



Configuring IP SLAs UDP Jitter Operations for VoIP

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This document describes how to configure an IP Service Level Agreements (SLAs) User Datagram Protocol (UDP jitter operation to proactively monitor Voice over IP (VoIP) quality levels in your network, allowing you to guarantee VoIP quality levels to your users in IPv4 or IPv6 networks. The IP SLAs VoIP UDP jitter operation accurately simulates VoIP traffic using common codecs, and calculates consistent voice quality scores (MOS and ICPIF) between Cisco devices in the network.



Note

The term “Voice” in this document should be taken to mean any Internet telephony applications. The term “Voice over IP” can include the transmission of multimedia (both voice and video) over IP networks.

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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 <http://www.cisco.com/go/cfn>. An account on Cisco.com is not required.

Restrictions for IP SLAs UDP Jitter Operations for VoIP

- This feature uses UDP traffic to generate approximate Voice over IP scores. It does not provide support for the Real-Time Transport Protocol (RTP).
- ICPIF and MOS values provided by this feature, while consistent within IP SLAs, are estimates only and are intended only for relative comparisons. The values may not match values determined using other methods.
- Predictions of customer opinion (such as those listed for the E-Model transmission rating factor R and derived Mean Opinion Scores) determined by any method are intended only for transmission planning and analysis purposes and should not be interpreted as reflecting actual customer opinions.

Information About IP SLAs UDP Jitter Operations for VoIP

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The Calculated Planning Impairment Factor (ICPIF)

The ICPIF originated in the 1996 version of ITU-T recommendation G.113, “Transmission impairments,” as part of the formula $Icpif = Itot - A$. ICPIF is actually an acronym for “(Impairment) Calculated Planning Impairment Factor,” but should be taken to simply mean the “calculated planning impairment factor.” The ICPIF attempts to quantify, for comparison and planning purposes, the key impairments to voice quality that are encountered in the network.

The ICPIF is the sum of measured impairment factors (total impairments, or $Itot$) minus a user-defined access Advantage Factor (A) that is intended to represent the user’s expectations, based on how the call was placed (for example, a mobile call versus a land-line call). In its expanded form, the full formula is expressed as:

$$Icpif = Io + Iq + Idte + Idd + Ie - A$$

where

- Io represents impairments caused by non-optimal loudness rating,
- Iq represents impairments caused by PCM quantizing distortion,
- $Idte$ represents impairments caused by talker echo,
- Idd represents impairments caused by one-way transmission times (one-way delay),

- I_e represents impairments caused by equipment effects, such as the type of codec used for the call and packet loss, and
- A represents an access Advantage Factor (also called the user Expectation Factor) that compensates for the fact that users may accept some degradation in quality in return for ease of access.

ICPIF values are expressed in a typical range of 5 (very low impairment) to 55 (very high impairment). ICPIF values numerically less than 20 are generally considered “adequate.” While intended to be an objective measure of voice quality, the ICPIF value is also used to predict the subjective effect of combinations of impairments. The table below, taken from G.113 (02/96), shows how sample ICPIF values are expected to correspond to subjective quality judgement.

Table 1: Quality Levels as a Function of Total Impairment Factor ICPIF

Upper Limit for ICPIF	Speech Communication Quality
5	Very good
10	Good
20	Adequate
30	Limiting case
45	Exceptional limiting case
55	Customers likely to react strongly (complaints, change of network operator)

For further details on the ICPIF, see the 1996 version of the G.113 specification.



Note

The latest version of the ITU-T G.113 Recommendation (2001), no longer includes the ICPIF model. Instead, it refers implementers to G.107: “The Impairment Factor method, used by the E-model of ITU-T G.107, is now recommended. The earlier method that used Quantization Distortion Units is no longer recommended.” The full E-Model (also called the ITU-T Transmission Rating Model), expressed as $R = R_o - I_s - I_d - I_e + A$, provides the potential for more accurate measurements of call quality by refining the definitions of impairment factors (see the 2003 version of the G.107 for details). Though the ICPIF shares terms for impairments with the E-Model, the two models should not be confused. The IP SLAs VoIP UDP Operation feature takes advantage of observed correspondences between the ICPIF, transmission rating factor R , and MOS values, but does not yet support the E-Model.

IP SLAs uses a simplified ICPIF formula, defined in more detail later in this document.

Mean Opinion Scores (MOS)

The quality of transmitted speech is a subjective response of the listener. Each codec used for transmission of Voice over IP provides a certain level of quality. A common benchmark used to determine the quality of sound produced by specific codecs is MOS. With MOS, a wide range of listeners have judged the quality of voice samples sent using particular codecs, on a scale of 1 (poor quality) to 5 (excellent quality). The opinion scores are averaged to provide the mean for each sample. The table below shows MOS ratings and the corresponding description of quality for each value.

Table 2: MOS Ratings

Score	Quality	Description of Quality Impairment
5	Excellent	Imperceptible
4	Good	Just perceptible, but not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying but not objectionable
1	Bad	Very annoying and objectionable

As the MOS ratings for codecs and other transmission impairments are known, an estimated MOS can be computed and displayed based on measured impairments. This estimated value is designated as MOS-CQE (Mean Opinion Score; Conversational Quality, Estimated) by the ITU in order to distinguish it from objective or subjective MOS values (see P.800.1 for details).

Voice Performance Monitoring Using IP SLAs

One of the key metrics in measuring voice and video quality over an IP network is jitter. Jitter is the name used to indicate the variation in delay between arriving packets (inter-packet delay variance). Jitter affects voice quality by causing uneven gaps in the speech pattern of the person talking. Other key performance parameters for voice and video transmission over IP networks include latency (delay) and packet loss. IP SLAs is an embedded active monitoring feature of Cisco software that provides a means for simulating and measuring these parameters in order to ensure your network is meeting or exceeding service-level agreements with your users.

IP SLAs provides a UDP jitter operation, which consists of UDP probe packets sent across the network from an origin device to a specific destination (called the operational target). This synthetic traffic is used to record the amount of jitter for the connection, as well as the round-trip time, per-direction packet loss, and one-way delay time (one-way latency). (The term “synthetic traffic” indicates that the network traffic is simulated; that is, the traffic is generated by IP SLAs.) Data, in the form of collected statistics, can be displayed for multiple tests over a user-defined period of time, allowing you to see, for example, how the network performs at different times of the day, or over the course of a week. The jitter probe has the advantage of utilizing the IP SLAs Responder to provide minimal latency at the receiving end.

The IP SLAs VoIP UDP jitter operation modifies the standard UDP jitter operation by adding the capability to return MOS and ICPIF scores in the data collected by the operation, in addition to the metrics already gathered by the UDP jitter operation. This VoIP-specific implementation provides even more useful information in determining the performance of your VoIP network, thereby improving your ability to perform network assessment, troubleshooting, and health monitoring.

Codec Simulation Within IP SLAs

The IP SLAs VoIP UDP jitter operation computes statistics by sending n UDP packets, each of size s , sent t milliseconds apart, from a given source router to a given target router, at a given frequency f . The target router must be running the IP SLAs Responder in order to process the probe operations.

To generate MOS and ICPIF scores, you specify the codec type used for the connection when configuring the VoIP UDP jitter operation. Based on the type of codec you configure for the operation, the number of packets (n), the size of each payload (s), the inter-packet time interval (t), and the operational frequency (f)

will be auto-configured with default values. (See the table below for specifics.) However, you are given the option, if needed, to manually configure these parameters in the syntax of the `udp-jitter` command.

The table below shows the default parameters that are configured for the operation by codec.

Table 3: Default VoIP UDP Jitter Operation Parameters by Codec

Codec	Default Request Size (Packet Payload) (s)	Default Interval Between Packets (t)	Default Number of Packets (n)	Frequency of Probe Operations (f)
G.711 mu-Law (g711ulaw)	160 + 12 RTP bytes	20 ms	1000	Once every 1 minute
G.711 A-Law (g711alaw)	160 + 12 RTP bytes	20 ms	1000	Once every 1 minute
G.729A (g729a)	20 + 12 RTP bytes	20 ms	1000	Once every 1 minute

For example, if you configure the VoIP UDP jitter operation to use the characteristics for the `g711ulaw` codec, by default a probe operation will be sent once a minute (f). Each probe operation would consist of 1000 packets (n), with each packet containing 180 bytes of synthetic data (s), sent 20 milliseconds apart (t).

The IP SLAs ICPIF Value

ICPIF value computation with Cisco software is based primarily on the two main factors that can impair voice quality: delayed packets and lost packets. Because packet delay and packet loss can be measured by IP SLAs, the full ICPIF formula, $Icpif = Io + Iq + Idte + Idd + Ie - A$, is simplified by assuming the values of Io , Iq , and $Idte$ are zero, resulting in the following formula:

$$\text{Total Impairment Factor (Icpif)} = \text{Delay Impairment Factor (Idd)} + \text{Equipment Impairment Factor (Ie)} - \text{Expectation/Advantage Factor (A)}$$

This means that the ICPIF value is computed by adding a Delay Impairment Factor, which is based on a measurement of delayed packets, and an Equipment Impairment Factor, which is based on a measurement of lost packets. From this sum of the total impairments measured in the network, an impairment variable (the Expectation Factor) is subtracted to yield the ICPIF.

This is the same formula used by Cisco Gateways to calculate the ICPIF for received VoIP data streams.

The Delay Impairment Factor

The Delay Impairment Factor (Idd) is a number based on two values. One value is fixed and is derived using the static values (as defined in the ITU standards) for Codec Delay, Look Ahead Delay, and Digital Signal Processing (DSP) Delay. The second value is variable and is based on the measured one-way delay (round-trip time measurement divided by 2). The one-way delay value is mapped to a number using a mapping table that is based on a G.107 (2002 version) analytic expression. The table below shows sample correspondences between the one-way delay measured by IP SLAs and Delay Impairment Factor values.

Table 4: Sample Correspondence of One-Way Delay to ICPIF Delay Impairment

One-Way Delay (ms)	Delay Impairment Factor
50	1
100	2
150	4
200	7

The Equipment Impairment Factor

The Equipment Impairment Factor (I_e) is a number based on the amount of measured packet loss. The amount of measured packet loss, expressed as a percentage of total number of packets sent, corresponds an Equipment Impairment Factor that is defined by codec. The table below shows sample correspondences between the packet loss measured by IP SLAs and Equipment Impairment Factor values.

Table 5: Sample Correspondence of Measured Packet Loss to ICPIF Equipment Impairment

Packet Loss (as a percentage of total number of packets sent)	Equipment Impairment Value for PCM (G.711) Codecs	Equipment Impairment Value for the CS-ACELP (G.729A) Codec
2%	12	20
4%	22	30
6%	28	38
8%	32	42

The Expectation Factor

The Expectation Factor, also called the Advantage Factor (A), is intended to represent the fact that users may accept some degradation in quality in return for ease of access. For example, a mobile phone user in a hard-to-reach location may have an expectation that the connection quality will not be as good as a traditional land-line connection. This variable is also called the Advantage Factor (short for Access Advantage Factor) because it attempts to balance an increased access advantage against a decline in voice quality.

The table below, adapted from ITU-T Rec. G.113, defines a set of provisional maximum values for A in terms of the service provided.

Table 6: Advantage Factor Recommended Maximum Values

Communication Service	Advantage / Expectation Factor: Maximum value of A
Conventional wire-line (land-line)	0
Mobility (cellular connections) within a building	5

Communication Service	Advantage / Expectation Factor: Maximum value of A
Mobility within a Geographical area or moving in a vehicle	10
Access to hard-to-reach location; (for example, via multi-hop satellite connections)	20

These values are only suggestions. To be meaningful, the use of the factor *A* and its selected value in a specific application should be used consistently in any planning model you adopt. However, the values in the table above should be considered as the absolute upper limits for *A*.

The default Advantage Factor for IP SLAs VoIP UDP jitter operations is always zero.

The IP SLAs MOS Value

IP SLAs uses an observed correspondence between ICPIF and MOS values to estimate an MOS value. Usage of the abbreviation MOS within the context of this feature should be taken to represent the MOS-CQE (Mean Opinion Score; Conversational Quality, Estimated).

The E model, as defined in G.107 (03/2003), predicts the subjective quality that is experienced by an average listener by combining the impairment caused by transmission parameters (such as loss and delay) into a single rating, the transmission rating factor R (the R Factor). This rating, expressed in a scale of 0 (worst) to 100 (best) can be used to predict subjective user reactions, such as the MOS. Specifically, the MOS can be obtained from the R Factor with a converting formula. Conversely, a modified inverted form can be used to calculate R Factors from MOS values.

There is also a relationship between the ICPIF value and the R Factor. IP SLAs takes advantage of this correspondence by deriving the approximate MOS score from an estimated R Factor, which, in turn, is derived from the ICPIF score. The table below shows the resulting MOS values that will be generated for corresponding ICPIF values.

Table 7: Correspondence of ICPIF Values to MOS Values

ICPIF Range	MOS	Quality Category
0 - 3	5	Best
4 - 13	4	High
14 - 23	3	Medium
24 - 33	2	Low
34 - 43	1	Poor

IP SLAs will always express the estimated MOS value as a number in the range of 1 to 5, with 5 being the best quality. A MOS value of 0 (zero) indicates that MOS data could not be generated for the operation.

How to Configure IP SLAs UDP Jitter Operations for VoIP

- [Configuring and Scheduling an IP SLAs VoIP UDP Jitter Operation, page 8](#)

Configuring and Scheduling an IP SLAs VoIP UDP Jitter Operation



Note

- Currently, IP SLAs supports only the following speech codecs (compression methods):
 - G.711 A Law (g711alaw: 64 kbps PCM compression method)
 - G.711 mu Law (g711ulaw: 64 kbps PCM compression method)
 - G.729A (g729a: 8 kbps CS-ACELP compression method)
- The following commands, available in UDP jitter configuration mode, are not valid for UDP jitter (codec) operations:
 - **history distributions-of-statistics-kept**
 - **history statistics-distribution-interval**
 - **request-data-size**
- Specifying the codec-type will configure the appropriate default values for the **codec-interval**, **codec-size**, and **codec-numpacket** options. You should not specify values for the interval, size, and number of packet options unless you have a specific reason to override the defaults (for example, approximating a different codec).
- The **show ip sla configuration** command will list the values for the “Number of statistic distribution buckets kept” and “Statistic distribution interval (milliseconds),” but these values do not apply to jitter (codec) operations.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **ip sla** *operation-number*
4. **udp-jitter** {*destination-ip-address* | *destination-hostname*} *destination-port* **codec** *codec-type* [**codec-
numpackets** *number-of-packets*] [**codec-size** *number-of-bytes*] [**codec-interval** *milliseconds*]
[**advantage-factor** *value*] [**source-ip** {*ip-address* | *hostname*}] [**source-port** *port-number*] [**control**
{**enable** | **disable**}]
5. **history enhanced** [**interval** *seconds*] [**buckets** *number-of-buckets*]
6. **frequency** *seconds*
7. **history hours-of-statistics-kept** *hours*
8. **owner** *owner-id*
9. **tag** *text*
10. **threshold** *milliseconds*
11. **timeout** *milliseconds*
12. Do one of the following:
 - **tos** *number*
 - **traffic-class** *number*
13. **flow-label** *number*
14. **verify-data**
15. **vrf** *vrf-name*
16. **exit**
17. **ip sla schedule** *operation-number* [**life** {**forever** | *seconds*}] [**start-time** {*hh:mm:ss*} [*month day* | *day*
month] | **pending** | **now** | **after** *hh:mm:ss*] [**ageout** *seconds*] [**recurring**]
18. **exit**
19. **show ip sla configuration** [*operation-number*]

DETAILED STEPS

	Command or Action	Purpose
Step 1	<p>enable</p> <p>Example:</p> <pre>Router> enable</pre>	<p>Enables privileged EXEC mode.</p> <ul style="list-style-type: none"> • Enter your password if prompted.
Step 2	<p>configure terminal</p> <p>Example:</p> <pre>Router# configure terminal</pre>	<p>Enters global configuration mode.</p>
Step 3	<p>ip sla <i>operation-number</i></p>	<p>Begins configuration for an IP SLAs operation and enters IP SLA configuration mode.</p>

	Command or Action	Purpose
	<p>Example:</p> <pre>Router(config)# ip sla 10</pre>	
Step 4	<p>udp-jitter {<i>destination-ip-address</i> <i>destination-hostname</i>} <i>destination-port</i> codec <i>codec-type</i> [codec-numpackets <i>number-of-packets</i>] [codec-size <i>number-of-bytes</i>] [codec-interval <i>milliseconds</i>] [advantage-factor <i>value</i>] [source-ip {<i>ip-address</i> <i>hostname</i>}] [source-port <i>port-number</i>] [control {enable disable}]</p> <p>Example:</p> <pre>Router(config-ip-sla)# udp-jitter 209.165.200.225 16384 codec g711alaw advantage-factor 10</pre>	Configures the operation as a jitter (codec) operation that will generate VoIP scores in addition to latency, jitter, and packet loss statistics.
Step 5	<p>history enhanced [interval <i>seconds</i>] [buckets <i>number-of-buckets</i>]</p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# history enhanced interval 900 buckets 100</pre>	(Optional) Enables enhanced history gathering for an IP SLAs operation.
Step 6	<p>frequency <i>seconds</i></p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# frequency 30</pre>	(Optional) Sets the rate at which a specified IP SLAs operation repeats.
Step 7	<p>history hours-of-statistics-kept <i>hours</i></p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# history hours-of- statistics-kept 4</pre>	(Optional) Sets the number of hours for which statistics are maintained for an IP SLAs operation.
Step 8	<p>owner <i>owner-id</i></p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# owner admin</pre>	(Optional) Configures the Simple Network Management Protocol (SNMP) owner of an IP SLAs operation.
Step 9	<p>tag <i>text</i></p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# tag TelnetPollServer1</pre>	(Optional) Creates a user-specified identifier for an IP SLAs operation.
Step 10	<p>threshold <i>milliseconds</i></p>	(Optional) Sets the upper threshold value for calculating network monitoring statistics created by an IP SLAs operation.

	Command or Action	Purpose
	<p>Example:</p> <pre>Router(config-ip-sla-jitter)# threshold 10000</pre>	
Step 11	<p>timeout <i>milliseconds</i></p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# timeout 10000</pre>	(Optional) Sets the amount of time an IP SLAs operation waits for a response from its request packet.
Step 12	<p>Do one of the following:</p> <ul style="list-style-type: none"> • tos <i>number</i> • traffic-class <i>number</i> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# tos 160</pre> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# traffic-class 160</pre>	<p>(Optional) In an IPv4 network only, defines the ToS byte in the IPv4 header of an IP SLAs operation.</p> <p>or</p> <p>(Optional) In an IPv6 network only, defines the traffic class byte in the IPv6 header for a supported IP SLAs operation.</p>
Step 13	<p>flow-label <i>number</i></p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# flow-label 112233</pre>	(Optional) In an IPv6 network only, defines the flow label field in the IPv6 header for a supported IP SLAs operation.
Step 14	<p>verify-data</p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# verify-data</pre>	(Optional) Causes an IP SLAs operation to check each reply packet for data corruption.
Step 15	<p>vrf <i>vrf-name</i></p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# vrf vpn-A</pre>	(Optional) Allows monitoring within Multiprotocol Label Switching (MPLS) Virtual Private Networks (VPNs) using IP SLAs operations.
Step 16	<p>exit</p> <p>Example:</p> <pre>Router(config-ip-sla-jitter)# exit</pre>	Exits UDP jitter configuration submode and returns to global configuration mode.
Step 17	<p>ip sla schedule <i>operation-number</i> [life {forever <i>seconds</i>}] [start-time {<i>hh:mm[:ss]</i> [<i>month day</i> <i>day month</i>] pending now after <i>hh:mm:ss</i>}] [ageout <i>seconds</i>] [recurring]</p>	Configures the scheduling parameters for an individual IP SLAs operation.

	Command or Action	Purpose
	<p>Example:</p> <pre>Router(config)# ip sla schedule 5 start-time now life forever</pre>	
Step 18	<p>exit</p> <p>Example:</p> <pre>Router(config)# exit</pre>	(Optional) Exits global configuration mode and returns to privileged EXEC mode.
Step 19	<p>show ip sla configuration [<i>operation-number</i>]</p> <p>Example:</p> <pre>Router# show ip sla configuration 10</pre>	(Optional) Displays configuration values including all defaults for all IP SLAs operations or a specified operation.

- [Troubleshooting Tips, page 12](#)
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Troubleshooting Tips

- If the IP SLAs operation is not running and generating statistics, add the **verify-data** command to the configuration of the operation (while configuring in IP SLA configuration mode) to enable data verification. When enabled, each operation response is checked for corruption. Use the **verify-data** command with caution during normal operations because it generates unnecessary overhead.
- Use the **debugipsla trace** and **debug ip sla error** commands to help troubleshoot issues with an IP SLAs operation.

What to Do Next

To add proactive threshold conditions and reactive triggering for generating traps, or for starting another operation, to an IP SLAs operation, see the "Configuring Proactive Threshold Monitoring" section.

To view and interpret the results of an IP SLAs operation use the **show ip sla statistics** command. Checking the output for fields that correspond to criteria in your service level agreement will help you determine whether the service metrics are acceptable.

Configuration Examples for IP SLAs UDP Jitter Operations for VoIP

- [Example IP SLAs VoIP UDP Operation Configuration, page 13](#)
- [Example IP SLAs VoIP UDP Operation Statistics Output, page 14](#)

Example IP SLAs VoIP UDP Operation Configuration

The following example assumes that the IP SLAs Responder is enabled on the device at 209.165.200.225.

```

Router> enable

Password:
Router# configure terminal

Enter configuration commands, one per line. End with the end command.
Router(config)# ip sla 10
Router(config-sla)# udp-jitter 209.165.200.225 16384 codec g711alaw advantage-factor 2

Router(config-sla-jitter)# owner admin_bofh
Router(config-sla-jitter)# exit

Router(config)# ip sla schedule 10 start-time now

Router(config)# exit

Router#
Router# show running-config | begin ip sla 10

ip sla 10
  udp-jitter 209.165.200.225 16384 codec g711alaw advantage-factor 2
  owner admin_bofh
ip sla schedule 10 start-time now
.
.
.
Router# show ip sla configuration 10

Entry number: 10
Owner: admin_bofh
Tag:
Type of operation to perform: jitter
Target address: 209.165.200.225
Source address: 0.0.0.0
Target port: 16384
Source port: 0
Operation timeout (milliseconds): 5000
Codec Type: g711alaw
Codec Number Of Packets: 1000
Codec Packet Size: 172
Codec Interval (milliseconds): 20
Advantage Factor: 2
Type Of Service parameters: 0x0
Verify data: No
Vrf Name:
Control Packets: enabled
Operation frequency (seconds): 60
Next Scheduled Start Time: Start Time already passed
Life (seconds): 3600
Entry Ageout (seconds): never
Status of entry (SNMP RowStatus): Active
Connection loss reaction enabled: No
Timeout reaction enabled: No
Verify error enabled: No
Threshold reaction type: Never
Threshold (milliseconds): 5000
Threshold Falling (milliseconds): 3000
Threshold Count: 5
Threshold Count2: 5
Reaction Type: None
Number of statistic hours kept: 2
Number of statistic distribution buckets kept: 1
Statistic distribution interval (milliseconds): 20
Enhanced History:
When a codec type is configured for a jitter operation, the standard jitter "Request size (ARR data
portion)," "Number of packets," and "Interval (milliseconds)" parameters will not be displayed in the show

```

ip sla configuration command output. Instead, values for “Codec Packet Size,” “Codec Number of Packets,” and “Codec Interval (milliseconds)” are displayed.

Example IP SLAs VoIP UDP Operation Statistics Output

Use the **show ip sla statistics** command to display Voice scores (ICPIF and MOS values) for the jitter (codec) operation.

```
Router# show ip sla statistics 10

Entry number: 10
Modification time: 12:57:45.690 UTC Sun Oct 26 2003
Number of operations attempted: 1
Number of operations skipped: 0
Current seconds left in Life: Forever
Operational state of entry: Active
Last time this entry was reset: Never
Connection loss occurred: FALSE
Timeout occurred: FALSE
Over thresholds occurred: FALSE
Latest RTT (milliseconds): 19
Latest operation start time: 12:57:45.723 Sun Oct 26 2003
Latest operation return code: OK
!
Voice Scores:
ICPIF: 20          MOS Score: 3.20
!
RTT Values:
NumOfRTT: 10      RTTAvg: 19      RTTMin: 19      RTTMax: 20
RTTSum: 191      RTTSum2: 3649
Packet Loss Values:
PacketLossSD: 0 PacketLossDS: 0
PacketOutOfSequence: 0 PacketMIA: 0 PacketLateArrival: 0
InternalError: 0      Busies: 0
Jitter Values:
NumOfJitterSamples: 9
MinOfPositivesSD: 0      MaxOfPositivesSD: 0
NumOfPositivesSD: 0      SumOfPositivesSD: 0      Sum2PositivesSD: 0
MinOfNegativesSD: 0      MaxOfNegativesSD: 0
NumOfNegativesSD: 0      SumOfNegativesSD: 0      Sum2NegativesSD: 0
MinOfPositivesDS: 1      MaxOfPositivesDS: 1
NumOfPositivesDS: 1      SumOfPositivesDS: 1      Sum2PositivesDS: 1
MinOfNegativesDS: 1      MaxOfNegativesDS: 1
NumOfNegativesDS: 1      SumOfNegativesDS: 1      Sum2NegativesDS: 1
Interarrival jitterout: 0      Interarrival jitterin: 0
One Way Values:
NumOfOW: 0
OWMinSD: 0      OWMaxSD: 0      OWSumSD: 0      OWSum2SD: 0
OWMinDS: 0      OWMaxDS: 0      OWSumDS: 0      OWSum2DS: 0
```

Additional References

Related Documents

Related Topic	Document Title
Cisco IOS commands	Cisco IOS Master Commands List, All Releases
Cisco IOS IP SLAs commands	Cisco IOS IP SLAs Command Reference

Related Topic	Document Title
Voice over IP (VoIP) codecs	Understanding Codecs: Complexity, Hardware Support, MOS, and Negotiation http://www.cisco.com/en/US/tech/tk1077/technologies_tech_note09186a00800b6710.shtml
Jitter in Packet Voice Networks	Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms) http://www.cisco.com/en/US/tech/tk652/tk698/technologies_tech_note09186a00800945df.shtml
PSTN Fallback for Voice Gateways	SIP: Measurement-Based Call Admission Control for SIP http://www.cisco.com/en/US/docs/ios/12_2t/12_2t15/feature/guide/ftcacsip.html
Standards	
Standard ¹	Title
ITU-T Recommendation G.107 (2003)	The E-model, a computation model for use in transmission planning
ITU-T Recommendation G.113 (1996)	<i>Transmission impairments</i>
ITU-T Recommendation G.113 (2001)	Transmission impairments due to speech processing
ITU-T Recommendation G.711 (1998)	<i>Pulse code modulation (PCM) of voice frequencies</i> (also known as the G.711 Voice Codec)
ITU-T Recommendation G.729 Annex A (1996)	<i>Reduced complexity 8 kbit/s CS-ACELP speech codec</i> (also known as the G.729/A/B Speech Codec)
ITU-T Recommendation P.800.1 (2003)	Mean Opinion Score (MOS) terminology
MIBs	
MIB	MIB Link
CISCO-RTTMON-MIB	To locate and download MIBs for selected platforms, Cisco IOS releases, and feature sets, use Cisco MIB Locator found at the following URL: http://www.cisco.com/go/mibs

¹ Full support by this feature for listed RFCs is not claimed. ITU Telecommunication Standards (“ITU-T Recommendations In Force”) can be obtained from <http://www.itu.ch>. Summary definitions are available from a variety of internet sources.

RFCs

RFC²	Title
RFC 768	<i>User Datagram Protocol</i>
RFC 1889	<i>RTP: A Transport Protocol for Real-Time Applications</i>

Technical Assistance

Description	Link
The Cisco Support and Documentation website provides online resources to download documentation, software, and tools. Use these resources to install and configure the software and to troubleshoot and resolve technical issues with Cisco products and technologies. Access to most tools on the Cisco Support and Documentation website requires a Cisco.com user ID and password.	http://www.cisco.com/cisco/web/support/index.html

Feature Information for IP SLAs UDP Jitter Operations for VoIP

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 <http://www.cisco.com/go/cfn>. An account on Cisco.com is not required.

Table 8: Feature Information for IP SLAs UDP Jitter Operations for VoIP

Feature Name	Releases	Feature Information
IP SLAs UDP Jitter Operation	12.2(31)SB2 12.2(33)SRB1 12.2(33)SXH 12.3(14)T 15.0(1)S	The Cisco IOS IP SLAs User Datagram Protocol (UDP) jitter operation allows you to measure round-trip delay, one-way delay, one-way jitter, one-way packet loss, and connectivity in networks that carry UDP traffic.

² Full support by this feature for listed RFCs is not claimed.

Feature Name	Releases	Feature Information
IPv6 - IP SLAs (UDP Jitter, UDP Echo, ICMP Echo, TCP Connect)	12.2(33)SRC 12.2(33)SB 12.4(20)T	Support was added for operability in IPv6 networks.

Glossary

codec --In the context of IP Telephony, a codec is a compression and decompression algorithm used to transfer voice and video data more efficiently. Voice codec types are typically referred to using the ITU recommendation number that defines the algorithm (for example, “G.711” instead of “PCM”).

CS-ACELP --The codec type defined in the reference documents G.729 and G.729A, *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)* .

ITU --The International Telecommunication Union. The ITU is an international organization within the United Nations System where governments and the private sector coordinate global telecom networks and services. The ITU Telecommunication Standardization Sector (ITU-T), responsible for defining standards (Recommendations) covering all fields of telecommunications, is one of the three operational sectors of the ITU. The ITU web site is at <http://www.itu.int>.

ITU-T --ITU Telecommunication Standardization Sector. The ITU-T is one of the three operational sectors of the ITU, and is responsible for defining standards (called ITU-T Recommendations) covering all fields of telecommunications.

MOS-CQE (Mean Opinion Score; Conversational Quality, Estimated)--The score calculated by a network planning model which aims at predicting the quality in a conversational application situation. Estimates of conversational quality carried out according to ITU-T Rec. G.107, when transformed to a mean opinion score (MOS), give results in terms of MOS-CQE.³

PCM --The codec type defined in the reference document G.711, *Pulse code modulation (PCM) of voice frequencies* .

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³ Definition from ITU-T Recommendation P.800.1. Used in accordance with the ITU Copyright and Disclaimer Notice.