



SIP Session Timer Support

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The SIP Session Timer Support feature adds the capability to periodically refresh Session Initiation Protocol (SIP) sessions by sending repeated INVITE requests. The repeated INVITE requests, or re-INVITES, are sent during an active call leg to allow user agents (UAs) or proxies to determine the status of a SIP session. Without this keepalive mechanism, proxies that remember incoming and outgoing requests (stateful proxies) may continue to retain the call state needlessly. If a UA fails to send a BYE message at the end of a session or if the BYE message is lost because of network problems, a stateful proxy does not know that the session has ended. The re-INVITES ensure that active sessions stay active and completed sessions are terminated.

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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 Session Timer Support

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 Session Timer Support

To configure the Session Timer feature, you should understand the following concepts:

Interoperability and Compatibility

- Interoperability--This feature provides a periodic refresh of SIP sessions. The periodic refresh allows user agents and proxies to monitor the status of a SIP session, preventing hung network resources from pausing indefinitely when network failures occur.
- Compatibility--Only one of the two user agent or proxy participants in a call needs to implement the SIP Session Timer Support feature. This feature is easily compatible with older SIP networks. The SIP Session Timer Support feature also adds two new general headers that are used to negotiate the value of the refresh interval.

Role of the User Agents

The initial INVITE request establishes the duration of the session and may include a Session-Expires header and a Min-SE header. These headers indicate the session timer value required by the user agent client (UAC). A receiving user agent server (UAS) or proxy can lower the session timer value, but not lower than the value of the Min-SE header. If the session timer duration is lower than the configured minimum, the proxy or UAS can also send out a 422 response message. If the UAS or proxy finds that the session timer value is acceptable, it copies the Session-Expires header into the 2xx class response.

A UAS or proxy can insert a Session-Expires header in the INVITE if the UAC did not include one. Thus a UAC can receive a Session-Expires header in a response even if none was present in the request.

In the 2xx response, the *refresher* parameter in the Session-Expires header indicates who performs the re-INVITES. For example, if the parameter contains the value *UAC*, the UAC performs the refreshes. For compatibility issues, only one of the two user agents needs to support the session timer feature, and in that case, the UA that supports the feature performs the refreshes. The other UA interprets the refreshes as repetitive INVITES and ignores them.

Re-INVITES are processed identically to INVITE requests, but go out in predetermined session intervals. Re-INVITES carry the new session expiration time. The UA responsible for generating re-INVITE requests sends a re-INVITE out before the session expires. If there is no response, the UA sends a BYE request to terminate the call before session expiration. If a re-INVITE is not sent before the session expiration, either the UAC or the UAS can send a BYE.

If the 2xx response does not contain a Session-Expires header, there is no session expiration and re-INVITES do not need to be sent.

Session-Expires Header

The Session-Expires header conveys the session interval for a SIP call. It is placed in an INVITE request and is allowed in any 2xx class response to an INVITE. Its presence indicates that the UAC wants to use the session timer for this call. Unlike the SIP-Expires header, it can contain only a delta-time, which is the current time, plus the session interval from the response.

For example, if a UAS generates a 200 OK response to a re-INVITE that contained a Session-Expires header with a value of 1800 seconds (30 minutes), the UAS computes the session expiration as 30 minutes

after the time when the 200 OK response was sent. For each proxy, the session expiration is 30 minutes after the time when the 2xx was received or sent. For the UAC, the expiration time is 30 minutes after the receipt of the final response.

The recommended value for the Session-Expires header is 1800 seconds.

The syntax of the Session-Expires header is:

```
Session-Expires = ("Session-Expires" |
"x"
) ":" delta-seconds
                [refresher]
refresher       = ";" "refresher" "=" "UAS" | "UAC"
```

The *refresher* parameter is optional in the initial INVITE, although the UAC can set it to *UAC* to indicate that it will do the refreshes. The 200 OK response must have the refresher parameter set.

Min-SE Header

Because of the processing load of INVITE requests you can configure a minimum timer value that the proxy, UAC, and UAS can accept. The proxy, UAC, and UAS. The **min-se** command sets the minimum timer, and it is conveyed in the Min-SE header in the initial INVITE request.

When making a call, the presence of the Min-SE header informs the UAS and any proxies of the minimum value that the UAC accepts for the session timer duration, in seconds. The default value is 1800 seconds (30 minutes). By not reducing the session timer below the value set, the UAS and proxies prevent the UAC from having to reject a call with a 422 error. Once set, the **min-se** command value affects all calls originated by the router. If the Min-SE header is not present, the UA accepts any value.

The syntax of the Min-SE header is:

```
Min-SE = "Min-SE" ":" delta-seconds
```

422 Response Message

If the value of the Session-Expires header is too small, the UAS or proxy rejects the call with a 422 *Session Timer Too Small* response message. With the 422 response message, the proxy or UAS includes a Min-SE header indicating the minimum session value it can accept. The UAC may then retry the call with a larger session timer value.

If a 422 response message is received after an INVITE request, the UAC can retry the INVITE.

Supported and Require Headers

The presence of the *timer* argument in the Supported header indicates that the UA supports the SIP session timer. The presence of the *timer* argument in the Require header indicates that the opposite UA must support the SIP session timer for the call to be successful.

How to Configure SIP Session Timer Support

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Prerequisites

- Ensure that the gateway has voice functionality that is configurable for SIP.
- Establish a working IP network.
- Configure VoIP--Information about configuring VoIP in a SIP environment can be found here: http://www.cisco.com/en/US/tech/tk652/tk701/tech_configuration_guides_list.html .

Restrictions

- Cisco SIP gateways cannot initiate the use of SIP session timers, but do fully support session timers if another UA requests it.
- The Min-SE value can be set only by using the **min-se** command in the configuration gateway. It cannot be set using the CISCO-SIP-UA-MIB.

Configuring SIP Session Timer Support

To configure the SIP: Session Timer Support feature, complete this task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **min-se** *seconds*
6. **min-se** *exit*
7. **min-se** *show sip-ua min-se*

DETAILED STEPS

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

Command or Action	Purpose
<p>Step 3 <code>voice service voip</code></p> <p>Example:</p> <pre>Router(config)# voice service voip</pre>	Enters voice service VoIP configuration mode.
<p>Step 4 <code>sip</code></p> <p>Example:</p> <pre>Router(conf-voi-serv)# sip</pre>	Enters SIP configuration mode.
<p>Step 5 <code>min-se seconds</code></p> <p>Example:</p> <pre>Router(conf-serv-sip)# min-se 600</pre>	Sets the minimum session expires header value, in seconds, for all calls. <ul style="list-style-type: none"> • Range is 90 to 86,400 (one day). The default value is 1800 (30 minutes).
<p>Step 6 <code>min-se exit</code></p> <p>Example:</p> <pre>Router(conf-serv-sip)# exit</pre>	Exits the current configuration mode.
<p>Step 7 <code>min-se show sip-ua min-se</code></p> <p>Example:</p> <pre>Router(config)# show sip-ua min-se</pre>	Verifies the value of the Min-SE header.

Example

This example contains partial output from the **show running-config** command. It shows that the Min-SE value has been changed from its default value.

```
!
voice service voip
  sip
    min-se 950
!
```

Troubleshooting Tips

To troubleshoot this feature, perform the following steps:

- 1 Make sure that you can make a voice call.
- 2 Use the **debug ccsip all** command to enable all SIP debugging capabilities, or use one of the following SIP **debug** commands:

- 3 **debug ccsip calls**
- 4 **debug ccsip error**
- 5 **debug ccsip events**
- 6 **debug ccsip messages**
- 7 **debug ccsip states**

Feature Information for SIP Session Timer Support

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. An account on Cisco.com is not required. ISR feature history entry.

Table 1 Feature Information for SIP - Session Timer Support

Feature Name	Releases	Feature Information
SIP - Session Timer Support	12.2(8)YN 12.2(15)T 12.2(11)YV 12.2(11)T 12.3(2)T	<p>The SIP Session Timer Support feature adds the capability to periodically refresh Session Initiation Protocol (SIP) sessions by sending repeated INVITE requests. The repeated INVITE requests, or re-INVITEs, are sent during an active call leg to allow user agents (UAs) or proxies to determine the status of a SIP session.</p> <p>The following commands were introduced or modified: min-se (SIP) and show sip-ua min-se.</p>

ASR feature history entry.

Table 2 **Feature Information for SIP - Session Timer Support**

Feature Name	Releases	Feature Information
SIP - Session Timer Support	Cisco XE Release 2.5	<p>The SIP Session Timer Support feature adds the capability to periodically refresh Session Initiation Protocol (SIP) sessions by sending repeated INVITE requests. The repeated INVITE requests, or re-INVITEs, are sent during an active call leg to allow user agents (UAs) or proxies to determine the status of a SIP session.</p> <p>The following commands were introduced or modified: min-se (SIP) and show sip-ua min-se.</p>

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