Service Assurance Agent (SAA) VoIP UDP Operation

The Service Assurance Agent (SAA) Voice over IP (VoIP) User Datagram Protocol (UDP) Operation feature adds the capability to return approximate Mean Opinion Score (MOS) and Calculated Planning Impairment Factor (ICPIF) voice quality scores in the data collected by the SAA Jitter operation.

Feature History for SAA VoIP UDP Operation Feature

Release	Modification
12.3(4)T	This feature was introduced.

Finding Support Information for Platforms and Cisco IOS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS software image support. Access Cisco Feature Navigator at http://www.cisco.com/go/fn. You must have an account on Cisco.com. If you do not have an account or have forgotten your username or password, click **Cancel** at the login dialog box and follow the instructions that appear.

Contents

CISCO SYSTEMS

Corporate Headquarters:
Cisco Systems, Inc., 170 West Tasman Drive, San Jose, CA 95134-1706 USA

Prerequisites for SAA VoIP UDP Operations

To use this feature, your networking devices on both ends of the connection must support SAA. SAA is an integrated feature of all Cisco IOS software-based platforms.

Restrictions for SAA VoIP UDP Operations

This feature uses UDP traffic to generate approximate Voice over IP scores. It does not provide support for the Real-Time Transport Protocol (RTP).

Note The term "Voice" in this document should be taken to mean any Internet telephony applications. For example, the term "Voice over IP" is commonly understood to include the transmission of multimedia (both voice and video) over IP networks.

ICPIF and MOS values provided by this feature are estimates only, and they are intended only for relative comparisons. The values may not match values determined using other methods.

Note 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 purposes and should not be interpreted as reflecting actual customer opinions.

This release 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)

Information About SAA VoIP UDP Operations

To use the SAA VoIP UDP Operation feature, you should understand the following concepts:

- The Calculated Planning Impairment Factor (ICPIF), page 2
- Mean Opinion Scores (MOS), page 4
- Voice Performance Monitoring Using SAA, page 4

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 initialism 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),
- *Ie* 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. Table 1, 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 $R = Ro - Is - Id - Ie + A$, 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 SAA 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.

SAA 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. Table 2 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 SAA

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. SAA is an embedded active monitoring feature of Cisco IOS 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.

SAA provides a Jitter monitoring 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[1] 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). 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 SAA (RTR) Responder to provide minimal latency at the receiving end.

1. The term “synthetic traffic” indicates that the network traffic is simulated; that is, the traffic is generated by SAA.

The SAA VoIP UDP Operation feature adds the capability to return approximate MOS and ICPIF scores in the data collected by the SAA Jitter operation, in addition to the metrics already gathered by the Jitter operation. This SAA feature 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 SAA

The SAA 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 SAA 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 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 Table 3 for specifics.) However, you are given the option, if needed, to manually configure these parameters in the syntax of the **type jitter** command.

Table 3 shows the default parameters that are configured for the operation by codec.

Table 3 Default 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 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 SAA ICPIF Value

ICPIF value computation with Cisco IOS 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 SAA, 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:

Total Impairment Factor (Icpif) = Delay Impairment Factor (Idd) + Equipment Impairment Factor (Ie) – 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 the measured one way delay. In this context, one way delay is a combination of the One Way Transmission Delay, as measured by the SAA operation, combined with the static values (as defined in the ITU standards) for the Codec Delay, the Look Ahead Delay, and the Digital Signal Processing (DSP) Delay. Table 4 shows sample correspondences between the measured One Way Transmission Delay and Delay Impairment Factor values.

Table 4 Sample Correspondence of Measured Delay to ICPIF Delay Impairment

One Way Delay (in milliseconds)	Delay Impairment Factor
150 or less	0
200	3
250	10
300	15
350	20
400	25
500	30
600	35
800 or greater	40

As shown in Table 4, Delay Impairment calculations are determined linearly for One Way Delay measurements between 150 and 800 milliseconds.

The Equipment Impairment Factor

The Equipment Impairment Factor (*Ie*) 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. Table 5 shows sample correspondences between the packet loss measured by SAA 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
0%	0	10
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 Dactor (short for Access Advantage Factor) because it attempts to balance an increased access advantage against a decline in voice quality.

Table 6, 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
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 Table 6 should be considered as the absolute upper limits for *A*.

The default Advantage Factor for SAA Jitter operations is always zero.

The SAA MOS Value

SAA 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. SAA 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. Table 7 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

SAA will always express the estimated MOS value as a whole 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 SAA VoIP UDP Operations

To return VoIP scores with SAA Jitter operation statistics, perform the following task.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **rtr** *operation-number*
4. **type jitter**
5. **dest-ipaddr** (optional)
6. **dest-port** (optional)
7. **frequency** (optional)
8. **enhanced-history** (optional)
9. **owner** (optional)
10. **tag** (optional)
11. **vrf** (optional)
12. **hours-of-statistics kept** (optional)
13. **threshold** (optional)
14. **timeout** (optional)
15. **tos** (optional)
16. **verify-data** (optional)
17. **exit**
18. **rtr schedule**

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	**rtr** *operation-number* **Example:** `Router(config)# rtr 10`	Specifies an identification number for the SAA operation to be configured, and enters SAA RTR configuration mode.
Step 4	**type jitter dest-ipaddr** {*hostname* | *ip-address*} **dest-port** *port-number* **codec** *codec-type* [**codec-numpackets** *number-of-packets*] [**codec-size** *number-of-bytes*] [**codec-interval** *milliseconds*] [**advantage-factor** *value*] [**source-ipaddr** {*hostname* | *ip-address*}] [**source-port** *port-number*] [**control** {**enable** | **disable**}] **Example:** `Router(config-rtr)#` **`type jitter dest-ipaddr 209.165.200.225 dest-port 16384 codec g711alaw advantage-factor 10`**	Configures the operation as a Jitter (codec) operation that will generate VoIP scores in addition to latency, jitter, and packet loss statistics. • For the *codec-type* argument, use one of the following keywords: – **g711alaw**—64 kbps PCM compression method – **g711ulaw**—64 kbps PCM compression method – **g729a**—8 kbps CS-ACELP compression method • 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 value you specify for the **advantage-factor** will be subtracted from the measured impairment values. You can use this option to correct the ICPIF and MOS values for network conditions. The default advantage factor (expectation factor) is 0. • When configuring a Jitter operation that uses a codec type, the **dest-port** should be an even numbered port in the range 16384 to 32766 or 49152 to 65534. • Do not use the **control** keyword with this command. The **control disable** keyword combination will disable SAA RTR control packets and cause the operation to malfunction.The default is **control enable**. • After entering this command, the command-line interface (CLI) enters SAA RTR Jitter configuration mode to allow you to specify optional characteristics for the operation.
Step 5	**dest-ipaddr** *ip-address*	(Optional) Specifies the destination IP address for the operation. • Use of this command will overwrite the IP address specified in the syntax of the **type jitter** command. • This command allows you to change the target device for the operation without disabling and reenabling the operation type.

	Command or Action	Purpose
Step 6	**dest-port** *port-number*	(Optional) The destination port number for the operation. • Use of this command will overwrite the port number specified in the syntax of the **type jitter** command. • This command allows you to change the target port for the operation without disabling and reenabling the operation type.
Step 7	**frequency** *seconds* **Example:** `Router(config-rtr-jitter)#` **`frequency 45`**	(Optional) Configures how often a probe operation is performed (number of seconds between probes). • When the operation is configured as **type jitter** (codec), by default the operation is configured to run once every 60 seconds.
Step 8	**enhanced-history interval** *seconds* **buckets** *number-of-buckets* **Example:** `Router(config-rtr-jitter)#` **`enhanced-history interval 900 buckets 100`**	(Optional) Enables enhanced history collection for the Jitter operation. • To view collected enhanced history statistics, use the **show rtr enhanced-history** command. • Enhanced-history statistics do not include voice scores. **Note** Standard SAA history statistics are not available for Jitter operations.
Step 9	**owner** *owner-id* **Example:** `Router(config-rtr-jitter)#` **`owner admin`**	(Optional) Allows you to specify a process owner for the operation, as a free text designation. • This option may be helpful in identifying who originated an operation in an environment where many operations are being run.
Step 10	**tag** *text* **Example:** `Router(config-rtr-jitter)#` **`tag TelnetPollServer1`**	(Optional) Allows you to specify a label to the operation, as a free text designation. • This option may be helpful in grouping multiple operations from the same or different routers.
Step 11	**vrf** *vrf-name* **Example:** `Router(config-rtr-jitter)#` **`vrf vpn-A`**	(Optional) Configures the Jitter operation to run over a specific VPN by binding the operation to a specific VRF table.
Step 12	**hours-of-statistics-kept** *hours* **Example:** `Router(config-rtr-jitter)#` **`hours-of-statistics-kept 4`**	The number of hours for which statistics are kept. • By default, the router will retain statistics for the last two hours the operation was running. If the operation runs more than the specified number of hours, the oldest data will be replaced by newer data. • If the operation stops running, the data will be kept indefinitely, but for only the last *x* hours the operation was running. (For example, 2 hours worth of data will be kept in memory indefinitely.)

	Command or Action	Purpose
Step 13	**threshold** *milliseconds* **Example:** `Router(config-rtr-jitter)# threshold 2000`	(Optional) Configures the rising threshold, in milliseconds, for timeout or connection-loss events. • By default, if the response time is greater than 5000 milliseconds (5 seconds), this is considered to be a rising threshold crossing event. You can configure triggers for these events (such as SNMP traps) using the **rtr reaction** commands in global configuration mode.
Step 14	**timeout** *milliseconds* **Example:** `Router(config-rtr-jitter)# timeout 10000`	(Optional) Configures the timeout value for a probe operation.
Step 15	**tos** *number* **Example:** `Router(config-rtr-jitter)# tos 160`	(Optional) Sets the Type Of Service byte value (including IP Precedence data) for request packtes. • For example, the Differentiated Services Code Point (DSCP) value 46 is ToS value 184.
Step 16	**verify-data** **Example:** `Router(config-rtr-jitter)# verify-data`	(Optional) Configures SAA to check each probe response for data corruption. **Note** Only use the **verify-data** command when you suspect corruption may be an issue. .

Step 17	**exit**
	
	Example:
	`Router(config-rtr-jitter)# exit`
	`Router(config)#`
		Exits from SAA RTR Jitter configuration mode to global configuration mode.

Step 18	**rtr schedule** *operation-number* [**life** {**forever** | *seconds*}] [**start-time** {**pending** | **now** | *hh***:***mm day month*}] [**ageout** *seconds*] **Example:** `Router(config)# rtr schedule 10 life forever start-time now`	Schedules the operation by specifying when the operation should start, and how long the operation should run. • The **life** keyword is used to configure how long the operation actively collects information. The default life time of the operation is 3600 seconds (1 hour). • The default state for this command is **start-time pending**. If the operation is in a pending state, you can define the conditions under which the operation makes the transition from pending to active with the **rtr reaction-trigger** global configuration command. When the operation is in an active state it immediately begins collecting information. • After starting the operation, you can stop the operation and clear the operation from the configuration using the **no rtr** *operation-number* command in global configuration mode.

The following commands, available in SAA RTR Jitter configuration mode, are not valid for Jitter (codec) operations:

- **buckets-of-history-kept**
- **filter-for-history**

- **lives-of-history-kept**
- **distributions-of-statistics-kept**
- **statistics-distribution-interval**
- **request-data-size**

Note The **show rtr configuration** command will list the values for th e"Number of statistic distribution buckets kept" and "Statistic distribution interval (milliseconds)", but these values not not apply to Jitter (codec) operations.

You can check the statistics for the latest operation at any time using the **show rtr operational-state** command. To view collected statistics after letting the operation run for a desired amount of time, use the **show rtr collection-statistics** command.

Configuration Example for SAA VoIP UDP Operations

In the following example, a Jitter (codec) operation is configured, then output from the corresponding show commands is given. This example assumes that the SAA RTR Responder is enabled on the device at 209.165.200.225.

- SAA VoIP UDP Operation Configuration: Example, page 12
- SAA VoIP UDP Operation Statistics Output: Example, page 13

SAA VoIP UDP Operation Configuration: Example

```
Router> enable
Password:
Router# configure terminal
Enter configuration commands, one per line.  End with the end command.
Router(config)# rtr 10
Router(config-rtr)# type jitter dest-ipaddr 209.165.200.225 dest-port 16384 codec g711alaw
advantage-factor 2
Router(config-rtr-jitter)# owner admin_bofh
Router(config-rtr-jitter)# exit
Router(config)# rtr schedule 10 start-time now
Router(config)# end
Router#
Router# show running-config | begin rtr 10

rtr 10
 type jitter dest-ipaddr 209.165.200.225 dest-port 16384 codec g711alaw advantage-factor 2
 owner admin_bofh
rtr schedule 10 start-time now
  .
  .
  .
Router# show rtr 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 rtr configuration** command output. Instead, values for "Codec Packet Size:", "Codec Number of Packets", and "Codec Interval (milliseconds)" are displayed.

SAA VoIP UDP Operation Statistics Output: Example

Use the **show rtr operational-state** command and the **show rtr collection-statistics** command to show Voice scores (ICPIF and MOS values) for the Jitter (codec) operation.

```
Router# show rtr operational-state 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
!
```

```
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

Router# show rtr collection-statistics 10
Entry number: 10
Start Time Index: 12:57:45.931 UTC Sun Oct 26 2003
Number of successful operations: 60
Number of operations over threshold: 0
Number of failed operations due to a Disconnect: 0
Number of failed operations due to a Timeout: 0
Number of failed operations due to a Busy: 0
Number of failed operations due to a No Connection: 0
Number of failed operations due to an Internal Error: 0
Number of failed operations due to a Sequence Error: 0
Number of failed operations due to a Verify Error: 0
Voice Scores:
MinOfICPIF: 2   MaxOfICPIF: 20   MinOfMos: 3   MaxOfMos: 5
RTT Values:
NumOfRTT: 600   RTTAvg: 20      RTTMin: 19      RTTMax: 22
RTTSum: 12100   RTTSum2: 244292
Packet Loss Values:
PacketLossSD: 0 PacketLossDS: 0
PacketOutOfSequence: 0  PacketMIA: 0    PacketLateArrival: 0
InternalError: 0        Busies: 0

Jitter Values:
NumOfJitterSamples: 540
MinOfPositivesSD: 1     MaxOfPositivesSD: 1
NumOfPositivesSD: 26    SumOfPositivesSD: 26    Sum2PositivesSD: 26
MinOfNegativesSD: 1     MaxOfNegativesSD: 1
NumOfNegativesSD: 19    SumOfNegativesSD: 19    Sum2NegativesSD: 19
MinOfPositivesDS: 1     MaxOfPositivesDS: 1
NumOfPositivesDS: 43    SumOfPositivesDS: 43    Sum2PositivesDS: 43
MinOfNegativesDS: 1     MaxOfNegativesDS: 2
NumOfNegativesDS: 43    SumOfNegativesDS: 44    Sum2NegativesDS: 46
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
```

Where to Go Next

For information on configuring additional characteristics of SAA operations, refer to the Cisco IOS Release 12.2 document, "Network Monitoring Using Cisco Service Assurance Agent." For additional command details, refer to the *Cisco IOS Configuration Fundamentals and Network Management Command Reference* for Release 12.3T.

Additional References

The following sections provide references related to the SAA VoIP UDP Operation feature.

Related Documents

Related Topic	Document Title
Voice over IP (VoIP) codecs	"Understanding Codecs: Complexity, Hardware Support, MOS, and Negotiation" http://www.cisco.com/warp/public/788/voip/codec_complexity.html
Jitter in Packet Voice Networks	"Understanding Jitter in Packet Voice Networks (Cisco IOS Platforms)" http://www.cisco.com/warp/public/788/voice-qos/jitter_packet_voice.html
Cisco Service Assurance Agent (SAA) configuration	"Network Monitoring Using Cisco Service Assurance Agent" http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122cgcr/ffun_c/fcfprt3/fcf017.htm
PSTN Fallback for Voice Gateways	"SIP: Measurement-Based Call Admission Control for SIP" http://www.cisco.com/univercd/cc/td/doc/product/software/ios122/122newft/122t/122t15/ftcacsip.htm

Related Standards

Standard	Title
ITU-T Recommendation G.107 (2003)	*The E-model, a computation model for use in transmission planning*
ITU-T Recommendation G.113 (1996)	*Transmission impairments*
ITU-T Recommendation G.113 (2001)	*Transmission impairments due to speech processing*
ITU-T Recommendation G.711 (1998)	*Pulse code modulation (PCM) of voice frequencies* (also known as the G.711 Voice Codec)
ITU-T Recommendation G.729 Annex A (1996)	*Reduced complexity 8 kbit/s CS-ACELP speech codec* (also known as the G.729/A/B Speech Codec)
ITU-T Recommendation P.800.1 (2003)	Mean Opinion Score (MOS) terminology

Full support for these standards 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.

Related MIBs

MIB	MIB Link
CISCO-RTTMON-MIB (The Cisco Service Assurance Agent MIB; also know as The Round Trip Time Monitor MIB) REVISION: "200304150000Z"	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

Related RFCs

RFC[1]	Title
RFC 768	*User Datagram Protocol*
RFC 1889	*RTP: A Transport Protocol for Real-Time Applications*

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

Technical Assistance

Description	Link
Technical Assistance Center (TAC) home page, containing 30,000 pages of searchable technical content, including links to products, technologies, solutions, technical tips, and tools. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/public/support/tac/home.shtml

Command Reference

This section documents new and modified commands. All other commands used with this feature are documented in the Cisco IOS Release 12.3T command reference publications.

- type jitter (codec)
- show rtr collection-statistics
- show rtr operational-state

type jitter (codec)

To configure a Service Assurance Agent (SAA) Jitter operation that returns VoIP scores, use the **type jitter** (codec) command in SAA RTR configuration mode. To reset the operation type to null, use the **no** form of this command.

type jitter dest-ipaddr {*hostname* | *ip-address*} **dest-port** *port-number* **codec** *codec-type* [**codec-numpackets** *number-of-packets*] [**codec-size** *number-of-bytes*] [**codec-interval** *time-interval*] [**advantage-factor** *value*] [**source-ipaddr** {*hostname* | *ip-address*}] [**source-port** *port-number*] [**control** {**enable** | **disable**}]

no type jitter dest-ipaddr {*hostname* | *ip-address*} **dest-port** *port-number* **codec** *codec-type*

Syntax Description

dest-ipaddr {*hostname* | *ip-address*}	Destination for the operation, as an IP host name or IP address.
dest-port *port-number*	Specifies the destination port number. For Jitter (codec) operations, the port number should be an even number in the range of 16384 to 32766 or 49152 to 65534.
codec *codec-type*	Enables the generation of estimated Voice quality scores in the form of Calculated Planning Impairment Factor (ICPIF) and Mean Opinion Score (MOS) values. The codec type should match the encoding algorithm you are using for VoIP transmissions. The following codec-type keywords are available: • **g711alaw**—The G.711 A-Law codec (64 kbps transmission) • **g711ulaw**—The G.711 muHmm-Law codec (64 kbps transmission) • **g729a**—The G.729A codec (8 kbps transmission) Configuring the codec type sets default values for the variables **codec-numpackets**, **codec-size**, and **codec-interval** in this command. See Table 8 for details.
codec-numpackets *number-of-packets*	(Optional) Specifies the number of packets to be transmitted per probe operation. The valid range is from 1 to 60000 packets. The default is 1000 packets.
codec-size *number-of-bytes*	(Optional) Specifies the number of bytes in each packet transmitted. (Also called the payload size or request size.) The valid range is from 16 to 1500 packets. The default varies by codec; see Table 8.
codec-interval *milliseconds*	Specifies the interval (delay) between packets that should be used for the operation, in milliseconds (ms). The valid range is from 1 to 60000 ms. By default, packets are sent 20 ms apart.
advantage-factor *value*	Specifies the expectation factor to be used for ICPIF calculations. This value is subtracted from the measured impairments to yield the final ICPIF value (and corresponding MOS value). See the "Usage Guidelines" section for recommended values. The valid range is from 0 to 20. The default is 0.
source-ipaddr {*hostname* | *ip-address*}	(Optional) Source IP address to be applied to operations, as a host name or IP address.

source-port *port-number*	(Optional) Source port that SAA probe packets should be sent from. All probe traffic for this Jitter operation will be sent from the specified port.
control {*enable* | *disable*}	(Optional) Enables or disables the sending of SAA control messages to the SAA RTR Responder. By default, SAA control messages are sent to the target device to establish a connection with the SAA RTR Responder. **Note** Control messages are enabled by default. Disabling the SAA control messages for Jitter operations is not recommended. If you disable SAA control messages, packet loss statistics and IP telephony scores will not be generated accurately.

Defaults

No SAA operation type is associated with the operation number being configured.

Command Modes

SAA RTR configuration (config-rtr)

Command History

Release	Modification
12.0(5)T	The **type jitter** command was introduced.
12.3(4)T	The **type jitter (codec)** command was introduced.

Usage Guidelines

The **type jitter** command is executed in SAA RTR configuration mode. You must configure the type of operation before you can configure any of the other characteristics of the operation.

Prior to sending operations to the target device, SAA sends control messages to the SAA RTR Responder to enable the destination port. You must enable the SAA RTR Responder on the target router (using the **rtr responder** command) for the Jitter operation to function correctly.

The **type jitter (codec)** command is a special implementation of the **type jitter** command. This command is documented separately from the standard **type jitter** command because when you specify the codec in the command syntax, the standard configuration options are replaced with codec-specific keywords and arguments.

The SAA 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**.

To generate MOS and ICPIF scores, you specify the codec type used for the connection when configuring the 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 Table 8 for specifics.) However, you are given the option, if needed, to manually configure these parameters in the syntax of the **type jitter** command.

Table 8 shows the default parameters that are configured for the operation by codec.

Table 8 Default Jitter Operation Parameters by Codec

Codec	Default Number of Packets (n); [codec-numpackets]	Packet Payload (s) [codec-size][1]	Default Interval between Packets (t) [codec-interval]	Frequency of Operations (f)
G.711 mu-Law (g711ulaw)	1000	160 bytes	20 milliseconds	Once every 60 seconds
G.711 A-Law (g711alaw)	1000	160 bytes	20 milliseconds	Once every 60 seconds
G.729A (g729a)	1000	20 bytes	20 milliseconds	Once every 60 seconds

1. The actual Request Size of each packet will contain an additional 12 bytes of RTP header data in order to simulate the RTP/UDP/IP/Layer2 protocol stack.

For example, if you configure the 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 160 bytes (plus 12 header bytes) of synthetic data (**s**), sent 20 milliseconds apart (**t**).

Note that the operational frequency is configured with the **frequency** SAA RTR Jitter configuration mode command, and is not one of the syntax options of the **type jitter** command. In other words, use of the **frequency** command when configuring the Jitter (codec) operation is optional.

The **advantage-factor** *value* syntax allows you to specify an access Advantage Factor (also called the Expectation Factor). Table 9, adapted from ITU-T Rec. G.113, defines a set of provisional maximum values for Advantage Factors in terms of the service provided.

Table 9 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
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 Table 9 should be considered as the absolute upper limits for *A*. The default Advantage Factor for SAA Jitter operations is always zero.

Examples

In the following example, SAA operation 10 is created and configured as a Jitter (codec) operation using the destination IP address 209.165.200.225 and the destination port number 3000. The operation is configured to use the characteristics of the G.711 A-Law codec, which means the operation will consist of 1000 packets, each of 172 bytes (160 plus 12 header bytes), sent 20 milliseconds apart. The default value for the Advantage Factor is used, and by default the operation will run once a minute.

```
Router(config)# rtr 10
Router(config-rtr)# type jitter dest-ipaddr 209.165.200.225 dest-port 3000 codec g711alaw
Router(config-rtr-jitter)# default frequency
Router(config-rtr-jitter)# tag jitter-with-voice-scores
Router(config-rtr-jitter)# exit
Router(config)# end
Router# show rtr configuration 10

Entry number: 10
Owner:
Tag: jitter-with-voice-scores
Type of operation to perform: jitter
Target address: 209.165.200.225
Source address: 0.0.0.0
Target port: 3000
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: 0
 .
 .
 .
Operation frequency (seconds): 60
 .
 .
 .
```

Related Commands

Command	Description
rtr	Specifies an SAA operation and enters SAA RTR configuration mode.
show rtr configuration	Displays the current configuration data specific to SAA operations.

show rtr collection-statistics

To display statistical errors for all Service Assurance Agent (SAA) operations or a specified operation, use the **show rtr collection-statistics** command in EXEC mode.

show rtr collection-statistics [*operation-number*] [**tabular** | **full**]

Syntax Description

operation-number	(Optional) Number of the SAA operation to display.

Defaults

Shows statistics for the past two hours.

Command Modes

EXEC

Command History

Release	Modification
11.2	This command was introduced.
12.0(5)T	The output for this command was expanded to show information for Jitter operations.
12.1	The **tabular** and **full** keywords were removed.
12.1(1)T	The output for this command was expanded to show information for the FTP operation type and for One Way Delay Jitter operations.
12.2(8)T, 12.2(8)S	Output for "NumOfJitterSamples" was added (CSCdv30022).
12.2(11)T	The SAA Engine II was implemented. The maximum number of operations was increased from 500 to 2000.
12.3(4)T	Output (MOS and ICPIF scores) for the Jitter (codec) operation type was added.

Usage Guidelines

Use the **show rtr collection-statistics** command to display information such as the number of failed operations and the failure reason. You can also use the **show rtr distribution-statistics** and **show rtr totals-statistics** commands to display additional statistical information.

This command shows information collected over the past two hours, unless you specify a different amount of time using the **hours-of-statistics-kept** command.

For One Way Delay Jitter operations, the clocks on each device must be synchronized using NTP (or GPS systems). If the clocks are not synchronized, one way measurements are discarded. (If the sum of the source to destination (SD) and the destination to source (DS) values is not within 10 percent of the round trip time, the one way measurement values are assumed to be faulty, and are discarded.)

Examples

The following is sample output from the **show rtr collection-statistics** command, where operation 10 is a Jitter (codec) operation:

```
Router# show rtr collection-statistics
```

```
Entry Number: 10

Start Time Index: 12:57:45.931 UTC Wed Mar 12 2003
Number of successful operations: 60
Number of operations over threshold: 0
Number of failed operations due to a Disconnect: 0
Number of failed operations due to a Timeout: 0
Number of failed operations due to a Busy: 0
Number of failed operations due to a No Connection: 0
Number of failed operations due to an Internal Error: 0
Number of failed operations due to a Sequence Error: 0
Number of failed operations due to a Verify Error: 0
Voice Scores:
 MinOfICPIF: 2   MaxOfICPIF:  20   MinOfMos: 3   MaxOfMos: 5
RTT Values:
 NumOfRTT: 600  RTTSum: 3789  RTTSum2: 138665
 Packet Loss Values:
 PacketLossSD: 0  PacketLossDS: 0
 PacketOutOfSequence: 0  PacketMIA: 0  PacketLateArrival: 0
 InternalError: 0  Busies: 0
Jitter Values:
 NumOfJitterSamples: 540
 MinOfPositivesSD: 1   MaxOfPositivesSD: 2
 NumOfPositivesSD: 26  SumOfPositivesSD: 31   Sum2PositivesSD: 41
 MinOfNegativesSD: 1   MaxOfNegativesSD: 4
 NumOfNegativesSD: 56  SumOfNegativesSD: 73   Sum2NegativesSD: 133
 MinOfPositivesDS: 1   MaxOfPositivesDS: 338
 NumOfPositivesDS: 58  SumOfPositivesDS: 409  Sum2PositivesDS: 114347
 MinOfNegativesDS: 1   MaxOfNegativesDS: 338
 NumOfNegativesDS: 48  SumOfNegativesDS: 396  Sum2NegativesDS: 114332
 Interarrival jitterout: 0       Interarrival jitterin: 0
One Way Values:
 NumOfOW: 440
 OWMinSD: 2  OWMaxSD: 6    OWSumSD: 1273  OWSum2SD: 4021
 OWMinDS: 2  OWMaxDS: 341  OWSumDS: 1643  OWSum2DS: 120295
```

The values shown indicate the aggregated values for the current hour. RTT stands for Round-Trip-Time. SD stands for Source-to-Destination. DS stands for Destination-to-Source. OW stands for One Way. Table 10 describes the significant fields shown in this output.

Table 10 show rtr collection-statistics Field Descriptions

Field	Description
Voice Scores:	Indicates that Voice over IP statistics appear on the following lines. Voice score data is computed when the operation type is configured as **type jitter (codec)**.

Table 10 show rtr collection-statistics Field Descriptions (continued)

Field	Description
ICPIF	The Calculated Planning Impairment Factor (ICPIF) value for the operation. The ICPIF value is computed by SAA using the formula $Icpif = Io + Iq + Idte + Idd + Ie - A$, where • the values for *Io*, *Iq*, and *Idte* are set to zero, • the value *Idd* is computed based on the measured one way delay, • the value *Ie* is computed based on the measured packet loss, • and the value of A is specified by the user. 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." **Note** This value is intended only for relative comparisons, and may not match ICPIF values generated using alternate methods.
MinOfICPIF:	The lowest (minimum) ICPIF value computed for the collected statistics.
MaxOfICPIF:	The highest (maximum) ICPIF value computed for the collected statistics.
Mos	The estimated Mean Opinion Score (Conversational Quality, Estimated) for the latest iteration of the operation. The MOS-CQE is computed by SAA as a function of the ICPIF. MOS values are expressed as a whole number from 1 to 5, with 5 being the highest level of quality, and 1 being the lowest level of quality. A MOS value of 0 (zero) indicates that MOS data could not be generated for the operation.
MinOfMos:	The lowest (minimum) MOS value computed for the collected statistics.
MaxOfMos:	The highest (maximum) ICPIF value computed for the collected statistics.
RTT Values:	Indicates that Round-Trip-Time statistics appear on the following lines.
NumOfRTT	The number of successful round trips.
RTTSum	The sum of all successful round trip values (in milliseconds).
RTTSum2	The sum of squares of those round trip values (in milliseconds).
PacketLossSD	The number of packets lost from source to destination.
PacketLossDS	The number of packets lost from destination to source.
PacketOutOfSequence	The number of packets returned out of order.
PacketMIA	The number of packets lost where the direction (SD/DS) cannot be determined.
PacketLateArrival	The number of packets that arrived after the timeout.

Table 10 show rtr collection-statistics Field Descriptions (continued)

Field	Description
InternalError	The number of times an operation could not be started due to other internal failures.
Busies	The number of times this operation could not be started because the previously scheduled run was not finished.
Jitter Values:	Indicates that Jitter statistics appear on the following lines. Jitter is inter-packet delay variance.
NumOfJitterSamples:	The number of jitter samples collected. This is the number of samples that are used to calculate the following jitter statitstics.
MinOfPositivesSD MaxOfPositivesSD	The minimum and maximum positive jitter values from source to destination, in milliseconds.
NumOfPositivesSD	The number of jitter values from source to destination that are positive (i.e., network latency increases for two consecutive test packets).
SumOfPositivesSD	The sum of those positive values (in milliseconds).
Sum2PositivesSD	The sum of squares of those positive values.
MinOfNegativesSD MaxOfNegativesSD	The minimum and maximum negative jitter values from source to destination. The absolute value is given.
NumOfNegativesSD	The number of jitter values from source to destination that are negative (i.e., network latency decreases for two consecutive test packets).
SumOfNegativesSD	The sum of those values.
Sum2NegativesSD	The sum of the squares of those values.
Interarrival jitterout:	The source to destination(SD) jitter value calculation, as defined in RFC 1889.
Interarrival jitterin:	The destination to souce (DS) jitter value calculation, as defined in RFC 1889.
One Way Values	Indicates that one way measurement statistics appear on the following lines. One Way (OW) Values are the amount of time it took the packet to travel from the source router to the target router (SD) or from the target router to the source router (DS).
NumOfOW	Number of successful one way time measurements.
OWMinSD	Minimum time from the source to the destination.
OWMaxSD	Maximum time from the source to the destination.
OWSumSD	Sum of the OWMinSD and OWMaxSD values.
OWSum2SD	Sum of the squares of the OWMinSD and OWMaxSD values.

The DS values show the same information as above for Destination-to-Source Jitter values.

Related Commands

Command	Description
show ntp status	Displays the status of the Network Time Protocol configuration on your system.
show rtr configuration	Displays configuration values including all defaults for all SA Agent operations or the specified operation.
show rtr distributions-statistics	Displays statistic distribution information (captured response times) for all SAA operations or the specified operation.
show rtr totals-statistics	Displays the total statistical values (accumulation of error counts and completions) for all SAA operations or the specified operation.

show rtr operational-state

To display the operational state of all Service Assurance Agent (SAA) operations or a specified operation, use the **show rtr operational-state** command in EXEC mode.

show rtr operational-state [*operation-number*]

Syntax Description

operation-number	(Optional) Number of the SAA operation to display.

Defaults

Displays output for all running SAA operations.

Command Modes

EXEC

Command History

Release	Modification
11.2	This command was introduced.
12.0(5)T	Output for the Jitter operation type was added.
12.1	The **tabular** and **full** keywords were removed.
12.2(8)T, 12.2(8)S	Output for “NumOfJitterSamples” was added (CSCdv30022).
12.3(4)T	Output (MOS and ICPIF scores) for the Jitter (codec) operation type was added.

Usage Guidelines

Use the **show rtr operational-state** command to determine whether a connection loss, timeout, and over threshold occurred; how much life the operation has left; whether the operation is active; and the completion time. It also displays the results of the latest operation attempt.

Examples

The following example shows sample output from the **show rtr operational-state** command when the specified operation is a Jitter (codec) operation:

```
Router# show rtr operational-state
        Current Operational State
Entry number: 10
Modification time: 12:57:45.690 UTC Sun Oct 26 2003
Number of operations attempted: 3
Number of operations skipped: 0
Current seconds left in Life: 3570
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
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
```

The values shown indicate the values for the last SAA operation. RTT stands for Round-Trip-Time. SD stands for Source-to-Destination. DS stands for Destination-to-Source. OW stands for One Way. The * symbol in front of the time stamps indicates the time is synchronized using NTP or SNTP. Table 11 describes the significant fields shown in this output.

Table 11 show rtr operational-state Field Descriptions

Field	Description
Voice Scores:	Indicates that Voice over IP statistics appear on the following lines. Voice score data is computed when the operation type is configured as **type jitter (codec)**.
ICPIF:	The Calculated Planning Impairment Factor (ICPIF) value for the latest iteration of the operation. The ICPIF value is computed by SAA using the formula $Icpif = Io + Iq + Idte + Idd + Ie - A$, where • the values for *Io*, *Iq*, and *Idte* are set to zero, • the value *Idd* is computed based on the measured one way delay, • the value *Ie* is computed based on the measured packet loss, • and the value of A is specified by the user. 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.” **Note** This value is intended only for relative comparisons, and may not match ICPIF values generated using alternate methods.

Table 11 show rtr operational-state Field Descriptions (continued)

Field	Description
MOS:	The estimated Mean Opinion Score (Conversational Quality, Estimated) for the latest iteration of the operation. The MOS-CQE is computed by SAA as a function of the ICPIF. MOS values are expressed as a whole number from 1 to 5, with 5 being the highest level of quality, and 1 being the lowest level of quality. A MOS value of 0 (zero) indicates that MOS data could not be generated for the operation.
RTT Values:	Indicates that Round-Trip-Time statistics appear on the following lines.
NumOfRTT	The number of successful round trips.
RTTSum	The sum of those round trip values (in milliseconds).
RTTSum2	The sum of squares of those round trip values (in milliseconds).
Packet Loss Values:	Indicates that Packet Loss statistics appear on the following lines.
PacketLossSD	The number of packets lost from source to destination.
PacketLossDS	The number of packets lost from destination to source.
PacketOutOfSequence	The number of packets returned out of order.
PacketMIA	The number of packets lost where the direction (SD or DS) cannot be determined (MIA: “missing in action”).
PacketLateArrival	The number of packets that arrived after the timeout.
InternalError	The number of times an operation could not be started due to other internal failures.
Busies	The number of times this operation could not be started because the previously scheduled run was not finished.
Jitter Values:	Indicates that jitter operation statistics appear on the following lines. Jitter is inter-packet delay variance.
NumOfJitterSamples:	The number of jitter samples collected. This is the number of samples that are used to calculate the following jitter statitstics.
MinOfPositivesSD MaxOfPositivesSD	The minimum and maximum positive jitter values from source to destination, in milliseconds.
NumOfPositivesSD	The number of jitter values from source to destination that are positive (i.e., network latency increases for two consecutive test packets).
SumOfPositivesSD	The sum of those positive values (in milliseconds).
Sum2PositivesSD	The sum of squares of those positive values.
MinOfNegativesSD MaxOfNegativesSD	The minimum and maximum negative jitter values from source to destination. The absolute value is given.

Table 11 show rtr operational-state Field Descriptions (continued)

Field	Description
NumOfNegativesSD	The number of jitter values from source to destination that are negative (i.e., network latency decreases for two consecutive test packets).
SumOfNegativesSD	The sum of those values.
Sum2NegativesSD	The sum of the squares of those values.
Interarrival jitterout:	The source to destination(SD) jitter value calculation, as defined in RFC 1889.
Interarrival jitterin:	The destination to souce (DS) jitter value calculation, as defined in RFC 1889.
One Way Values	Indicates that One Way measurement statistics appear on the following lines. One Way (OW) Values are the amount of time it took the packet to travel from the source router to the target router (SD) or from the target router to the source router (DS).
NumOfOW	Number of successful one way time measurements.
OWMinSD	Minimum time from the source to the destination.
OWMaxSD	Maximum time from the source to the destination.
OWSumSD	Sum of the OWMinSD and OWMaxSD values.
OWSum2SD	Sum of the squares of the OWMinSD and OWMaxSD values.

The DS values show the same information as above for Destination-to-Source Jitter values.

Related Commands

Command	Description
show rtr configuration	Displays configuration values including all defaults for all SAA operations or a specified operation.

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.[1]

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

Note Refer to *Internetworking Terms and Acronyms* for terms not included in this glossary.

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