockets are opened and bound to listen to any address. When the pulled cable is replaced, the result is as documented for **no shutdown** interfaces.
A **bind interface is shutdown**, or its **IP Address is changed**, or the **physical cable is pulled** while SIP calls are active	The call becomes a one-way call with media flowing in only one direction. It flows from the gateway where the change or shutdown took place, to the gateway where no change occurred. Thus, the gateway with the status change no longer receives media. The call is then disconnected, but the disconnected message is not understood by the gateway with the status change, and the call is still assumed to be active.

Note If there are active calls, the **bind** command will **not** take effect if it is issued for the first time or if it is issued while another **bind** command is in effect. A message is printed reminding you that there are active calls and that the **bind** command change cannot take effect.

Benefits

The SIP Gateway Support for the Bind Command feature has the following benefits:

- With the **bind** command, SIP signaling and media paths can advertise the same source IP address on the gateway for certain applications, even if the paths used different addresses to reach the source. This eliminates confusion for firewall applications that, prior to the use of binding, may have taken action on several different source address packets.

Related Features and Technologies

- Cisco SIP Proxy Server
- Cisco VoIP

Related Documents

The following documents contain information related to the Cisco SIP functionality:

- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2
- *Cisco IOS Voice, Video, and Fax Command Reference*, Release 12.2
- *Cisco IOS IP Configuration Guide,* Release 12.2
- *Cisco IOS IP Command Reference, Volume 1 of 3: Addressing and Services*, Release 12.2
- *Cisco IOS IP Command Reference, Volume 2 of 3: Routing Protocols*, Release 12.2
- *Cisco IOS IP Command Reference, Volume 3 of 3: Multicast*, Release 12.2
- SIP call flows are described in: *SIP Call Flows,* Release 12.2(4)T
- *SIP Gateway Support of RSVP and TEL URL,* Release 12.2(2)XB

Supported Platforms

- Cisco 2600 series
- Cisco 3600 series
- Cisco 3700 series
- Cisco AS5300 universal access server
- Cisco AS5350 universal gateway
- Cisco AS5400 universal gateway
- Cisco AS5850 universal gateway
- Cisco 7200 series

Table 1 Cisco IOS Release and Platform Support for this Feature

Platform	12.2(2)XB	12.2(2)XB1	12.2(8)T	12.2(11)T
Cisco 2600 series	X	X	X	X
Cisco 3600 series	X	X	X	X
Cisco 3700 series	Not supported	Not supported	X	X
Cisco 7200 series	X	X	X	X
Cisco AS5300	X	X	Not supported	X
Cisco AS5350	X	X	Not supported	X
Cisco AS5400	X	X	Not supported	X
Cisco AS5850 universal gateway	Not supported	X	Not supported	X

Determining Platform Support Through Cisco Feature Navigator

Cisco IOS software is packaged in feature sets that support specific platforms. To get updated information regarding platform support for this feature, access Cisco Feature Navigator. Cisco Feature Navigator dynamically updates the list of supported platforms as new platform support is added for the feature.

Cisco Feature Navigator is a web-based tool that enables you to determine which Cisco IOS software images support a specific set of features and which features are supported in a specific Cisco IOS image. You can search by feature or release. Under the release section, you can compare releases side by side to display both the features unique to each software release and the features in common.

To access Cisco Feature Navigator, you must have an account on Cisco.com. If you have forgotten or lost your account information, send a blank e-mail to cco-locksmith@cisco.com. An automatic check will verify that your e-mail address is registered with Cisco.com. If the check is successful, account details with a new random password will be e-mailed to you. Qualified users can establish an account on Cisco.com by following the directions at http://www.cisco.com/register.

Cisco Feature Navigator is updated regularly when major Cisco IOS software releases and technology releases occur. For the most current information, go to the Cisco Feature Navigator home page at the following URL:

http://www.cisco.com/go/fn

Availability of Cisco IOS Software Images

Platform support for particular Cisco IOS software releases is dependent on the availability of the software images for those platforms. Software images for some platforms may be deferred, delayed, or changed without prior notice. For updated information about platform support and availability of software images for each Cisco IOS software release, refer to the online release notes or, if supported, Cisco Feature Navigator.

Supported Standards, MIBs, and RFCs

Standards

No new or modified standards are supported by this feature.

MIBs

- CISCO-SIP-UA-MIB

To obtain lists of supported MIBs by platform and Cisco IOS release, and to download MIB modules, go to the Cisco MIB website on Cisco.com at the following URL:

http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml.

RFCs

- RFC 2543, *SIP: Session Initiation Protocol*
- RFC 2806, *URLs for Telephone Calls*

Prerequisites

The following are general prerequisites for SIP deployment.

- Ensure that your Cisco 2600 series, Cisco 3600 series, or Cisco 7200 series router has 16-MB Flash memory and 64-MB DRAM memory, minimum. A Cisco AS5300 must have a minimum of 16-MB Flash memory and 128-MB DRAM memory. A Cisco AS5400 must have a minimum of 32-MB Flash memory and 256-MB DRAM memory.
- Ensure the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network.

  For more information about configuring IP, refer to:
  Cisco IOS IP Configuration Guide, Release 12.2.
- Configure VoIP.

  For more information about configuring VoIP, refer to:
  Cisco IOS Voice, Video, and Fax Command Reference, Release 12.2.
- Set the bind address prior to using the **bind** command.

Setting the Bind Address

To set the bind address, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`dial-peer voice`** *`number`* **`voip`**	Enters dial peer mode to configure a VoIP dial-peer.
Step 2	`Router(config-dial-peer)#` **`session target ipv4`**`:`*`destination address`*	Specifies a network-specific address for a dial peer. This command must be set to the bind address of the receiving gateway before using the **bind** command. **ipv4**:*destination address:* Sets the IP address of the dial peer. A valid IP address is in this format: *xxx.xxx.xxx.xxx*
Step 3	`Router(config-dial-peer)#` **`exit*`**	Exits dial peer configuration mode.

Configuration Tasks

See the following section for configuration tasks for the SIP Gateway Support for the Bind Command feature. Each task in the list is identified as either required or optional.

- Configuring the Bind Command (required)

Configuring the Bind Command

SIP configuration mode starts from voice-service VoIP configuration mode. When in SIP configuration mode several options are available, including the **bind** command.

To configure the **bind** command, complete the following steps beginning in global configuration mode:

	Command	Purpose
Step 1	`Router(config)#` **`voice service voip`**	Enters the voice-service VoIP configuration mode. (Required)
Step 2	`Router(config-voi-serv)#` **`sip`**	Enters the SIP configuration mode. (Required)
Step 3	`Router(conf-serv-sip)#` **`session transport`** `{`**`udp`** `|` **`tcp`**`}`	Sets the session transport type for the SIP user agent. The default is UDP. (Optional) The transport protocol (**udp** or **tcp**) specified with the **session transport** command, and the protocol specified with the **transport** command, must be identical.
Step 4	`Router(conf-serv-sip)#` **`bind`** `{`**`control`** `|` **`all`**`}` **`source-interface`** *`interface-id`*	Sets a source address for signaling and media packets. (Required) **control**: Binds SIP signaling packets. **all**: Binds SIP signaling packets and media packets. **source-interface**: Specifies an interface as the source address of SIP packets, where *interface-id* specifies the type of interface: Async, BVI, CTunnel, Dialer, Ethernet, FastEthernet, Lex, Loopback, Multilink, Null, Serial, Tunnel, Vif, Virtual-Template, or Virtual-TokenRing.
Step 5	`Router(conf-serv-sip)#` **`default`** `{`*`command`*`}`	Resets the default value of a SIP command. (Optional)
Step 6	`Router(conf-serv-sip)#` **`exit`**	Exits dial peer configuration mode.

Verifying Configurations

There are two **show** commands that verify the correct settings for the **bind** command. The first enables you to verify a bound IP address. The second indicates the status of bind (enabled or disabled):

- Verifying a Bound IP Address
- Verifying Bind Status

Verifying a Bound IP Address

The following examples show output for the **show ip socket** command, indicating that the bind address of the receiving gateway is set.

```
Router# show ip socket
Proto    Remote       Port      Local        Port  In Out Stat TTY OutputIF
17 0.0.0.0             0 --any--          2517   0   0    9   0
17 --listen--          172.18.192.204   1698   0   0    1   0
17 0.0.0.0             0 172.18.192.204     67   0   0  489   0
17 0.0.0.0             0 172.18.192.204   5060   0   0   A1   0
```

Verifying Bind Status

The following example shows output for the **show sip-ua status** command, indicating that bind is enabled.

```
Router# show sip-ua status
SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): ENABLED 172.18.192.204
SIP User Agent bind status(media): ENABLED 172.18.192.204
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)
```

Troubleshooting Tips

To troubleshoot this feature, perform the following:

- Use the **debug ccsip all** command to enable all SIP debugging capabilities, or use one of the following SIP debug commands:
  - **debug ccsip calls**
  - **debug ccsip error**
  - **debug ccsip events**
  - **debug ccsip messages**
  - **debug ccsip states**
- Use the **show ip socket** command to display IP socket information.
- Use the **show sip-ua status** command to verify if binding is enabled. See **show sip-ua status** for details on this command.

- Use the **show running configuration** command under voice-service VoIP mode to verify bind functionality.

Configuration Examples

This section shows partial output from the **show running config** command indicating that bind is functional on receiving router `172.18.192.204`.

```
ip subnet-zero
ip ftp source-interface Ethernet0
!
voice service voip
 sip
  bind all source-interface FastEthernet0
!
interface FastEthernet0
 ip address 172.18.192.204 255.255.255.0
 duplex auto
 speed auto
 fair-queue 64 256 1000
 ip rsvp bandwidth 75000 100
!!
```

Command Reference

This section documents new or modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.2 command reference publications.

New Commands

- **bind**
- **sip**

Modified Commands

- **default (SIP)**
- **session transport**
- **show sip-ua status**

bind

To configure the source address for signaling and media packets to the IP address of a specific interface, use the **bind** command in SIP configuration mode. To disable binding, use the **no** form of the command.

bind {**control** | **all**} **source-interface** *interface-id*

no bind

Syntax Description

control	Binds SIP signaling packets.
all	Binds SIP signaling and media packets. The source address (the address that shows where the SIP request came from) of the signaling and media packets is set to the IP address of the interface specified.
source-interface	Specifies an interface as the source address of SIP packets
interface-id	Specifies one of the following interfaces: **Async**: Asynchronous Transfer Mode interface BVI: Bridge-Group Virtual Interface CTunnel: CTunnel interface Dialer: Dialer interface **Ethernet**: IEEE 802.3 **FastEthernet**: Fast Ethernet Lex: Lex interface **Loopback**: Loopback interface Multilink: Multilink-group interface Null: Null interface **Serial**: Serial Interface (Frame Relay) Tunnel: Tunnel interface Vif: PGM Multicast Host interface Virtual-Template: Virtual Template interface Virtual-TokenRing: Virtual Token Ring

Note **Async, Ethernet**, **Fast Ethernet**, **Loopback**, and **Serial** (including Frame Relay) are tested interfaces within the SIP application.

Defaults

The default is that binding is disabled (**no bind**).

Command Modes

SIP configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms.
12.2(2)XB2	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco 3700 series. Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms.

Usage Guidelines

If the **bind** command is not enabled, the IP layer will still provide the best local address.

Enter the SIP configuration mode from voice-service configuration mode, as shown in the example:

Examples

The following example shows how to set up binding on a SIP network:

```
Router(config)# voice serv voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# bind control source-interface FastEthernet 0
```

Related Commands

Command	Description
sip	Enter SIP configuration mode from voice-service VoIP configuration mode.

default (SIP)

To reset a SIP command to its default value, use the **default** command in SIP configuration mode.

default *command*

Syntax Description

command	One of the SIP configuration commands. Valid choices are: • **bind**: Configures the source address of signaling and media packets to a specific interface's IP address. • **rel1xx**: Enables all SIP provisional responses (other than 100 Trying) to be sent reliably to the remote SIP endpoint. • **session transport**: Configures the underlying transport layer protocol for SIP messages to TCP or UDP. • **url**: Configures URLs to either the SIP or TEL format for your voip sip calls.

Defaults

The default is that binding is disabled (**no bind**).

Command Modes

SIP configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms.
12.2(2)XB2	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco 3700 series. Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms.

Examples

The following example shows how to reset the value of the SIP **bind** command:

```
Router(config)# voice serv voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# default bind
```

Related Commands

Command	Description
sip	Enter SIP configuration mode from voice-service VoIP configuration mode.

session transport

To configure the underlying transport layer protocol for SIP messages to Transmission Control Protocol (TCP) or User Datagram Protocol (UDP), use the **session transport** command in SIP configuration mode. To reset the value of this command to the default, use the **no** form of this command.

session transport {**udp** | **tcp**}

no session transport

Syntax Description

udp	Configure SIP messages to use the UDP transport layer protocol.
tcp	Configure SIP messages to use the TCP transport layer protocol.

Defaults

The default for this command is UDP.

Command Modes

SIP configuration

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and the Cisco AS5300 platforms.
12.2(2)XA	Support was added for the Cisco AS5350 and Cisco AS5400 platforms.
12.2(2)XB	This command was moved from dial-peer mode configuration to SIP configuration mode.
12.2(2)XB2	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco 3700 series. Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms.

Usage Guidelines

Use **show sip-ua status** to verify that the transport protocol set with the **session transport** command matches the protocol set using the **transport** command under SIP user-agent configuration mode.

Examples

The following example configures the underlying transport layer protocol for SIP messages to UDP:

```
Router(config)# voice serv voip
Router(conf-voi-serv)# sip
Router(conf-serv-sip)# session transport udp
```

Related Commands

Command	Description
transport	Used in SIP user-agent configuration mode to configure the SIP gateway for SIP signaling messages on inbound calls through the SIP TCP or UDP socket.

show sip-ua status

To show SIP UA status, use the **show sip-ua status** command in privileged EXEC mode.

show sip-ua status

Syntax Description

This command has no arguments or keywords.

Defaults

There are no default behaviors or values for this command.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(1)T	This command was introduced on the Cisco 2600 series, Cisco 3600 series, and the Cisco AS5300 platforms.
12.1(3)T	The statistics portion of the output was removed and is now included in the **show sip-ua statistics** command.
12.2(2)XA	Support was added for the Cisco AS5350 and Cisco AS5400 platforms.
12.2(2)XB	Support was added for the Cisco AS5350 and Cisco AS5400 platforms. Command output was enhanced to display: • If signaling or media binding is enabled. • The style of DNS SRV query: 1 for RFC 2052 or 2 for RFC 2782.
12.2(2)XB2	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco 3700 series. Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms.

Usage Guidelines

Use the **show sip-ua status** command to verify SIP configurations.

Examples

The following is sample output from the **show sip-ua status** command and shows bind status as disabled:

```
Router# show sip-ua status
SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)
```

Table 2 show sip-ua timers Command Field Descriptions

Field	Description
SIP User Agent Status	Indicates UA status.
SIP User Agent for UDP	Toggle; UDP enabled or disabled.
SIP User Agent for TCP	Toggle; TCP enabled or disabled.
SIP User Agent bind status (signaling)	Toggle; Binding for signaling is enabled or disabled.
SIP User Agent bind status (media)	Toggle; Binding for media is enabled or disabled.
SIP max-forwards	Value of max-forwards of SIP messages.
SIP DNS SRV version	Style of DNS SRV query: 1 for RFC2052 or 2 for RFC2782

Related Commands

Command	Description
show sip-ua statistics	Displays response, traffic, and retry SIP statistics.
show sip-ua retry	Displays SIP retry statistics.
show sip-ua timers	Displays the current settings for SIP UA timers.

sip

To enter the SIP configuration mode, use the **sip** command in voice-service VoIP configuration mode.

sip

Syntax Description

This command has no arguments or keywords.

Defaults

No default behavior or values.

Command Modes

Voice-service VoIP configuration

Command History

Release	Modification
12.2(2)XB	This command was introduced on the Cisco 2600 series, Cisco 3600 series, Cisco 7200 series, Cisco AS5300, Cisco AS5350, and Cisco AS5400 platforms.
12.2(2)XB2	This command was implemented on the Cisco AS5850 platform.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and support was added for the Cisco 3700 series. Cisco AS5300, Cisco AS5350, Cisco AS5850, and Cisco AS5400 platforms were not supported in this release.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 platforms.

Usage Guidelines

From the voice-service VoIP configuration mode, the **sip** command enables you to enter SIP configuration mode. From this mode, several SIP commands are available, such as **bind**, **session transport**, and **url**.

Examples

The following example illustrates entering SIP configuration mode and then setting the **bind** command on the SIP network:

```
Router(config)# voice service voip
Router(config-voi-srv)# sip
Router(conf-serv-sip)# bind control source-interface FastEthernet 0
```

Related Commands

Command	Description
voice service voip	Enters the voice-service configuration mode.
session transport	Configures the voice dial peer to use Transmission Control Protocol (TCP) or User Datagram Protocol (UDP) as the underlying transport layer protocol for SIP messages.

Glossary

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.

CLI—command-line interface.

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.

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.

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.

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

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.

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.

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.

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.

SPI—service provider interface.

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

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.

UAC—User Agent Client. A client application that initiates a SIP request.

UAS—User Agent Server (or user agent). A server application that contacts the user when a SIP request is received, 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.

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**.

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.