Cisco BTS 10200 Softswitch SIP Traffic Measurement Enhancements Feature Module

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This feature module describes the Session Initiation Protocol (SIP) Traffic Measurement Enhancements feature for Release 6.0 of the Cisco BTS 10200 Softswitch.

Understanding the SIP Traffic Measurement Enhancements Feature

SIP call failures are due to network failure or end user conditions. The SIP Traffic Measurement Enhancements feature helps the customer to identify the call failures due to end user conditions such as:

- If the caller abandons the call before the receiver of the call answers (called party answers).
- If the receiver of the call is busy or not responding to the incoming call.

These call traffic statistics serve as input for further network planning and expansions.

This feature introduces the following counters for call failures due to end user conditions:

- Call abandoned—Call abandoned by caller before called party answers (abandoned call is one where caller releases the call before called party answers).
- User busy—Called party busy.
- No answer— Called party is not responding to an incoming call.

Counters are maintained at the system level (both SIP endpoints and SIP trunks) and at each SIP trunk group level. The counters are maintained separately for originating and terminating calls. Call Processing (CallP), SIP stack, and SIP Adapter update the Traffic Measurement (TMM) counters at run-time for SIP traffic.

Note Prior to Release 6.0 these call failure counters were not captured as a part of the summary report.

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Summary Report Changes

This section provides information on the counters included in Release 6.0. The Call Abandon, User Busy, and No Answers counters for call failures due to end user conditions are captured in summary reports. The summary report changes can be viewed at two levels:

- System Level
- Trunk Group Level

System Level

Users can obtain the call processing and SIP statistics by means of the following commands:

- measurement-callp-summary
- measurement-sia-summary

The **measurement-callp-summary** command provides the summary reports of call processing statistics for system-wide traffic that are captured for a specified call agent during that collection interval (time-interval). Use the following command to query the counters:

```
# show measurement_callp_summary or report measurement_callp_summary
```

The **measurement-sia-summary** command provides the summary reports of SIP (both SIP endpoints and SIP trunks) and the SIP interface adapter statistics that are captured for a specified call agent during a collection interval (time-interval). Each collection interval starts on the hour, half-hour, or quarter-hour.

```
# show measurement_sia_summary or report measurement_sia_summary
```

Note The SIP stack counters that capture the statistics related to ingress and egress of 3xx, 4xx, 5xx, and 6xx SIP responses do not increment call abandoned, user busy, and no answers counters on retransmissions (both reception and transmission).

Trunk Group Level

Use the following command to query the counters at trunk group level. This command provides the trunk group usage information.

```
# show measurement_tg_usage_summary or report measurement_tg_usage_summary
```

Trunk Group Usage Counters

This section lists the new and changed trunk group usage counters.

New Trunk Group Usage Counters

Table 1 lists the new trunk group usage counters. These counters are specific to SIP trunks. Other types of trunks are not supported.

Table 1 New Trunk Group Usage Traffic Measurement Counters with Descriptions

Counter	Description
TRKGRP_SIP_3xx_RX	Number of 3xx class (REDIRECTION) messages the reporting call agent received on a SIP trunk group.
TRKGRP_SIP_3xx_TX	Number of 3xx class (REDIRECTION) messages the reporting call agent transmitted on a SIP trunk group.
TRKGRP_SIP_4xx_RX	Number of 4xx class (REQUEST FAILURES) messages the reporting call agent received on a SIP trunk group.
TRKGRP_SIP_4xx_TX	Number of 4xx class (REQUEST FAILURES) messages the reporting call agent transmitted on a SIP trunk group.
TRKGRP_SIP_5xx_RX	Number of 5xx class (SERVER FAILURES) messages the reporting call agent received on a SIP trunk group.
TRKGRP_SIP_5xx_TX	Number of 5xx class (SERVER FAILURES) messages the reporting call agent transmitted on a SIP trunk group.
TRKGRP_SIP_6xx_RX	Number of 6xx class (GLOBAL FAILURES) messages the reporting call agent received on a SIP trunk group.
TRKGRP_SIP_6xx_TX	Number of 6xx class (GLOBAL FAILURES) messages the reporting call agent transmitted on a SIP trunk group.
TRKGRP_INBOUND_FAIL	Number of failed inbound calls on a SIP trunk. However, calls that fail for Abandon, User busy, or No answer scenarios are not pegged.
TRKGRP_INBOUND_SUCC	Number of established inbound calls on a SIP trunk.
TRKGRP_INCOM_CALL_ABDN	Number of incoming abandoned SIP trunk calls at the reporting call agent.
TRKGRP_INCOM_CALL_NOT_ANS	Number of incoming SIP trunk calls not answered by a called party.
TRKGRP_INCOM_END_USR_BUSY	Number of incoming SIP trunk calls not completed because the called party was busy.
TRKGRP_OUTBOUND_SUCC	Number of established outbound calls on a SIP trunk.
TRKGRP_OUTG_CALL_ABDN	Number of outgoing abandoned SIP trunk calls at the reporting call agent.
TRKGRP_OUTG_CALL_NOT_ANS	Number of outgoing SIP trunk calls not answered by a called party.
TRKGRP_OUTG_END_USR_BUSY	Number of SIP trunk calls not completed because the called party was busy.

Changed Trunk Group Usage Counters

Table 2 lists the changed trunk group usage counters.

Table 2 ***Changed Trunk Group Usage Counters***

Counters	Description
TRKGRP_OUTBOUND_FAIL	This counter is pegged when a call on an outgoing trunk group fails. This counter is not pegged for Abandon, User busy, or No answer scenarios.

Call Processing Counters

This section lists the new and changed call processing counters.

New Call Processing Counters

Table 3 lists the new call processing counters.

Table 3 ***New Call Processing Traffic Measurement Counters with Descriptions***

Counter	Description
CALLP_SIP_ORIG_CALL_ABDN	This counter is pegged when the caller abandons the call before the called party answers the call.
CALLP_SIP_ORIG_CALL_NOT_ANS	This counter is pegged when the incoming call is not answered by the called party.
CALLP_SIP_ORIG_END_USR_BUSY	This counter is pegged when the incoming call is not complete due to the called party being busy.
CALLP_SIP_TERM_CALL_ABDN	This counter is pegged when the outgoing call is abandoned by the originator before the called party answers.
CALLP_SIP_TERM_CALL_NOT_ANS	This counter is pegged when the outgoing call is not answered by the called party.
CALLP_SIP_TERM_END_USR_BUSY	This counter is pegged when the outgoing call is terminated as the called party was busy. **Note** The originating and terminating counters are pegged for both originating and terminating calls if both are using SIP.
CALLP_SIP_ORIG_SUCC	Number of incoming established SIP calls.
CALLP_SIP_TERM_SUCC	Number of outgoing established SIP calls.

Note The New Call Processing counters are pegged when the SIP signalling protocol is used.

Changed Call Processing Counters

Table 4 lists the changed Call Processing counters.

Table 4 Changed Call Processing Counters

Counters	Description
CALLP_SIP_ORIG_FAIL	The number of incoming SIP calls that are not established. This counter does not include the calls in these conditions: • When the call is abandoned by the caller before the called party answers. • The called party is busy. • The called party is not responding to the incoming call.
CALLP_SIP_TERM_FAIL	The number of outgoing SIP calls that are not established. This counter does not include the calls in these conditions: • When the call is abandoned by the caller before the called party answers. • The called party is busy. • The called party is not responding to the incoming call.