SIP Gateway Support for the bind Command

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The Gateway Support for the bind Command feature introduces the **bind** command, which allows you to configure the source IP address of signaling packets or both signaling and media packets. Befor this feature was introduced the source address of a packet going out of a Cisco IOS gateway is not deterministic. The session protocols and VoIP layers depended on the IP layer to give the best local address and then used the address for the source address in signaling or media or both, even if multiple interfaces can support a route to the destination address.

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest feature information and caveats, see 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 document.

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. An account on Cisco.com is not required.

Prerequisites for SIP Gateway Support for the bind Command

Cisco Unified Border Element

- Cisco IOS Release 12.2(8)T or a later release must be installed and running on your Cisco Unified Border Element.

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Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 2.5 or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Information About SIP Gateway Support for the bind Command

Prior to the Gateway Support for the bind Command feature 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, because a firewall could not be configured with an exact address and would take action on several different source address packets.

The **bind** interface command allows you to configure a specific interface's IP address as the source IP address of signaling and media packets. 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 specify a bind address, or if the interface is down, the IP layer still provides the best local address.

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 use different addresses to reach the source. This eliminates confusion for firewall applications that, Without the binding, may have taken action on several different source address packets.

The table below lists the results of the bind command based on the state of the interface.

Table 1   ***Command functions for the bind command based on the state of the interface***

Interface State	Result Using Bind Command
A **bind interface is shut down**, 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.
No Shutdown-- With 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.

Interface State	Result Using Bind Command
No Shutdown-- With active calls.	The TCP and UDP socket listeners are initially closed.
	Then the sockets are opened to listen to any IP address.
Shutdown-- With or without active calls.	The 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.
The 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 displayed.
	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.
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.

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 displayed reminding you that there are active calls and that the **bind** command change cannot take effect.

How to Configure SIP Gateway Support for the bind Command

Setting the Bind Address

To set the bind address, complete the task in this section.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *number* **voip**
4. **session target ipv4:** *destination-address*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** Router> enable	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** Router# configure terminal	Enters global configuration mode.
Step 3	**dial-peer voice** *number* **voip** **Example:** Router(config)# dial-peer voice 2 voip	Enters dial peer configuration mode to configure a VoIP dial-peer.
Step 4	**session target ipv4:** *destination-address* **Example:** Router(config-dial-peer)# session target ipv4: 172.16.43.3	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 5	**exit** **Example:** Router(config-dial-peer)# exit	Exits dial peer voice configuration mode.

Setting a Source IP Address for Signaling and Media Packets

SIP configuration mode starts from voice-service VoIP configuration mode. When the router is in SIP configuration mode, several options are available, including the **bind** command. To enable this feature, review the prerequisites to make sure your network is compliant, and then complete the task in this section.

- Endure you have Cisco IOS XE Release 2.5 or a later release installed and running on your Cisco ASR 1000 Series Router.
- Ensure that the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network.
- Configure VoIP.
- Set the bind address prior to using the **bind** command.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **session transport** {**udp** | **tcp**}
6. **bind** {**control** | **all**} **source-interface** *interface-id*
7. **default** {**bind|rel1xx|session-transport|url**}
8. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** Router> enable	Enables privileged EXEC mode. - Enter your password if prompted.
Step 2	**configure terminal** **Example:** Router# configure terminal	Enters global configuration mode.
Step 3	**voice service voip** **Example:** Router(config)# voice service voip	Enters voice-service configuration mode
Step 4	**sip** **Example:** Router(config-voi-srv)# sip	Enters the SIP configuration mode.

Command or Action	Purpose
Step 5 **session transport** {**udp** | **tcp**} **Example:** Router(conf-serv-sip)# session transport udp	(Optional) Sets the session transport type for the SIP user agent. • The default is UDP. • The transport protocol (**udp** or **tcp**) specified with the **session transport**command, and the protocol specified with the **transport** command, must be identical.
Step 6 **bind** {**control** | **all**} **source-interface** *interface-id* **Example:** Router(conf-serv-sip)# bind all source- interface fastethernet	Sets a source address for signaling and media packets. • **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. • *interface-id* argument specifies the type of interface: 　◦ Async 　◦ BVI 　◦ CTunnel 　◦ Dialer 　◦ Ethernet 　◦ FastEthernet 　◦ Lex 　◦ Loopback 　◦ Multilink 　◦ Null 　◦ Serial 　◦ Tunnel 　◦ Vif 　◦ Virtual-Template 　◦ Virtual-TokenRing
Step 7 **default** {**bind**|**rel1xx**|**session-transport**|**url**} **Example:** Router(conf-serv-sip)# bind	(Optional) Resets the default value of a SIP command. • **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

	Command or Action	Purpose
Step 8	**exit** **Example:** `Router(conf-serv-sip)# exit`	Exits the current configuration mode.

Verifying and Troubleshooting Tips

Two **show** commands 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

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)

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 the **show sip-ua status**command for details.

Configuration Examples for SIP Gateway Support for the bind Command

SIP Gateway Support for the bind Command Example

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
!!
```

Feature Information for SIP Gateway Support for the bind Command

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.

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Feature History Table entry for the Cisco Unified Border Element.

Table 2 **Feature Information for SIP: Gateway Support for the bind Command**

Feature Name	Releases	Feature Information
SIP: Gateway Support for the bind Command	12.2(8)T, 12.3(2)T, 12.2(11)T, 12.2(15)T	In Cisco IOS XE Release 2.5, This feature was introduced on the Cisco ASR 1000 Series Routers. The following commands were introduced or modified: **bind**and **sip**.

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

Table 3 **Feature Information for SIP: Gateway Support for the bind Command**

Feature Name	Releases	Feature Information
SIP: Gateway Support for the bind Command	Cisco IOS XE Release 2.5	In Cisco IOS XE Release 2.5, This feature was introduced on the Cisco ASR 1000 Series Routers. The following commands were introduced or modified: **bind**and **sip**.