Reporting End-of-Call Statistics in SIP BYE Message

The Reporting End-of-Call Statistics in Session Initiation Protocol (SIP) BYE Message feature enables you to send call statistics to a remote end when a call terminates. The call statistics are sent as a new header in the BYE message or in the 200 OK message (response to BYE message). The statistics include Real-time Transport Protocol (RTP) packets sent or received, total bytes sent or received, total number of packets that are lost, delay jitter, round-trip delay, and call duration.

This feature enables Cisco Unified Border Element (Cisco UBE) to use the call statistics to update the call data records in Cisco Unified Communications Manager (Cisco UCM) or Cisco Unified Communications Manager Express (Cisco UCME).

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled by default on Cisco UBE.

A new header P-RTP-Stat is added to the BYE and 200 OK messages. The format of P-RTP-Stat is as follows:

P-RTP-Stat: PS=<Packets Sent>, OS=<Octets Sent>, PR=<Packets Recd>, OR=<Octets Recd>, PL=<Packets Lost>, JI=<Jitter>, LA=<Round Trip Delay in ms>, DU=<Call Duration in seconds>

Table 1describes the P-RTP-Stat header.

Table 1 P-RTP-Stat Header Fields

Field	Description	Range of Values
PS	Packets Sent	0 to 4294967295
OS	Octets Sent	0 to 4294967295
PR	Packets Received	0 to 4294967295
OR	Octets Received	0 to 4294967295
PL	Packets Lost	0 to 4294967295
JI	Jitter	0 to 4294967295
LA	Round Trip Delay, in milliseconds (ms)	-2147483648 to +2147483647
DU	Call Duration, in seconds	0 to 4294967295

Prerequisites for Reporting End-of-Call Statistics in SIP BYE Message

Cisco Unified Border Element

- Cisco IOS Release 15.1(3)T or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.3S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions for Reporting End-of-Call Statistics in SIP BYE Message

- If the **media flow-around** command is configured, the call statistics are not sent for a 200 OK message.

- If the **media flow-around** command is configured, the call statistics are passed through the Cisco UBE for a BYE message.
- The values are not validated when the incoming statistics are passed to the endpoints. Hence, in some cases the values may be invalid.
- The value of round-trip delay is valid only if the remote end supports Real-Time Control Protocol (RTCP).

Disabling Reporting End-of-Call Statistics in SIP BYE Message

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled by default on the Cisco UBE. That is, the P-RTP-Stat header is added to the list of headers that can be processed through the SIP profiles. You must apply SIP profile rules to remove the header from the mandatory header list.

This section contains the following tasks:

- Defining SIP Profile Rules to Remove a Header, page 2 (required)
- Disabling Reporting End-of-Call Statistics in SIP BYE Message at the Global Level, page 3 (optional)
- Disabling Reporting End-of-Call Statistics in SIP BYE Message at the Dial Peer Level, page 4 (optional)

Defining SIP Profile Rules to Remove a Header

Perform this task to define SIP profile rules to remove a header.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice class sip-profiles** *tag*
4. **request bye sip-header p-rtp-stat remove**
5. **response 200 sip-header p-rtp-stat remove**
6. **exit**

DETAILED STEPS

	Command or Action	Purpose

Step 1	`enable`
	
	Example:
	`Router> enable`
		Enables privileged EXEC mode.
		• Enter your password if prompted.

Step 2	`configure terminal`
	
	Example:
	`Router# configure terminal`
		Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice class sip-profiles** *tag* **Example:** `Router(config)# voice class sip-profiles 100`	Configures SIP profiles for a voice class and enters voice class configuration mode.
Step 4	**request bye sip-header p-rtp-stat remove** **Example:** `Router(config-class)# request bye sip-header p-rtp-stat remove`	Removes the P-RTP-Stat SIP header from the BYE message.
Step 5	**response 200 sip-header p-rtp-stat remove** **Example:** `Router(config-class)# response 200 sip-header p-rtp-stat remove`	Removes the P-RTP-Stat SIP header from the 200 OK message.
Step 6	**exit** **Example:** `Router(config-class)# exit`	Exits voice class configuration mode.

Disabling Reporting End-of-Call Statistics in SIP BYE Message at the Global Level

Perform this task to disable the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the global level.

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled by default on Cisco UBE. Hence, to disable the feature, you must modify the SIP profiles to remove the P-RTP-Stat SIP header from the request and the response messages and then configure the modified SIP profile on the Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **sip**
5. **sip-profiles** *tag*
6. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Router(config)# voice service voip`	Specifies VoIP as the voice encapsulation method and enters voice-service configuration mode.
Step 4	**sip** **Example:** `Router(conf-voi-serv)# sip`	Enters service SIP configuration mode.
Step 5	**sip-profiles** *tag* **Example:** `Router(conf-serv-sip)# sip-profiles 100`	Disables the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the global level. • Here, the Cisco UBE is configured to use the modify SIP profiles as defined in "Defining SIP Profile Rules to Remove a Header" section on page 2 to disable the configuration.
Step 6	**exit** **Example:** `Router(config-class)# exit`	Exits service SIP configuration mode.

Disabling Reporting End-of-Call Statistics in SIP BYE Message at the Dial Peer Level

Perform this task to disable the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the dial peer level.

The Support for Reporting End-of-Call Statistics in SIP BYE Message feature is enabled by default. Hence, to disable the feature, you must modify the SIP profiles to remove the P-RTP-Stat SIP header from the request and the response messages and then configure the modified SIP profile on the Cisco UBE.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **dial-peer voice** *tag* **voip**
4. **voice-class sip profiles** *tag*
5. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**dial-peer voice** *tag* **voip** **Example:** `Router(config)# dial-peer voice 100 voip`	Defines a dial peer to specify the method of voice encapsulation and enters dial peer configuration mode.
Step 4	**voice-class sip profiles** *tag* **Example:** `Router(config-dial-peer)# voice-class sip profiles 100`	Disables the Support for Reporting End-of-Call Statistics in SIP BYE Message feature at the dial peer level. • In this example, the Cisco UBE is configured to use the modify SIP profiles as defined in "Defining SIP Profile Rules to Remove a Header" section on page 2 to disable the configuration.
Step 5	**exit** **Example:** `Router(config-dial-peer)# exit`	Exits dial-peer configuration mode.