# Verifying and Troubleshooting SIP Features

This chapter describes how to verify and troubleshoot Cisco SIP features.

## Contents



## Basic Troubleshooting Procedures

Cisco routers provide numerous integrated commands to assist you in monitoring and troubleshooting your internetwork:

- **show** commands help you monitor installation behavior and normal network behavior, and isolate problem areas.
- **debug** commands help you isolate protocol and configuration problems.
- **ping** commands help you determine connectivity between devices on your network.
- **trace** commands provide a method of determining the route by which packets reach their destination.

This chapter discusses use of **show** and **debug** commands.

**Note** Under moderate traffic loads, **debug** commands produce a high volume of output. We therefore recommend that, as a general rule, you use **show** commands first and use **debug** commands with caution.

Generally, you should proceed as follows:

1. Determine whether or not VoIP is working.
2. Determine whether or not you can make a voice call.


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3. Verify that SIP-supported codecs are used. Support for codecs varies on different platforms; use the **codec ?** command to determine the codecs available on a specific platform.
4. Isolate and reproduce the failure.
5. Collect relevant information from **show** and **debug** commands, configuration files, and protocol analyzers.
6. Identify the first indication of failure in protocol traces or internal **debug** command output.
7. Look for the cause in configuration files.

**Note** General troubleshooting of problems affecting basic functionality such as dial peers, digit translation, and IP connectivity is beyond the scope of this chapter. For links to additional troubleshooting help, see the "Additional References" section on page 7.

# Using show Commands

To verify SIP gateway status and configuration, perform the following steps as appropriate (commands are listed in alphabetical order).

## SUMMARY STEPS

1. **show sip service**
2. **show sip-ua register status**
3. **show sip-ua statistics**
4. **show sip-ua status**
5. **show sip-ua timers**

## DETAILED STEPS

**Step 1** **show sip service**

Use this command to display the status of SIP call service on a SIP gateway.

The following sample output shows that SIP call service is enabled:

```
Router# show sip service

SIP Service is up
```

The following sample output shows that SIP call service was shut down with the **shutdown** command:

```
Router# show sip service

SIP service is shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop** command:

```
Router# show sip service

SIP service is shut
under 'voice service voip', 'sip' submode
```

The following sample output shows that SIP call service was shut down with the **shutdown forced** command:

```
Router# show sip service

SIP service is forced shut globally
under 'voice service voip'
```

The following sample output shows that SIP call service was shut down with the **call service stop forced** command:

```
Router# show sip service

SIP service is forced shut
under 'voice service voip', 'sip' submode
```

**Step 2** **show sip-ua register status**

Use this command to display the status of E.164 numbers that a SIP gateway has registered with an external primary SIP registrar.

```
Router# show sip-ua register status

Line peer expires(sec) registered
4001 20001 596          no
4002 20002 596          no
5100 1     596          no
9998 2     596          no
```

**Step 3** **show sip-ua statistics**

Use this command to display response, traffic, and retry SIP statistics, including whether call redirection is disabled.

The following sample shows that four registers were sent:

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
    Informational:
      Trying 0/0, Ringing 0/0,
      Forwarded 0/0, Queued 0/0,
      SessionProgress 0/0
     Success:
      OkInvite 0/0, OkBye 0/0,
      OkCancel 0/0, OkOptions 0/0,
      OkPrack 0/0, OkPreconditionMet 0/0,
      OkSubscribe 0/0, OkNOTIFY 0/0,
      OkInfo 0/0, 202Accepted 0/0
      OkRegister 12/49
     Redirection (Inbound only except for MovedTemp(Inbound/Outbound)) :
      MultipleChoice 0, MovedPermanently 0,
      MovedTemporarily 0/0, UseProxy 0,
      AlternateService 0
      Client Error:
      BadRequest 0/0, Unauthorized 0/0,
      PaymentRequired 0/0, Forbidden 0/0,
      NotFound 0/0, MethodNotAllowed 0/0,
      NotAcceptable 0/0, ProxyAuthReqd 0/0,
      ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
      ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
      UnsupportedMediaType 0/0, BadExtension 0/0,
      TempNotAvailable 0/0, CallLegNonExistent 0/0,
      LoopDetected 0/0, TooManyHops 0/0,
```

```
      AddrIncomplete 0/0, Ambiguous 0/0,
      BusyHere 0/0, RequestCancel 0/0,
      NotAcceptableMedia 0/0, BadEvent 0/0,
      SETooSmall 0/0
     Server Error:
      InternalError 0/0, NotImplemented 0/0,
      BadGateway 0/0, ServiceUnavail 0/0,
      GatewayTimeout 0/0, BadSipVer 0/0,
      PreCondFailure 0/0
     Global Failure:
      BusyEverywhere 0/0, Decline 0/0,
      NotExistAnywhere 0/0, NotAcceptable 0/0
      Miscellaneous counters:
      RedirectRspMappedToClientErr 0

SIP Total Traffic Statistics (Inbound/Outbound)
      Invite 0/0, Ack 0/0, Bye 0/0,
      Cancel 0/0, Options 0/0,
      Prack 0/0, Comet 0/0,
      Subscribe 0/0, NOTIFY 0/0,
      Refer 0/0, Info 0/0
      Register 49/16

Retry Statistics
      Invite 0, Bye 0, Cancel 0, Response 0,
      Prack 0, Comet 0, Reliable1xx 0, NOTIFY 0
      Register 4

SDP application statistics:
Parses: 0, Builds 0
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0
Last time SIP Statistics were cleared: <never>
```

The following sample output shows the RedirectResponseMappedToClientError status message. An incremented number indicates that 3*xx* responses are to be treated as 4*xx* responses. When call redirection is enabled (default), the RedirectResponseMappedToClientError status message is not incremented.

```
Router# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
    Informational:
      Trying 0/0, Ringing 0/0,
      Forwarded 0/0, Queued 0/0,
      SessionProgress 0/0
    Success:
      OkInvite 0/0, OkBye 0/0,
      OkCancel 0/0, OkOptions 0/0,
      OkPrack 0/0, OkPreconditionMet 0/0,
      OKSubscribe 0/0, OkNotify 0/0,
      202Accepted 0/0
    Redirection (Inbound only):
      MultipleChoice 0, MovedPermanently 0,
      MovedTemporarily 0, UseProxy 0,
      AlternateService 0
    Client Error:
      BadRequest 0/0, Unauthorized 0/0,
      PaymentRequired 0/0, Forbidden 0/0,
      NotFound 0/0, MethodNotAllowed 0/0,
      NotAcceptable 0/0, ProxyAuthReqd 0/0,
      ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
      ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
```

```
      UnsupportedMediaType 0/0, BadExtension 0/0,
      TempNotAvailable 0/0, CallLegNonExistent 0/0,
      LoopDetected 0/0, TooManyHops 0/0,
      AddrIncomplete 0/0, Ambiguous 0/0,
      BusyHere 0/0, RequestCancel 0/0
      NotAcceptableMedia 0/0, BadEvent 0/0
    Server Error:
      InternalError 0/0, NotImplemented 0/0,
      BadGateway 0/0, ServiceUnavail 0/0,
      GatewayTimeout 0/0, BadSipVer 0/0,
      PreCondFailure 0/0
    Global Failure:
      BusyEverywhere 0/0, Decline 0/0,
      NotExistAnywhere 0/0, NotAcceptable 0/0
    Miscellaneous counters:
      RedirectResponseMappedToClientError 1,
SIP Total Traffic Statistics (Inbound/Outbound)
    Invite 0/0, Ack 0/0, Bye 0/0,
    Cancel 0/0, Options 0/0,
    Prack 0/0, Comet 0/0,
    Subscribe 0/0, Notify 0/0,
    Refer 0/0

Retry Statistics
    Invite 0, Bye 0, Cancel 0, Response 0,
    Prack 0, Comet 0, Reliable1xx 0, Notify 0

SDP application statistics:
 Parses: 0,  Builds 0
 Invalid token order: 0,  Invalid param: 0
 Not SDP desc: 0,  No resource: 0
```

**Step 4** **show sip-ua status**

Use this command to display status for the SIP user agent (UA), including whether call redirection 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)
Redirection (3xx) message handling: ENABLED
```

**Step 5** **show sip-ua timers**

Use this command to display the current settings for the SIP user-agent (UA) timers.

The following sample output shows the waiting time before a register request is sent—that is, the value that is set with the **timers register** command:

```
Router# show sip-ua timers

SIP UA Timer Values (millisecs)
trying 500, expires 180000, connect 500, disconnect 500
comet 500, prack 500, rel1xx 500, notify 500
refer 500, register 500
```

# Using debug Commands

**Note** Commands are listed in alphabetical order.

- Use the **debug aaa authentication** command to display high-level diagnostics related to AAA logins.
- Use the **debug asnl events** command to verify that the SIP subscription server is up. The output displays a pending message if, for example, the client is unsuccessful in communicating with the server.
- Use the **debug call fallback** family of commands to display details of VoIP call fallback.
- Use the **debug cch323** family of commands to provide debugging output for various components within an H.323 subsystem.
- Use the **debug ccsip** family of commands for general SIP debugging, including viewing direction-attribute settings and port and network address-translation traces. Use any of the following related commands:
  - **debug ccsip all—**Enables all SIP-related debugging
  - **debug ccsip calls**—Enables tracing of all SIP service-provider interface (SPI) calls
  - **debug ccsip error**—Enables tracing of SIP SPI errors
  - **debug ccsip events**—Enables tracing of all SIP SPI events
  - **debug ccsip info**—Enables tracing of general SIP SPI information, including verification that call redirection is disabled
  - **debug ccsip media**—Enables tracing of SIP media streams
  - **debug ccsip messages**—Enables all SIP SPI message tracing, such as those that are exchanged between the SIP user-agent client (UAC) and the access server
  - **debug ccsip preauth**—Enables diagnostic reporting of authentication, authorization, and accounting (AAA) preauthentication for SIP calls
  - **debug ccsip states**—Enables tracing of all SIP SPI state tracing
  - **debug ccsip transport**—Enables tracing of the SIP transport handler and the TCP or User Datagram Protocol (UDP) process
- Use the **debug isdn q931** command to display information about call setup and teardown of ISDN network connections (layer 3) between the local router (user side) and the network.
- Use the **debug kpm**l command to enable debug tracing of KPML parser and builder errors.
- Use the **debug radius** command to enable debug tracing of RADIUS attributes.
- Use the **debug rpms-proc preauth** command to enable debug tracing on the RPMS process for H.323 calls, SIP calls, or both H.323 and SIP calls.
- Use the **debug rtr trace** command to trace the execution of an SAA operation.
- Use the **debug voip** family of commands, including the following:
  - **debug voip ccapi protoheaders—**Displays messages sent between the originating and terminating gateways. If no headers are being received by the terminating gateway, verify that the **header-passing** command is enabled on the originating gateway.
  - **debug voip ivr script—**Displays any errors that might occur when the Tcl script is run

– **debug voip rtp session named-event 101—**Displays information important to DTMF-relay debugging, if you are using codec types g726r16 or g726r24. Be sure to append the argument *101* to the command to prevent the console screen from flooding with messages and all calls from failing.

Sample output for some of these commands follows:


- Sample Output for the debug ccsip events Command, page 7
- Sample Output for the debug ccsip info Command, page 7


### Sample Output for the debug ccsip events Command

- The example shows how the Proxy-Authorization header is broken down into a decoded username and password.

```
Router# debug ccsip events

CCSIP SPI: SIP Call Events tracing is enabled

21:03:21: sippmh_parse_proxy_auth: Challenge is 'Basic'.
21:03:21: sippmh_parse_proxy_auth: Base64 user-pass string is
'MTIzNDU2Nzg5MDEyMzQ1Njou'.
21:03:21: sip_process_proxy_auth: Decoded user-pass string is '1234567890123456:.'.
21:03:21: sip_process_proxy_auth: Username is '1234567890123456'.
21:03:21: sip_process_proxy_auth: Pass is '.'.
21:03:21: sipSPIAddBillingInfoToCcb: sipCallId for billing records =
10872472-173611CC-81E9C73D-F836C2B6@172.18.192.19421:03:21: ****Adding to UAS Request
table
```

### Sample Output for the debug ccsip info Command

This example shows only the portion of the debug output that shows that call redirection is disabled. When call redirection is enabled (default), there are no debug line changes.

```
Router# debug ccsip info

00:20:32: HandleUdpSocketReads :Msg enqueued for SPI with IPaddr: 172.18.207.10
:5060
00:20:32: CCSIP-SPI-CONTROL:  act_sentinvite_new_message
00:20:32: CCSIP-SPI-CONTROL:  sipSPICheckResponse
00:20:32: sip_stats_status_code
00:20:32: ccsip_get_code_class: !!Call Redirection feature is disabled on the GW
00:20:32: ccsip_map_call_redirect_responses: !!Mapping 302 response to 480
00:20:32:  Roundtrip delay 4 milliseconds for method INVITE
```

# Additional References

- "SIP Features Roadmap" on page 1—Describes how to access Cisco Feature Navigator; also lists and describes, by Cisco IOS release, SIP features for that release.
- *Cisco IOS Debug Command Reference,* Release 12.3T at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123tcr/123dbr/index.htm
- *Cisco IOS Voice Troubleshooting and Monitoring Guide* at http://www.cisco.com/univercd/cc/td/doc/product/software/ios123/123cgcr/vvfax_c/voipt_c/
- *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2 at http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/fvvfax_c/index.htm
- Cisco Technical Support at http://www.cisco.com/en/US/support/index.html

- *Troubleshooting and Debugging VoIP Call Basics* at http://www.cisco.com/warp/public/788/voip/voip_debugcalls.html
- *VoIP Debug Commands* at http://www.cisco.com/univercd/cc/td/doc/product/access/acs_mod/1700/1750/1750voip/debug.htm

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