



# SIP: ISDN Suspend/Resume Support

The SIP: ISDN Suspend/Resume Support feature adds Session Initiation Protocol (SIP) call-hold support to SIP gateways when an ISDN Suspend event is triggered. Because Suspend and Resume support already exists for H.323, the SIP implementation of Suspend and Resume provides feature parity.

## Feature Specifications for SIP: ISDN Suspend/Resume Support

### Feature History

Release	Modification
12.2(15)T	This feature was introduced.

### Supported Platforms

Cisco 1751, Cisco 2610–Cisco 2613, Cisco 2620, Cisco 2621, Cisco 2650, Cisco 2651, Cisco 2691, Cisco 3620, Cisco 3631, Cisco 3640, Cisco 3660, Cisco 3725, Cisco 3745, Cisco 7200 series, Cisco AS5300, Cisco AS5350, Cisco AS5400, Cisco AS5850, Cisco CVA120 series, and Cisco uBR925.

## Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at <http://www.cisco.com/go/fn>. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

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## Prerequisites for SIP: ISDN Suspend/Resume Support

The following are general prerequisites for SIP deployment:

- Ensure that the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network.

For more information about configuring IP, refer to the following document:

[Cisco IOS IP Configuration Guide](#), Release 12.2

- Configure VoIP.

For more information about configuring VoIP, refer to the following document:

[Cisco IOS Voice, Video, and Fax Configuration Guide](#), Release 12.2

## Information About SIP: ISDN Suspend/Resume Support

To configure the SIP: ISDN Suspend/Resume Support feature, you need to understand the following concepts:

- [Suspend and Resume Overview](#), page 2
- [SIP Call-Hold Process](#), page 2

## Suspend and Resume Overview

Suspend and Resume are basic functions of ISDN and ISDN User Part (ISUP) signaling procedures and now are a part of SIP functionality. Suspend is described in ITU Q.764 as a message that indicates a temporary cessation of communication that does not release the call. A Suspend message can be accepted during a conversation. A Resume message is received after a Suspend message and is described in ITU Q.764 as a message that indicates a request to recommence communication. If the calling party requests to release the call, the Suspend and Resume sequence is overridden.

## SIP Call-Hold Process

When a SIP originating gateway receives an ISDN Suspend message, the originating gateway informs the terminating gateway that there is a temporary cessation of media; that is, the call is placed on hold. There are two ways that SIP gateways receive notice of a call hold. The first way is for the originating gateway to use a connection IP address of 0.0.0.0 (c=0.0.0.0) in the Session Description Protocol (SDP). The information in the SDP is sent in a re-Invite to the terminating gateway. The second way is for the originating gateway to use a=sendonly in the SDP of a re-Invite.



Note

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For a SIP gateway to initiate call hold, the c=0.0.0.0 method must be used.

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The purpose of the c=0.0.0.0 line is to notify the terminating gateway to stop sending media packets. When the hold is cancelled and communication is to resume, an ISDN Resume message is sent. The SIP originating gateway takes the call off hold by sending out a re-Invite with the actual IP address of the remote SIP entity in the c= line (in place of 0.0.0.0).

Multiple media fields (m-lines) in the SDP of a re-Invite message are used to indicate media forking, with each m-line representing one media destination. SIP gateways negotiate multiple media streams by using multiple m- and/or c-lines. When an originating gateway receives an ISDN Suspend on a gateway that has negotiated multiple media streams, all of the media streams are placed on hold. The originating gateway sends out a re-Invite that has a c= line that advertises the IP address as 0.0.0.0 on all streams. The originating gateway also mutes the SIP calls for each media stream so that no media is sent to the terminating gateway. When the originating gateway receives an ISDN Resume, it initiates a re-Invite with the original SDP and takes the call off hold.

If the media inactivity timer is configured on the network, the timer is stopped for all active streams. The purpose of the media inactivity timer is to monitor and disconnect calls if no Real-Time Control Protocol (RTCP) packets are received within a configurable time period. However, on initiating the call hold, the originating gateway disables the media inactivity timer for that particular call, so the call remains active. The terminating gateway behaves in the same way when it receives the call-hold re-Invite from the originating gateway. When the call resumes, the originating gateway reenables the Media Inactivity Timer. For more information on the timer, refer to the *SIP Media Inactivity Timer* document.

All billing and accounting procedures are unaffected by the SIP: ISDN Suspend/Resume Support feature.

## How to Configure Suspend and Resume Support

This section contains the following procedure:

- [Configuring Suspend and Resume, page 3](#) (required)

### Configuring Suspend and Resume

Suspend and Resume functionality is enabled by default. However, Suspend and Resume functionality is also configurable. To configure Suspend and Resume for all dial peers on the VoIP network, enter the commands shown below on the originating and terminating gateways.

#### SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **sip-ua**
4. **suspend-resume**

## DETAILED STEPS

	Command or Action	Purpose
Step 1	<code>enable</code>  <b>Example:</b> Router> <code>enable</code>	Enables privileged EXEC mode. <ul style="list-style-type: none"><li>• Enter your password if prompted.</li></ul>
Step 2	<code>configure terminal</code>  <b>Example:</b> Router# <code>configure terminal</code>	Enters global configuration mode.
Step 3	<code>sip-ua</code>  <b>Example:</b> Router(config)# <code>sip-ua</code>	Enters SIP user agent configuration mode.
Step 4	<code>suspend-resume</code>  <b>Example:</b> Router(config-sip-ua)# <code>suspend-resume</code>	Enables support for Suspend and Resume.

## Troubleshooting Tips

To troubleshoot this feature, perform the following steps:

- Use the **show sip-ua status** command to display whether Suspend and Resume support is enabled or 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)
SDP application configuration:
  Version line (v=) required
  Owner line (o=) required
  Session name line (s=) required
  Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udpt1
SIP support for ISDN SUSPEND/RESUME: ENABLED
```

- Make sure that you can make a voice call.
- 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 info**
- **debug ccsip media**
- **debug ccsip messages**
- **debug ccsip states**

## Configuration Examples for SIP: ISDN Suspend/Resume Support

This section provides the following configuration example:

- [SIP: ISDN Suspend/Resume Support Example, page 5](#)

### SIP: ISDN Suspend/Resume Support Example

The following example contains output from the **show running-config** command. Because SIP Suspend and Resume is enabled by default on the gateway, this example shows SIP Suspend and Resume disabled on the gateway.

```
Router# show running-config

Building configuration...

Current configuration : 2791 bytes
!
version 12.2
service config
no service single-slot-reload-enable
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
service internal
service udp-small-servers
!
interface FastEthernet2/0
ip address 172.18.200.24 255.255.255.0
duplex auto
no shut
speed 10
ip rsvp bandwidth 7500 7500
!
voice-port 1/1/1
no supervisory disconnect lcfo
!
dial-peer voice 1 pots
application session
destination-pattern 8185551111
port 1/1/1
!
dial-peer voice 3 voip
application session
destination-pattern 7175551111
session protocol sipv2
session target ipv4:172.18.200.36
codec g711ulaw
!
```

```
dial-peer voice 4 voip
application session
destination-pattern 6165551111
session protocol sipv2
session target ipv4:172.18.200.33
codec g711ulaw
!
gateway
!
sip-ua
no suspend-resume
retry invite 1
retry bye 1
!
line con 0
line aux 0
line vty 0 4
login
!
end
```

## Additional References

For additional information related to SIP: ISDN Suspend/Resume Support, refer to the following references:

## Related Documents

Related Topic	Document Title
Cisco SIP Functionality	<a href="#">Cisco IOS Voice, Video, and Fax Configuration Guide</a> , Release 12.2 <a href="#">Cisco IOS Voice, Video, and Fax Command Reference</a> , Release 12.2 T <a href="#">Session Initiation Protocol (SIP) for VoIP</a> , Release 12.2(8)T <a href="#">Session Initiation Protocol Gateway Call Flows</a> , Release 12.2(4)T <a href="#">Call Transfer Capabilities Using the Refer Method</a> , Release 12.2(8)T <a href="#">SIP Media Inactivity Timer</a> , Release 12.2(8)T <a href="#">Dual Tone Multifrequency Relay for SIP Calls Using Named Telephone Events</a> , Release 12.2(2)XB
Cisco IOS References	<a href="#">Cisco IOS Debug Command Reference</a> , Release 12.2 T <a href="#">Cisco IOS IP Configuration Guide</a> , Release 12.2 <a href="#">Cisco IOS IP Command Reference, Volume 1 of 3: Addressing and Services</a> , Release 12.2 T <a href="#">Cisco IOS IP Command Reference, Volume 2 of 3: Routing Protocols</a> , Release 12.2 T <a href="#">Cisco IOS IP Command Reference, Volume 3 of 3: Multicast</a> , Release 12.2 T <a href="#">Cisco IOS Dial Technologies Configuration Guide</a> , Release 12.2

## Standards

Standards <sup>1</sup>	Title
ANSI T1.113	<i>ISDN User Part</i>
Q.764	<i>Signalling System No. 7 - ISDN User Part signalling procedures</i>
Q.931	<i>ISDN user-network interface layer 3 specification for basic call control</i>
<i>draft-ietf-sip-isup-03.txt</i>	<i>ISUP to SIP Mapping draft</i>

1. Not all supported standards are listed.

## MIBs

MIBs <sup>1</sup>	MIBs Link
<ul style="list-style-type: none"> <li>CISCO-SIP-UA-MIB</li> </ul>	<p>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:</p> <p><a href="http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml">http://www.cisco.com/public/sw-center/netmgmt/cmtk/mibs.shtml</a></p>

1. Not all supported MIBs are listed.

To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL:

<http://tools.cisco.com/ITDIT/MIBS/servlet/index>

If Cisco MIB Locator does not support the MIB information that you need, you can also obtain a list of supported MIBs and download MIBs from the Cisco MIBs page at the following URL:

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

To access Cisco MIB Locator, 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](mailto: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 found at this URL:

<http://www.cisco.com/register>

## RFCs

RFCs <sup>1</sup>	Title
RFC 2543	<i>SIP: Session Initiation Protocol</i>

1. Not all supported RFCs are listed.

## Technical Assistance

Description	Link
Technical Assistance Center (TAC) home page, containing 30,000 pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	<a href="http://www.cisco.com/public/support/tac/home.shtml">http://www.cisco.com/public/support/tac/home.shtml</a>

## Command Reference

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

### New Commands

- [suspend-resume \(SIP\)](#)

### Modified Commands

- [show sip-ua status](#)

# suspend-resume (SIP)

To enable SIP Suspend and Resume functionality, use the **suspend-resume** command in SIP user agent configuration mode. To disable SIP Suspend and Resume functionality, use the **no** form of this command.

**suspend-resume**

**no suspend-resume**

**Syntax Description** This command has no arguments or keywords.

**Defaults** Enabled

**Command Modes** SIP user agent configuration

Release	Modification
12.2(15)T	This command was introduced.

**Usage Guidelines** Session Initiation Protocol (SIP) gateways are now enabled to use Suspend and Resume. Suspend and Resume are basic functions of ISDN and ISDN User Part (ISUP) signaling procedures. A Suspend message temporarily halts communication (call hold), and a Resume message is received after a Suspend message and continues the communication.

**Examples** The following example disables Suspend and Resume functionality:

```
Router(config)# sip-ua
Router(config-sip-ua)# no suspend-resume
```

Command	Description
<b>show sip-ua status</b>	Displays SIP UA status.
<b>sip-ua</b>	Enables the SIP user-agent configuration commands.

# show sip-ua status

To display status for the Session Initiation Protocol (SIP) user agent (UA), use the **show sip-ua status** command in privileged EXEC mode.

## show sip-ua status

**Syntax Description** This command has no arguments or keywords.

**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 Cisco AS5300.
	12.1(3)T	The statistics portion of the output was removed and included in the <b>show sip-ua statistics</b> command.
	12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.
	12.2(2)XB	Command output was enhanced to display whether media or signaling binding is enabled and the style of DNS SRV query: <ul style="list-style-type: none"> <li>• 1 for RFC 2052</li> <li>• 2 for RFC 2782</li> </ul>
	12.2(2)XB1	This command was implemented on the Cisco AS5850.
	12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support for the Cisco AS5300, Cisco AS5350, and Cisco AS5400 is not included in this release. For the purposes of display, this command was separated from the generic <b>show sip-ua</b> command.
	12.2(11)T	Command output was enhanced to display information on Session Description Protocol (SDP) application configuration. This command is supported on the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850 in this release.
	12.2(13)T	Command output was enhanced to display the following: <ul style="list-style-type: none"> <li>• Information on redirection message handling.</li> <li>• Information on handling of 180 responses with SDP.</li> </ul>
	12.2(15)T	Command output was enhanced to display Suspend and Resume support.

**Usage Guidelines** Use this command to verify SIP configurations.

**Examples** The following is sample output from the **show sip-ua status** command:

```
Router# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
```

## ■ show sip-ua status

```

SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: DISABLED
SIP max-forwards : 6
SIP DNS SRV version: 1 (rfc 2052)
Redirection (3xx) message handling: ENABLED
SDP application configuration:
  Version line (v=) required
  Owner line (o=) required
  Session name line (s=) required
  Timespec line (t=) required
  Media supported: audio image
  Network types supported: IN
  Address types supported: IP4
  Transport types supported: RTP/AVP udpt1
SIP support for ISDN SUSPEND/RESUME: ENABLED

```

Table 1 describes significant fields in this output.

**Table 1** *show sip-ua status Field Descriptions*

Field	Description
SIP User Agent Status	UA status.
SIP User Agent for UDP	UDP is enabled or disabled.
SIP User Agent for TCP	TCP is enabled or disabled.
SIP User Agent bind status (signaling)	Binding for signaling is enabled or disabled.
SIP User Agent bind status (media)	Binding for media is enabled or disabled.
SIP early-media for 180 responses with SDP	Early media cut-through treatment for 180 responses with SDP can be enabled, the default treatment, or disabled, with local ringback provided.
SIP max-forwards	Value of max-forwards of SIP messages.
SIP DNS SRV version	Style of DNS SRV query: 1 for RFC 2052 or 2 for RFC 2782.
Redirection (3xx) message handling	Redirection can be enabled, which is the default status as per RFC 2543. Or handling of redirection 3xx messages can be disabled, allowing the gateway to treat 3xx redirect messages as 4xx error messages.
Version	SDP version.
Owner	Session originator.
Session name	Identifies the session name.
Timespec	Identifies the session start and stop times.
Media supported	Media information.
Network types supported	Always IN for Internet.
Address types supported	Identifies the Internet Protocol version.
Transport type supported	Identifies the transport protocols supported.
SIP support for ISDN SUSPEND/RESUME	Suspend and Resume support is enabled or disabled.

**Related Commands**

<b>Command</b>	<b>Description</b>
<b>show sip-ua retry</b>	Displays SIP retry statistics.
<b>show sip-ua statistics</b>	Displays response, traffic, and retry SIP statistics.
<b>show sip-ua timers</b>	Displays the current settings for SIP UA timers.
<b>sip-ua</b>	Enables the SIP user-agent configuration commands.

# Glossary

**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.

**ISDN**—Integrated Services Digital Network. Communication protocol offered by telephone companies that permits telephone networks to carry data, voice, and other source traffic.

**ISUP**—ISDN User Part. Provides the interexchange signaling to support SS7 trunks that are set up for switched voice and data applications in an ISDN environment.

**OGW**—Cisco originating gateway.

**PSTN**—public switched telephone network.

**RES**—ISUP Resume message defined in ITU Q.764.

**Resume**—ISDN Resume message defined in ITU Q.931.

**RTCP**—RTP Control Protocol. The protocol monitors an RTP connection and conveys information about the ongoing session.

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

**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.

**SUS**—ISUP Suspend message defined in ITU Q.764.

**Suspend**—ISDN Suspend message defined in ITU Q.931.

**TGW**—Cisco terminating gateway.

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

**VoIP**—Voice over IP. The ability to carry normal telephone-style voice over an IP-based network.



## Note

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Refer to the [Internetworking Terms and Acronyms](#) for terms not included in this glossary.

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