CHAPTER 6

SIP System Features

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This chapter describes features that apply to all SIP system operations. It includes the following topics:

Differentiated Services Codepoint

The SIP differentiated services codepoint (DSCP) feature enables you to configure the system such that SIP signaling traffic is sent at a desired priority over IP. This is important because SIP messages travel over the same network as the voice traffic. If this network is congested, the voice data might delay the SIP signaling packets, increasing call setup time. Raising the SIP packet priority in relation to other traffic reduces the delay.

Note We recommend using the default values. for the DSCP parameters. These values should be changed only after careful consideration, or if there is a specific need.

Caution If you change any parameters in the ca-config table, these changes do not take effect until the CA platform switches over or restarts.

The applicable SIP DSCP parameters are described in Appendix F, "Data Values for TOS, DSCP, and PHB Parameters," in the *Cisco BTS 10200 Softswitch Call Processing Command Line Interface Reference*.

Limitations on Number of URLs, Parameters, and Headers (Release 5.0, Maintenance Release 1 and Later)

The system imposes limits on the decoding of incoming SIP messages. These limits are applicable to both subscriber-related and trunk-related incoming SIP messages. These limitations are intended to protect the system from decoding extremely large messages, which in turn could overload the system and cause performance problems.

Note These limits are not provisionable. If you need to change any of these limits, contact your Cisco account team.

Table 6-1 lists the limits related to URL and ReqUri.

Table 6-1 Limits on URL and ReqUri

Description	Limit
Maximum number of URLs (SIP+Tel+Unknown) in a SIP message	25
Maximum number of parameters in the ReqUri of a message	10
Maximum number of header parameters (parameters occurring after "?" character) in the Request-URI of a message	5
Maximum number of parameters in a SIP URL	10
Maximum number of header parameters (parameters occurring after "?" character) in a SIP URL	5
Maximum number of parameters in a Tel URL	5

Table 6-2 lists the maximum number of parameters allowed in each SIP message header.

Table 6-2 Maximum Number of Parameters Allowed in SIP Message Headers

Header Name	Maximum Number of Parameters Allowed in Header
Contact	10
Via	10
Route	5
Record-Route	5
Diversion	10
Call-Info	5
Alert-Info	5
Error-Info	5
P-Asserted-Identity	5
Accept-Contact	5
To	5
From	5

Table 6-2 Maximum Number of Parameters Allowed in SIP Message Headers (continued)

Header Name	Maximum Number of Parameters Allowed in Header
Referred-By	5
Refer-To	5

Table 6-3 lists the maximum number of unknown Option tags of a specified kind in a SIP message.

Table 6-3 Maximum Number of Unknown Option Tags in SIP Message

Message	Maximum Number of Unknown Option Tags Allowed
Supported	5
Unsupported	5
Require	5

Table 6-4 lists the maximum number of parameters allowed in each SIP message header.

Table 6-4 Maximum Number of Parameters Allowed in SIP Message Headers

Header Name	Parameter Type	Maximum Number of Parameters Allowed in Header
Replaces	All parameters	5
Event	All parameters	5
Reason	All parameters	5
Accept	All parameters	5
Session-Expires	All parameters	5
Min-SE	All parameters	5
Warnings	All parameters	5
Accept-Language	Number of languages	5
Accept-Language	Language parameters	5
Accept-Encoding	All parameters	5
Authorization	All parameters	15
Retry-After	All parameters	5

Table 6-5 lists the maximum number of headers allowed in a SIP message.

Table 6-5 Maximum Number of Headers Allowed in a SIP Message

Header Name	Maximum Number of Headers Allowed
Contact	5
Via	5
Route	5
Record-Route	5

Table 6-5 Maximum Number of Headers Allowed in a SIP Message (continued)

Header Name	Maximum Number of Headers Allowed
Diversion	5
Call-Info	5
Alert-Info	5
Error-Info	5
P-Asserted-Identity	5
Contact	5
To	1
From	1
Call-ID	1
CSeq	1
Session-Expires	1
Min-SE	1
Referred-By	1
Refer-To	1
Replaces	1
Allow-Events	5
Event	1
Reason	5
Accept	5
Accept-Encoding	5
Authorization	1
Retry-After	1

Limitation On Transient Calls During Switchover

If the active Call Agent experiences a problem and switches over to the standby side, stable calls are preserved. However, calls that are in a transient state (call setup is not complete) might be dropped or improperly set up. During a Call Agent switchover, the BTS 10200 cannot complete call setup for these transient calls. You can provision the BTS 10200 to set an Expires header for INVITEs sent on outbound calls. This provisioning is done through the SIA-DEFAULT-INVITE-EXPIRES-SECONDS token in the CA-CONFIG table. (The system default behavior is to omit the Expires header.)

In addition, transient calls and inactive connected calls originated on the BTS 10200 are cleaned up through a periodic audit mechanism that runs once per hour. The frequency of this audit can be modified. However, changing this requires careful consideration to avoid adverse effects on call processing. Contact Cisco TAC if you have identified a need to change this frequency.

Message Handling Based On Content-Length Header

This section describes the handling of SIP messages based on the Content-Length header.

For outbound TCP and UDP messages, the BTS 10200 complies with RFC 3261 by including a Content-Length header with the correct value for the body of the request.

For inbound UDP messages, the BTS 10200 complies with RFC 3261 by assuming the length in the Content-Length header is correct and discarding additional bytes (if any) in the content body. If the actual content length is shorter than the length indicated in the header, the BTS 10200 reads the content and attempts to complete the call with the content that was received. This handling of shortened content is not compliant with RFC 3261 (which requires messages with shortened content to be discarded with a 400 Bad Request response), but it is intended as a more tolerant treatment for inbound messages. Regardless of the content length, the BTS 10200 attempts to complete calls based on the inbound message. However, if the content itself is invalid, the BTS 10200 rejects the call.

For inbound TCP messages, the BTS 10200 requires the received length to be correct, because the TCP message contains is a continuous stream of bytes rather than discrete packets. This treatment is compliant with RFC 3261.

SIA Process Restart

If a SIA process fails, the BTS 10200 can restart it. After a SIA process restart, new calls can be set up, and calls that were established (answered) prior to the SIA process failure continue to be handled. However, any transactions that were pending at the time of a SIA process failure are not processed after the SIA restart.

A SIA process restart can occur a maximum of three times every 30 minutes. If a restart happens more than three times in 30 minutes, a failover from an active CA to a standby CA occurs.

SIP Timer Values

This section describes the SIP timers supported by the BTS 10200.

Tip The provisioning information for SIP timers is provided in the "SIP Timer Values for SIP Subscribers" section on page 2-17 (for SIP subscriber lines) and the "SIP Timer Values for SIP Trunks" section on page 4-14 (for SIP trunks).

Tip For more information about these timers, or for common SIP term definitions from this section, see RFC 3261.

Rules for Configuring the SIP Timers

Use the following rules to configure the SIP timers in the BTS 10200. The rules are necessary due to mutual dependency between the timers. If any rules fail, the system computes the values of the timers.

```
TIMER-T2-SECS * 1000 > TIMER-T1-MILLI
TIMER-T2-SECS * 1000 > TIMER-G-MILLI
```

```
TIMER-B-SECS * 1000 > TIMER-A-MILLI
TIMER-F-SECS * 1000 > TIMER-E-MILLI
TIMER-D-SECS > 32
```

In addition to these rules, the timer values must be in the range of values specified in the "Detailed Description of Timers" section on page 6-6.

Detailed Description of Timers

The following list describers the timer parameters in the sip-timer-profile table.

- TIMER-T1-MILLI (range 100–5000, default=500, in milliseconds)—T1 is an estimate of the round-trip time (RTT). The system uses this timer to calculate the default values of the transaction timers A through H and J in the following list. Many of those timers scale with T1; therefore, changing the T1 value changes the default values for timers A through H and J. The calculation is shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-T2-SECS (range 1–10, default=4, in seconds)—The maximum allowed interval for non-INVITE requests. It is also used as the maximum retransmit interval for SIP INVITE responses.
- TIMER-T4-SECS (range 1–10, default=5, in seconds)—The timer represents the maximum amount of time the network takes to clear messages between client and server transactions. The system uses this timer to calculate the default value of the transaction timer TIMER-H-SECS; therefore, changing the T4 value changes the default value for TIMER-H-SECS. The calculation is shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-A-MILLI (range 100–5000, default=0, in milliseconds)—The UAC timer for INVITE request retransmit interval. For example, if the value is 500 ms, the INVITE request retransmissions occ2 seconds). Applicable to UDP only. If TIMER-A-MILLI is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-B-SECS (range 1–3600, default=0, in seconds)—The UAC INVITE transaction timer limits the INVITE transaction timeout. For SIP TCP trunk connections, there are certain scenarios in which the BTS 10200 does not immediately detect a loss of connection to an IP address endpoint after transmitting an INVITE request. As a result, we recommend provisioning this timer to 6 seconds when you are configuring TCP trunks, so that advancing to the FQDN's next IP address occurs in a timely manner. If TIMER-B-SECS is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-D-SECS (range 33–65, default=33, in seconds, set to 0 for TCP)—The UAC timer used for the wait time of response retransmissions. For INVITE, because an ACK could be lost, the UAS must wait at least 32 seconds (assuming the default transaction timer on the other end is 32 seconds) to receive any retransmissions of responses from the UAS and send an ACK. In a Cisco BTS 10200 implementation, this transaction clearing timer is applicable only for INVITE requests. For non-INVITE messages, the transaction is cleared immediately upon receipt of final response. If TIMER-D-SECS is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-E-MILLI (range 100–5000, default=0, in milliseconds)—The UAC timer for a non-INVITE request retransmit interval. For example, if the value is 500 ms, the non-INVITE request retransmissions occur at intervals of 500 ms, 1s, 2s, 4s, 4s, 4s, 4s, 4s, 4s, and 4s (assuming

TIMER-F-SECS defined below is 32 seconds and TIMER-T2-SECS defined above is 4 seconds). This parameter is applicable to UDP only. If TIMER-E-MILLI is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.

- TIMER-F-SECS (range 1–3600, default=0, in seconds)—The UAC non-INVITE transaction timer that limits the number of retransmissions for non-INVITE requests. If TIMER-F-SECS is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-G-MILLI (range 100–5000, default=0, in milliseconds)—Specifies the INVITE response retransmit interval. The UAS timer implemented to achieve reliability of successful final responses to INVITE requests. It starts when you are using a reliable transport protocol such as TCP. Even though the transport protocol might be reliable up to the next hop, it is not guaranteed reliable end-to-end if there are several proxy servers along the path when the call is set up. This timer is started when a a final response is sent for an INVITE request. The timer stops when a matching ACK is received for the final response sent. For example, if a 200 OK is sent for INVITE, the UAS must receive the matching ACK for the 200 OK. If the TIMER-G-MILLI is 500 ms, the final response to the INVITE from the UAS retransmits at intervals of 500 ms, 1s, 2s, 4s, 8s, 16s, 32s (assuming that TIMER-H-SECS is 32 seconds). If TIMER-G-MILLI is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-H-SECS (range 1–3600, default=0, in seconds)—The UAS timer responsible for clearing an incomplete INVITE UAS transaction. It also controls the number of INVITE final response retransmissions sent to UAC. The timer is started upon sending a final response for the INVITE request. It is the total wait time for ACK receipt from UAC. If TIMER-H-SECS is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9.
- TIMER-I-SECS (range 1–10, default=0, in seconds)—This UAS timer is the wait time for ACK retransmits. It frees the server transaction resources and starts when the first ACK to the final response is received for INVITE requests. Upon receipt of an ACK for certain INVITE final responses (401, 415, 420, 422, 423, 480 and 484), the value of timer I is set to a fixed duration of 32 seconds. The responses result in resubmission of the original INVITE with modifications, and prevent the resources from prematurely freeing. A 481 (Call-Leg/Transaction does not exist) or a 408 (Request Timeout) response sent for the INVITE results in a much smaller fixed duration of 4 seconds for timer I. This ensures that CCB resources are promptly freed when the call is not set up, allowing reuse for other calls. For ACK to all other INVITE final responses, which are not typically followed by a re-attempt, the timer duration for this timer is set at TIMER-I-SECS.

  When a BYE is subsequently sent or received on a call in progress, and timer I is running for that call, it is canceled and restarted for a smaller fixed duration of 4 seconds to reduce CCB hold time after call completion, and to optimize CCB resource usage.

  If TIMER-I-SECS is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9

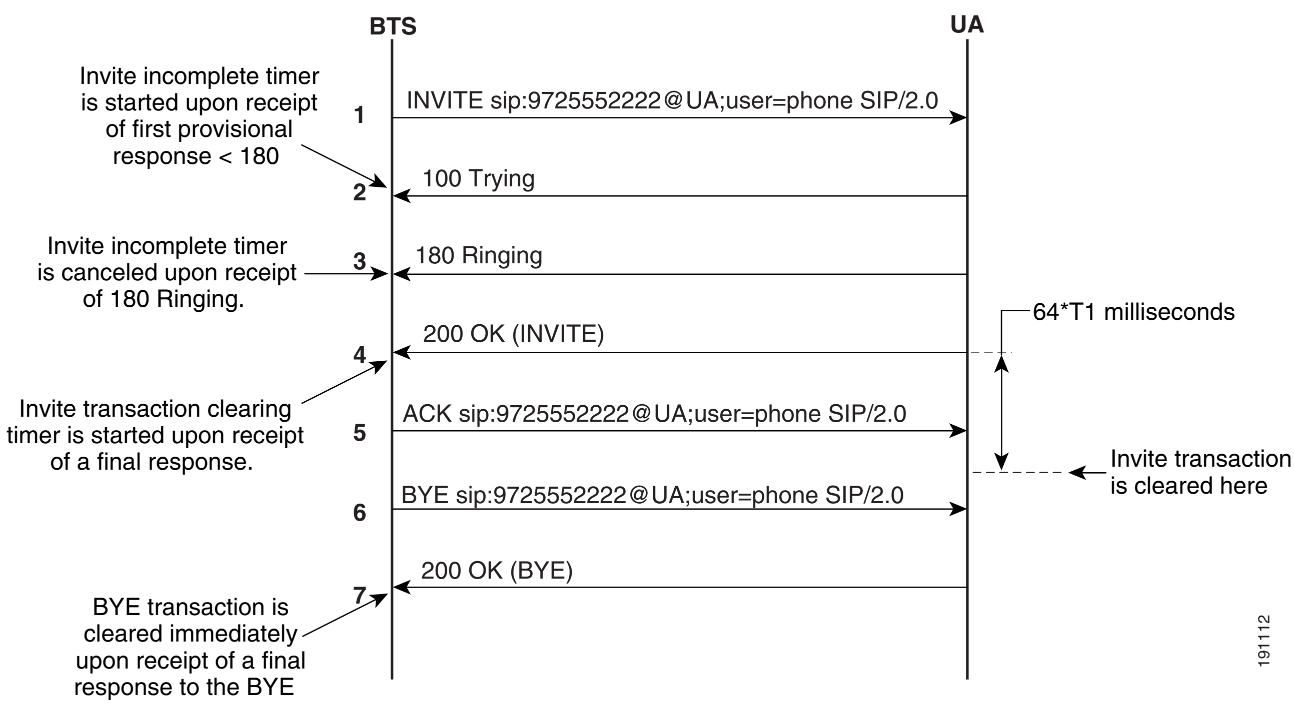
  TIMER-J-SECS (range 1–3600, default=0, in seconds, set to 0 for TCP)—This UAS timer cleans up non-INVITE UAS transactions. A shorter nonconfigurable timer of four seconds is used for BYE and CANCEL. Additionally, when a BYE or CANCEL is sent or received on a call in progress, if timer J is running for any non-INVITE transaction associated with that call, it is canceled and restarted for a smaller fixed duration of 4 seconds to reduce CCB hold time after call completion, and to optimize CCB resource usage. If TIMER-J-SECS is set to the default value of 0, the system automatically calculates a value for it, as shown in the "Computation of Default Timer Values A Through J from Timers T1 and T4" section on page 6-9

- INVITE-INCOMPLETE-TIMER-SECS (range 15–600, default=40, in seconds)—This UAC timer cleans up UAC INVITE transactions for which a provisional response less than 180 was received, but no ringing or final response was received within a reasonable period of time. This timer starts upon receipt of the first provisional response (>=100 and <180) for the INVITE message sent. Upon receipt of the final response or 18x response to INVITE request, this timer is canceled.

  This timer is also started if a CANCEL is sent, to clean up the INVITE transaction in case of a final response (487), indicating that the request was canceled, is not received.

  The process involving receipt of the 180 response is shown in Figure 6-1.

Figure 6-1 Invite Incomplete Timer Process with 180 Response



- MIN-SE (range 100–1800, default=900, in seconds)—This is a session timer. It specifies the minimum session-expires allowed on the Cisco BTS 10200. Any INVITE request received with a session-expires lower than the MIN-SE is rejected with a 422 response that has a header Min-SE = MIN-SE.
- SESSION-EXPIRES-DELTA-SECS (range 100–7200, default=1800, in seconds)—This is a session timer. It cleans up resources in case of an abnormal session end. The Cisco BTS 10200 sends the SESSION-EXPIRES-DELTA-SECS as the session-expires header in the initial INVITE. When a session is established, a session timer is started based on the negotiated value (it can be lower or equal to the SESSION-EXPIRES-DELTA-SECS). If the BTS 10200 is determined as the refresher, it starts a session timer for duration of half the negotiated time. A re-INVITE or update is sent out upon timer expiry to refresh the session. If the remote end is determined as the refresher, then a session timer is started for duration of (negotiated session-expires – 10 seconds). In this case, a BYE is sent to end the session if a session refresh (re-INVITE or update) is not received before the session timer expires.

Note When the SESSION-EXPIRES-DELTA-SECS timer expires, the BTS 10200 might send a Re-Invite (as opposed to an Update) with the previously sent SDP. If the BTS 10200 receives a 200 OK with the SDP changed from the previously received SDP, the BTS 10200 does not send this changed SDP to the origination.

Computation of Default Timer Values A Through J from Timers T1 and T4

If the following timer values are not explicitly provisioned, the system computes them automatically, based on the values of TIMER-T1-MILLI and TIMER-T4-SECS, as follows:

```
TIMER-A-MILLI = TIMER-T1-MILLI
TIMER-B-SECS = (64 * TIMER-T1-MILLI) / 1000
TIMER-E-MILLI = TIMER-T1-MILLI
TIMER-F-SECS = (64 * TIMER-T1-MILLI) / 1000
TIMER-G-MILLI = TIMER-T1-MILLI
TIMER-H-SECS = (64 * TIMER-T1-MILLI) / 1000
TIMER-I-SECS = TIMER-T4-SECS
TIMER-J-SECS = (64 * TIMER-T1-MILLI) / 1000
```

Calculation of Timer Retransmission Count

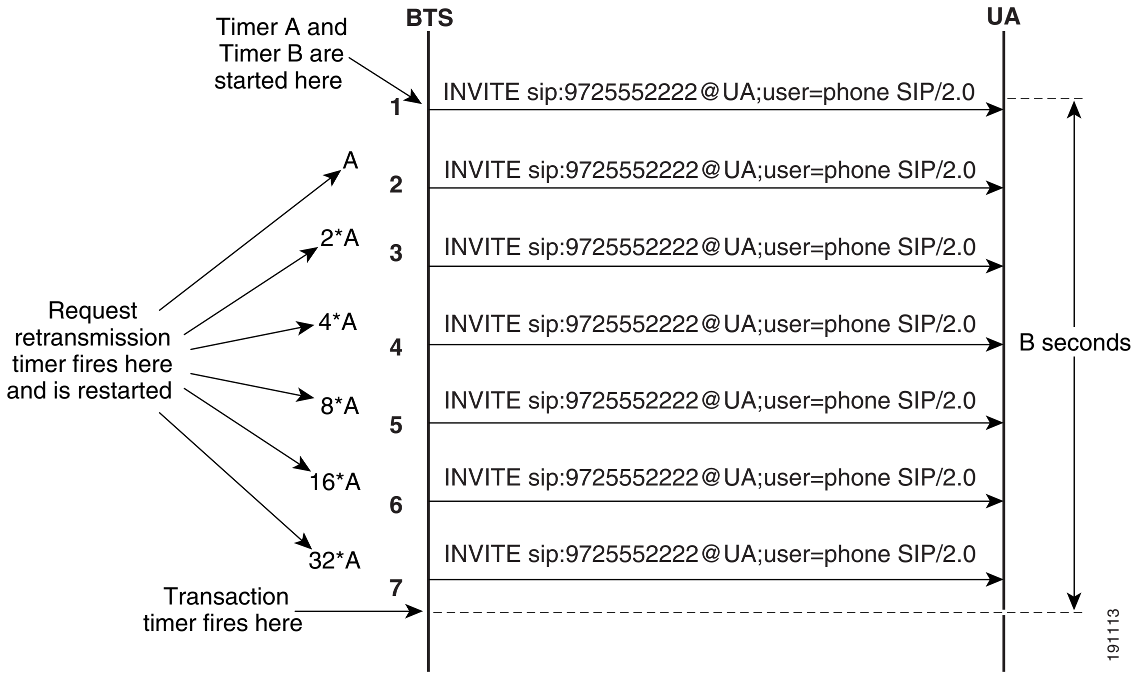
The retransmit count is defined as the number of times the same request or response is retransmitted after the message is sent once to the transport layer. The BTS 10200 computes this retransmit count based on RFC 3261 recommendations.

INVITE Retransmit Count

The invite retransmission process is shown in Figure 6-2. If there is no response for the initial INVITE request, then INIVTE requests are retransmitted as shown.

For example, if TIMER-A-MILLI is 500 ms and TIMER-B-SECS is 32 seconds, then there are six retransmissions after the first request, for a total of seven requests from the UAC. The retransmissions occur at intervals of 500 ms, 1s, 2s, 4s, 8s, 16s, and 32s.

Figure 6-2 Invite Retransmissions with No Response



Non-INVITE Retransmit Count

If there is no response for the initial non-INVITE request, INIVTE requests are retransmitted as shown.

For example, if TIMER-E-MILLI is 500 ms, TIMER-T2-SECS is 4 seconds and TIMER-F-SECS is 32 seconds, then non-INVITE retransmissions occur at intervals of 500 ms, 1s, 2s, 4s, 4s, 4s, 4s, 4s, 4s, 4s. This means that retransmissions occur with an exponentially increasing interval that caps at T2. In this particular scenario, there are 10 retransmissions which is a total of 11 requests from UAC.

Response Retransmit Count

If no ACK is received for the final response of the INVITE request, the responses are retransmitted. This process is shown in Figure 6-3.

Figure 6-3 Invite Server Transaction Timer Cancelled Upon Receipt of ACK