DSP Voice Quality Metrics for MGCP

First Published: October 6, 2003
Last Updated: May 7, 2007

Digital signal processor (DSP) voice quality metrics improve your ability to monitor, analyze, and ultimately meet your quality of service (QoS) objectives for your network.

Finding Feature Information in This Module

Your Cisco IOS software release may not support all of the features documented in this module. To reach links to specific feature documentation in this module and to see a list of the releases in which each feature is supported, use the "Feature Information for DSP Voice Quality Metrics for MGCP" section on page 54.

Finding Support Information for Platforms and Cisco IOS and Catalyst OS Software Images

Use Cisco Feature Navigator to find information about platform support and Cisco IOS and Catalyst OS software image support. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

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Prerequisites for DSP Voice Quality Metrics for MGCP

- Both the source and destination routers must be running a Cisco IOS image with the Cisco IOS IP Voice or higher grade feature package.
- The source router must have a network module with a c5510 or c549 DSP. The destination router need not have a network module with a DSP.

Restrictions for DSP Voice Quality Metrics for MGCP

Note Do not enable the DSP Voice Quality Metrics for MGCP feature on all media gateways associated with a single Cisco PGW 2200 pair. Doing so can severely impact the call-processing ability of your system. You can use the statistics control function in the feature to limit the impact on call processing.

Information About DSP Voice Quality Metrics for MGCP

Metrics for DSP performance on Media Gateway Control Protocol (MGCP) networks are useful for maintaining service, tracking quality issues, and troubleshooting problems. Voice quality metrics on the DSP can help you to evaluate the performance of your VoIP network.DSP metrics can be used to do the following:

- Determine which calls have unacceptable voice quality and provide data on possible reasons for the problem.
- Determine the path taken by low-quality calls, including the network source (TDM or IP), endpoint or midpoint equipment, or other call control processing problems.
- Collect call quality data for trend analysis.

Voice quality metrics for DSPs on MGCP networks can be gathered through the following methods:

- Voice-Quality Statistics in DLCX Messages
- Call Detail Records
- IP Service Level Agreements
- Syslog

Voice-Quality Statistics in DLCX Messages

The DSP statistics gathering function on Cisco media gateways provides a way to trace an MGCP call between a Cisco gatekeeper and the Cisco IOS gateway by including the MGCP call ID and the DS0 and DSP channel ID in call-active and call-history records. These DSP statistics are sent as part of the MGCP Delete Connection (DLCX) message. By correlating an MGCP call on the Cisco gatekeeper with the call record on the gateway, additional statistics from the DSP can be understood and debugged for problems related to voice quality. This feature also provides a method to limit the amount of statistics sent to the Cisco gatekeeper to control the impact to call processing performance. The Support of DSP Voice Quality Statistics feature gathers the DSP voice quality statistics on the Cisco gatekeeper.

For more information on the DSP voice quality statistics that can be gathered on Cisco media gateways and how to configure the priority settings on the media gateway, see the *DSP Voice Quality Statistics in DLCX Messages* feature module.

Parameter Priorities

Running voice quality statistics can impact the router performance. To reduce performance issues, the data can be provisioned by priority.

- Priority 1 parameters are: DSP/TX, DSP/RX, DSP/PD, DSP/LE, DSP/EC, DSP/CS, DSP/DL.
- Priority 2 parameters are: DSP/PE, DSP/ER, DSP/IC, DSP/KF, DSP/RF, DSP/UC. Priority 2 parameters include all of the priority 1 parameters as well.

Call Detail Records

Call Detail Records (CDRs) for voice calls can be output in RADIUS VSAs or system log (syslog) messages. A RADIUS server can be configured to collect accounting data during the accounting process for each call leg created on the Cisco voice gateway. An integration partner can use this information for post-processing activities, such as generating billing records and network analysis.

IP Service Level Agreements

Cisco IOS IP Service Level Agreements (SLAs) allows you to analyze IP service levels for IP applications and services, to increase productivity, to lower operational costs, and to reduce the frequency of network outages. Cisco IOS IP SLAs uses active traffic monitoring—the generation of traffic in a continuous, reliable, and predictable manner—for measuring network performance. Using Cisco IOS IP SLAs, service provider customers can measure and provide service level agreements, and enterprise customers can verify service levels, verify outsourced service level agreements, and understand network performance. Cisco IOS IP SLAs can perform network assessments, verify quality of service (QOS), ease the deployment of new services, and assist administrators with network troubleshooting. You can Cisco IOS IP SLAs using the Cisco IOS command-line interface (CLI) or Simple Network Management Protocol (SNMP) through the Cisco Round-Trip Time Monitor (RTTMON) and SYSLOG Management Information Bases (MIBs).

You can find detailed information about Cisco IP SLAs in the following documents:

- If you want to configure multiple Cisco IOS IP SLAs operations at once, see the "IP SLAs—Multiple Operation Scheduling" chapter of the *Cisco IOS IP SLAs Configuration Guide*, Release 12.4.
- If you want to configure threshold parameters for an IP SLAs operation, see the "IP SLAs—Proactive Threshold Monitoring" chapter of the *Cisco IOS IP SLAs Configuration Guide*, Release 12.4.
- If you want to configure other types of IP SLAs operations, see the "Where to Go Next" section of the "Cisco IOS IP SLAs Overview" chapter of the *Cisco IOS IP SLAs Configuration Guide*, Release 12.4.

Syslog

Syslog is a method to collect messages from devices to a server running a syslog daemon. Logging to a central syslog server helps in aggregation of logs and alerts. Cisco devices can send their log messages to a UNIX-style SYSLOG service that simply accepts messages and stores them in files or prints them according to a simple configuration file. This form of logging is very useful for Cisco devices because it can provide protected long-term storage for logs. This is useful both in routine troubleshooting and in incident handling.

Troubleshooting Using DSP Voice Quality Metrics

The Cisco call agent can capture voice quality statistics sent from MGCP-controlled media gateways and can propagate the statistics into call detail records (CDRs) at the end of each call. Cisco gateways send voice quality statistics to the Cisco call agent.

Most voice quality statistics are available from the DSP and are controlled with RTP Control Protocol (RTCP) report interval statistics polling. The mean and maximum values are calculated by Cisco IOS software-based polling. This results in an additional CPU load for each call, which can be controlled by the configured polling interval using the **ip rtcp report interval** commands.

The playout delay, playout error, and DSP receive and transmit statistics are automatically polled periodically. You can add polling for the voice quality statistics, level, and error parameters. For logging the voice quality statistics using Syslog, the existing VoIP gateway accounting has been extended. Use the **ip rtcp report interval** command reference for more information about statistics polling.

DSP voice quality metrics can be used to troubleshoot the following problems:

- Troubleshooting One Way Audio, page 4
- Troubleshooting Echo, page 4
- Troubleshooting Voice Levels, page 5

For additional DSP troubleshooting information, see the "Checking the Digital Signal Processors" section in the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.

Troubleshooting One Way Audio

Troubleshoot one-way audio problems by using the following DSP voice quality metrics:

- Use the tranmission (DSP/TX) and receive (DSP/RX) statistics to determine if packets are only moving in one direction.
- Use the level (DSP/LE) statistics to check incoming and outgoing activity.
- Use endpoint configuration (DSP/CE) statistics to determine if there is a codec mismatch.

Troubleshooting Echo

Troubleshoot echo problems by using the following DSP voice quality metrics:

- Check the delay (DSP/DL) and R-factor (DSP/RF) statistics. You might find perceptible delay between when the originating signal is transmitted and when the echo returns. In most telephones, sidetone helps mask some of the echo. Echos must be delayed by at least 20 milliseconds to be perceived.

- Check the level (DSP/LE) statitstic for sufficient echo amplitude. If the amplitude of the echo is low, it can go unnoticed.

For additional echo troubleshooting information, see the "Troubleshooting Quality of Service for VoIP" section in the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.

Troubleshooting Voice Levels

Use the level (DSP/LE) statistics to check voice levels. As much as possible, try to achieve a uniform input decibel level to the packet voice network. Doing so eliminates or at least reduces any voice distortion because of incorrect input and output decibel levels. Adjustments to levels might be required by the type of equipment connected to the network or by local country-specific conditions.

For additional voice level troubleshooting information, see the "Troubleshooting Quality of Service for VoIP" section in the *Cisco IOS Voice Troubleshooting and Monitoring Guide*.

DSP Voice Quality Metrics Descriptions

DSP voice quality metrics are described in the following sections:

- DSP Support, page 5
- DSP Voice Quality Metrics Summary, page 6
- DSP Voice Quality Metrics Details, page 9

DSP Support

DSP voice quality metrics vary by DSP technology. Use Table 1 to determine which DSP metrics your platform supports.

Table 1 Voice Quality Statistics Usage by DSP Technology

DSP Technology	Platform	Voice Quality Statistics
MSA V6	Cisco AS5350, Cisco AS5350XM Cisco AS5400, Cisco AS5400HPX Cisco 5400XM, and Cisco AS5850 with a NPE60 or NPE108 universal port feature card	DSP/TX DSP/RX DSP/PD DSP/PE DSP/LE DSP/ER DSP/IC
TIC5510	• Cisco 2800 series and Cisco 3800 series integrated services routers with PVDM2 modules • Cisco VG224 voice gateway • Cisco IAD2430 series integrated access devices • Cisco 2600XM, Cisco 2691, Cisco 3700 series access routers, and Cisco 2811, Cisco 2821, Cisco 2851, Cisco 3800 series integrated services routers with the following network modules: NM-HDV2, NM-HDV2-1T1/E1, NM-HD-1V, NM-HD-2V, NM-HD-2VE • Cisco 2821, Cisco 2851, Cisco 3825, and Cisco 3845 with the EVM-HD-8FXS/DID module	DSP/TX DSP/RX DSP/PD DSP/PE DSP/LE DSP/ER DSP/IC DSP/EC DSP/KF DSP/CS DSP/RF DSP/UC DSP/DL

DSP Voice Quality Metrics Summary

Table 2 contains a summary of all supported DSP voice quality metrics:

Table 2 DSP Metrics for MGCP

Parameter	Parameter Description	Element	Element Description	Units	Priority	DSP
DSP/TX	Transmission statistics	PK	Transmission (TX) packets	packet count	1	All
		SG	Signaling packets	packet count		
		NS	Noise packets	packet count		
		DU	Transmission duration	millliseconds		
		VO	Voice transmission duration	milliseconds		

Table 2 DSP Metrics for MGCP (continued)

Parameter	Parameter Description	Element	Element Description	Units	Priority	DSP
DSP/RX	Receive statistics	PK	Voice packets	packet count	1	All
		SG	Signaling packets	packet count		
		CF	Comfort noise packets	packet count		
		RX	Receive duration	milliseconds		
		VO	Voice receive duration	milliseconds		
		BS	Bad sequence	sequence count		
		BP	Bad protocol	event count		
		LP	Late packets	packet count		
DSP/PD	Playout delay	CU	Current size of playout delay	milliseconds	1	All
		MI	Minimum value of CU	milliseconds		
		MA	Maximum value of CU	milliseconds		
		CO	RTP clock offset	ticks (0.125ms)		
		IJ	Interarrival jitter	milliseconds		
DSP/PE	Playout error	PC	Predictive concealment, cumulative duration	milliseconds	1	All
		IC	Interpolative concealment, cumulative duration	milliseconds		
		SC	Silence concealment, cumulative duration	milliseconds		
		RM	Retroactive memory update	event count		
		BO	Buffer overflow, cumulative number buffer overflow errors.	error count		
		EE	Number of talkspurt endpoint errors	error count		
DSP/LE	Level	TP	Transmission power	0.1 dBm units	1	All
		TX	Transmission mean	0.1 linear PCM units		
		RP	Receive power in	0.1 dBm units		
		RM	Receive mean in	0.1 linear PCM units		
		BN	Background noise	0.1 dBm units		
		ER	Echo return loss (ERL) level	0.1 dBm units		
		AC	Acom level	0.1 dBm units		
		TA	Current transmit activity	milliseconds		
		RA	Current receive activity	milliseconds		

Table 2 DSP Metrics for MGCP (continued)

Parameter	Parameter Description	Element	Element Description	Units	Priority	DSP
DSP/ER	Error statistics	RD	Receive dropped	packet count	2	All
		TD	Transmission dropped	packet count		
		RC	Receive control	packet count		
		TC	Transmission control	packet count		
DSP/IC	ICPIF statistics	IC	ICPIF value for measuring voice quality	ICPIF value	2	All
DSP/EC	Endpoint configuration	CI	Codec ID	codec type	1	TIC5510 only
		FM	Frame size	milliseconds		
		FP	Frames per packet	frame count		
		VS	VAD enabled flag	0=disable, 1=enabled		
		GT	Transmission gain factor (linear)	>1=loss pad		
		GR	Reception gain factor (linear)	>1=loss pad		
		JD	Jitter buffer mode	1 through 4		
		JN	Jitter buffer nominal playout delay	milliseconds		
		JM	Minimum playout delay	milliseconds		
		JX	Maximum playout delay	milliseconds		
DSP/KF	K-factor statistics	KF	K-factor MOS-LQO estimate (instantaneous)	1 (very poor) to 5 (very good)	2	TIC5510 only
		AV	Average k-factor score	1 (very poor) to 5 (very good)		
		MI	Minimum k-factor score	1 (very poor) to 5 (very good)		
		BS	Baseline (maximum) k-factor score	1 (very poor) to 5 (very good)		
		NB	Number of bursts	burst count		
		FL	Average frame loss count	frame count		
		NW	Number of windows	window count		
		VR	Version ID	ID		

Table 2 DSP Metrics for MGCP (continued)

Parameter	Parameter Description	Element	Element Description	Units	Priority	DSP
DSP/CS	Concealment statistics	CR	Concealment ratio (instantaneous)	concealment time over speech time	1	TIC5510 only
		AV	Average k-factor score	score		
		MX	Maximum CR	score		
		CT	Total concealment time	milliseconds		
		TT	Total speech time	milliseconds		
		OK	Ok seconds	seconds		
		CS	Concealed seconds	seconds		
		SC	Severely concealed seconds	seconds		
		TS	Concealment threshold	milliseconds		
		DC	Dead Connection Detector (not supported)	N/A		
DSP/RF	R-factor statistics	ML	R-factor MOS listening quality objective	score	2	TIC5510 only
		MC	R-factor MOS-CQE	score		
		R1	R-factor for LQ profile1	score		
		R2	R-factor for LQ profile2	score		
		IF	Effective codec impairment (Ie_eff)	score		
		ID	Delay factors	score		
		IE	Codec baseline score (Ie)	score		
		BL	Codec baseline (Bpl)	score		
		R0	Nominal value for R0 (default)	score		
DSP/UC	User concealment	U1	User concealment seconds 1 count (UCS1)	seconds	2	TIC5510 only
		U2	User concealment seconds 2 count (UCS2)	seconds		
		T1	UCS1 threshold in ms	seconds		
		T2	UCS2 threshold in ms	seconds		
DSP/DL	Delay statistics	RT	Round trip delay	milliseconds	1	TIC5510 only
		ED	End system delay	milliseconds		

DSP Voice Quality Metrics Details

This section contains detailed descriptions of the DSP voice quality metrics parameters:

- DSP/TX: Transmission Statistics, page 10
- DSP/RX: Receive Statistics, page 10
- DSP/PD: Playout Delay (Jitter Buffer), page 11

DSP/TX: Transmission Statistics

Transmission statistics can show if a router is effectively transmitting packets. The statistics show transmissions of outgoing voice, signaling, and comfort noise packets. In addition, the lengths of both the overall transmission and the voice call are tracked.

The following example shows the parameters for transmission statistics on the DSP.

DSP/TX: PK=65, SG=65, NS=65, DU=95, VO=80

Metric	Description
PK	Transmission (TX) packets
SG	Signaling packets
NS	Noise packets
DU	Transmission duration
VO	Voice transmission duration

DSP/RX: Receive Statistics

Receive statistics can show if a router is receiving packets properly. The statistics show the receipt of incoming voice, signaling, and comfort noise packets. In addition, the lengths of both the overall call and the voice portion are tracked. Errors such as bad sequence or protocol, or late packets are also shown.

The following example shows the parameters for receive statistics on the DSP.

DSP/RX: PK=65, SG=65, NS=65, RX=95, VO=80, BS=0, BP=0, LP=0

Metric	Description
PK	Transmission (TX) packets
SG	Signaling packets
NS	Noise packets
RX	Receive duration

Metric	Description
VO	Voice receive duration
BS	Bad sequence
BP	Bad protocol
LP	Late packets

DSP/PD: Playout Delay (Jitter Buffer)

Variable-length delays (also known as *jitter*) can cause a conversation to break and become unintelligible. Jitter is not usually a problem with PSTN calls because the bandwidth of calls is fixed and each call has a dedicated circuit for the duration of the call. However, in VoIP networks, data traffic might be bursty, and jitter from the packet network can become an issue. Packets from the same conversation can arrive at different interpacket intervals, especially during times of network congestion, which can disrupt the steady, even delivery needed for voice calls.

The adaptive jitter buffer adjusts the jitter buffer size and amount of playout delay during a call based on current network conditions An adaptive jitter buffer is a mechanism that seeks to make an intelligent trade-off between effective frame loss (leading to audible distortions or interruptions in speech) and overall delay (leading to difficulty in maintaining a smooth conversational rhythm).

The imposition of a small delay at the receiver (playout delay) can be used to 'soak up' jitter in the received packet stream. You must impose a delay sufficient to compensate for the jitter, minimizing the chance of an audible frame drop, but still short enough to minimize the adverse effects of delay on the conversation.

In fixed mode, the playout delay of the de-jitter buffer is a fixed or constant quantity regardless of the jitter observed on the connection.

The following example shows the parameters for playout delay statistics on the DSP.

DSP/PD: CU=65, MI=65, MA=65, CO=-1825458605, IJ=0

Metric	Description
CU	Current (instantaneous) size of the playout delay imposed by the adaptive jitter buffer. A packet arriving 'on-time' can expect to spend this much time (in ms) waiting in the jitter buffer. A packet arriving late will spend less time in the buffer, but will still be played out at the correct time. A low value indicate small jitter, and hence low overall delay experienced by the users. High values indicate the need to compensate for high levels of jitter in the received packet stream, at the cost of high overall delay.
MI	Minimum value of CU observed over the course of this call.
MA	Maximum value of CU observed over the course of this call.

Metric	Description
CO	RTP clock offset. This is the difference (in clock 'ticks' of 0.125ms duration) between the initial values of the clocks in the sending and receiving endpoints, as communicated via RTCP. This is an internal value used in the calculation of round-trip-delay. This measurement is not a meaningful voice quality metric, as the offset is essentially a random number.
IJ	Interarrival jitter. Jitter metric J (ms) is calculated according to RRFC3550. Interarrival jitter is a smoothed, or average, jitter measure, valid over a very short time window. This measurement is included for standards compliance, and as a rough measure of network jitter. RFC3550 is not recommended for use in the provisioning of jitter buffers, as it represents an average value, whereas jitter buffers are driven by the peak jitter observed. Adaptive mode is preferred for this reason. Observing RFC3550 jitter J is always less than 10ms, for example, and does not imply that 10ms is an adequate playout delay setting for a fixed jitter buffer.

DSP/PE: Playout Error Statistics

Playout error statistics can track concealment based on predictive or interpolative algorithms. Silence concealment is shown. Additional error statistics, such as retroactive memory updates, buffer overflow, and endpoint errors are also provided.

The following example shows the parameters for playout error statistics on the DSP.

DSP/PE: PC=65, IC=65, SC=65, RM=95, BO=80, EE=0

Metric	Description
PC	Predictive concealment, cumulative duration (in ms)
IC	Interpolative concealment, cumulative duration (in ms)
SC	Silence concealment, cumulative duration (in ms)
RM	Retroactive memory update
BO	Buffer overflow, cumulative number buffer overflow errors.
EE	Number of talkspurt endpoint errors

DSP/LE: Level Statistics

Level statistics show various volume, pulse, noise, and echo measurements for incoming and outgoing calls. Tranmit and receive activity are also tracked.

The following example shows the parameters for level statistics on the DSP.

DSP/LE: TP=65, TX=65, RP=65, RM=95, BN=80, ER=0, AC=0, TA=0, RA=0

Metric	Description
TP	Transmission power in 0.1 dBm units.
TX	Transmission mean in 0.1 linear PCM units.
RP	Receive power in 0.1 dBm units.
RM	Receive mean in 0.1 linear PCM units.
BN	Background noise.
ER	Echo return loss (ERL) level.
AC	Acom level. Current ACOM level estimate is in 0.1 dB increments. The term ACOM is used in G.165, "General Characteristics of International Telephone Connections and International Telephone Circuits: Echo Cancellers." ACOM is the combined loss achieved by the echo canceller, which is the sum of the ERL, ERL enhancement, and nonlinear processing loss for the call.
TA	Current transmit activity.
RA	Current receive activity.

DSP/ER: Error Statistics

DSP error statistics track errors in dropped and control packets for both incoming and outgoing calls.

The following example shows the parameters for receive statistics on the DSP.

DSP/ER: RD=65, TD=65, RC=65, TC=95

Metric	Description
RD	Receive dropped
TD	Transmission dropped
RC	Receive control
TC	Transmission control

DSP/IC: ICPIF

The ICPIF statistic tracks the calculated planning impairment factor. This measure is for predicting a user’s perceptions of voice quality. This measure has mostly been superceded by the R-factor measurement.

The following example shows the parameters for ICPIF statistics on the DSP.

DSP/IC: IC=65

Metric	Description
IC	ICPIF value for measuring voice-quality.

DSP/EC : Endpoint Configuration

Endpoint configuration and provisioning statistics are settings controlled by the user rather than performance metrics. These statistics represent the effective settings of the endpoint as reported by the DSP, they are useful for debug and logging purposes because they capture the state of the endpoint.

The following example shows the parameters for the configuration of the VoIP endpoint:

DSP/EC: CI=1, FM=640, FP=2, VS=0, GT=1.0000, GR=1.0000, JD=2, JN=60, JM=40, JX=200

Metric	Description
CI	Codec ID. A string or number that identifies the voice codec which is currently used in the call.
FM	Frame size in milliseconds. Native frame size of the selected codec. An example of a frame size and codec combination is G.729a/30ms. For the G.711 codec, the frame size is a value that is provisioned by the user in the voice dial peer. For example, G.711 at 80 bytes gives 10 milliseconds per frame. G.711 at 240 bytes gives 30 milliseconds per frame.
FP	Frames per packet. Number of codec speech frames encapsulated into a single RTP packet. Typical values are 1, 2, and 3. Packing low frames per packet results in low efficiency of IP bandwidth usage. The trade-off is low delays and high robustness of the network. High frames-per-packet provisioning is a tradeoff of speech quality in favor of reduced IP overhead cost.
VS	VAD enabled flag. VAD is enabled when VS has a value of one. This value results in compression of silent periods leading to reduced or zero packets per second. VAD is disabled when VS has a value of zero. This value results in the transmission of continuous packets per second irrespective of active or silent periods on the transmission path. Some codecs have built-in VAD functionality where VAD is always enabled, such as G.729br8, G.723ar63, or G.723ar5. An external VAD circuit is used for codecs that do not have a built-in VAD.
GT	Transmission gain factor (linear). This factor is the digital gain multiplier applied to transmission on the signal path from the PSTN toward the network. The value is applied at the echo canceller *Sout* port. A gain factor less than one indicates a loss pad. The digital gain is applied to 16-bit linear PCM samples. The samples are independent and add to any analog gain settings associated with analog voice ports.
GR	Reception gain factor (linear). This factor is the digital gain multiplier applied to the reception on the signal path from the network toward the PSTN. The multiplier is applied at the echo canceller *Rin* port. A gain factor less than one indicates a loss pad. The digital gain is applied to 16-bit linear PCM samples. The samples are independent and add to any analog gain settings associated with analog voice ports.

Metric	Description
JD	Jitter buffer mode • Adaptive mode = 1 • Fixed mode (no timestamps) = 2 • Fixed mode (with timestamps) = 3 • Fixed mode (with passthrough) = 4
JN	Jitter buffer nominal playout delay. Size of the jitter buffer in milliseconds. An adaptive jitter buffer tries to make the playout delay equal to the nominal (desired) delay when the observed jitter is small enough to allow this adjustment. For a fixed-mode jitter buffer, the nominal setting is the constant playout delay itself.
JM	Minimum playout delay. Minimum playout delay setting for an adaptive-mode jitter buffer. The playout delay never goes below the minimum playout setting, even if the observed jitter is zero. This setting is not used for a fixed-mode jitter buffer because the playout delay is fixed and constant at the nominal setting.
JX	Maximum playout delay. Sets the limit for increasing the playout delay of an adaptive-mode jitter buffer. An adaptive buffer increases when the jitter is higher than the instantaneous playout delay value. JX represents the largest allowable playout delay.

DDSP/KF : MOS/K-Factor Statistics

K-factor is an endpoint mean opinion score (MOS) estimation algorithm defined in ITU standard P.VTQ. This alogrithm is a general estimator and is used to estimate the mean value of a perceptual evaluation of speech quality (PESQ) population for a specific impairment pattern.

ITU standard P.862 defines and describes the PESQ as an objective method for end-to-end speech quality assessment of narrow band telephone networks and speech codecs.

MOS is a term that relates to the output of a well designed listening experiment. All MOS experiments use a five point PESQ scale as defined in ITU standard P.862.1. The MOS estimate is a number that is inversely proportional to frame loss density. Clarity decreases as more frames are lost or discarded at the receiving end.

K-factor represents the following:

- A weighted estimate of average user annoyance due to distortions caused by effective packet loss, such as dropouts and warbles.
- Does not estimate the impact of delay-related impairments, such as echo.
- An estimate of listening quality (MOS-LQO) rather than conversational quality (MOS-CQO), and measurements of average user annoyance range from 1 (poor voice quality) to 5 (very good voice quality).
- The algorithm is trained or conditioned by speech samples from numerous speech databases where each training sentence or network condition associated with a P.862.1 value has a duration of eight seconds. For more accurate scores, k-factor estimates are generated for every eight seconds of active speech.

K-factor and other MOS estimators are considered to be secondary or derived statistics because they warn a network operator of frame loss only after the problem becomes significant. Packet counts, concealment ratios, and concealment second counters are primary statistics because they alert the network operator before network impairment has an audible impact or is visible through MOS.

The following example shows the parameters for the k-factor configuration:

DSP/KF: KF=4.4001, AV=4.4001, MI=4.4001, BS=4.4001, NB=0, FL=0, NW=15, VR=99.88

Metric	Description
KF	K-factor MOS-LQO estimate (instantaneous). Estimate of the MOS score of the last eight seconds of speech on the reception signal path on a scale of 1 (very poor) to 5 (very good). If VAD is active, the MOS calculation is suspended during periods of received silence to avoid inflation of MOS scores for calls with high silence fractions.
AV	Average k-factor score. Running average of scores observed since the beginning of a call. One k-factor score is produced for every eight seconds of active speech.
MI	Minimum k-factor score. Minimum score observed since the beginning of a call, and represents the worst sounding eight second interval.
BS	Baseline (maximum) k-factor score. K-factor score that can be obtained for the provisioned codec. Under conditions of zero loss (perfect transmission), the average P.862.1 score for the given codec. For example, G.711 has a baseline score of ~4.4 under perfect conditions. G.729a has a baseline score of about 3.7. G.711 has better clarity than G.729a. Under conditions of packet loss, MOS scores can only get worse than the baseline (decrease).
NB	Number of bursts. Number of burst loss events after starting a call. A burst loss is a contiguous run of concealment events of length greater than one. For frames-per-packet (DSP/EC:FP) greater than one, every packet loss event will be counted as a burst.
FL	Average frame loss count. Total number of frame losses and concealment events observed after starting a call. The ratio of FL/NB provides the mean burst length in frames. The total concealment duration of the call is provided in the parameter *DSP/CS: CT*.
NW	Number of windows. Total number of k-factor windows observed after starting a call. The number of windows is directly proportional to the duration of a call.
VR	Version ID. Version number that identifies a specific k-factor MOS score.

DSP/CS: Concealment Statistics

The concealment statistics metrics are measures of effective packet/frame loss or concealment events. Concealment measures the overall impact of network impairments of voice quality. Packets can be late, lost, or corrupted for many reasons. Even without packet loss, a small difference in the clock frequency of the sending and terminating devices can lead to jitter buffer over- or underflows at the receiver. The result of this loss is the same: the audio decoder asks the jitter buffer for a new frame, and the jitter buffer is unable to oblige. The codec is momentarily 'starved,' and it must generate a fake or 'spoofed' or 'concealment' frame of output audio. This potentially audible occurance is the result of these various network- and device-related impairments.

The following example shows the parameters for concealment:

DSP/CS: CR=0.0000, AV=0.0000, MX=0.0000, CT=0l, TT=123000l, CS=0, SC=0, TS=50, DC=0

Metric	Description
CR	Concealment ratio (instantaneous). An interval-based average concealment rate which is the ratio of concealment time over speech time for the last three seconds of active speech. When VAD is enabled, calculation of the concealment ratio is suspended during periods of silence. During this suspension, it may take more than three seconds for a new value to be generated.
AV	Average k-factor score. Running average of scores observed since the beginning of a call.
MX	Maximum CR. The maximum concealment ratio observed after starting a call.
CT	Total concealment time in milliseconds. The total duration of time during which concealment is observed after starting a call that is being received.
TT	Total speech time in milliseconds. The duration of time during which active speech is observed after starting a call that is being received.
OK	Ok seconds. The duration of time, in seconds, during which no concealment is observed.
CS	Concealed seconds. The duration of time during which some concealment is observed.
SC	Severely concealed seconds. The duration of time during which a significant amount of concealment is observed. If the concealment observed is usually greater than fifty milliseconds or approximately five percent, it is possible that the speech is not very audible.
TS	Concealment threshold in milliseconds (ms). The threshold used to determine a second as severely concealed. The threshold for concealed seconds is 0 ms, and for severely concealed seconds the threshold is 50 ms.
DC	Dead Connection Detector (not supported).

DSP/RF: R-Factor Statistics

The R-factor helps in planning voice transmission. In ITU standards G.107 and G.113, the R-factor is defined as:

R = Ro - Is - Id - Ie-eff + A

- Ro is based on the signal to noise ratio.
- Is is the simultaneous impairment factor and includes the overall loudness rating.
- Id is the delay impairment factor and includes talker (Idte) and listener (Idle) echos, and delays (Idd).
- Ie-eff is the equipment impairment factor and includes packet losses and the types of codecs.
- A is the advantage factor.

The following example shows the parameters for the R-factor:

DSP/RF: ML=0.0000, MC=0, R1=0, R2=0, IF=0, ID=1, IE=0, BL=5, R0=94

Metric	Description
ML	R-factor MOS listening quality objective. Reflects only packet-loss and codec-related impairments and does not include delay effects.
MC	R-factor MOS-CQE. MOS Conversational Quality score derived from R-factor score R2 according to the curve fit of G.107 annex B. This score includes a delay impairment contribution, in addition to codec and packet loss effects.
R1	R-factor for LQ profile1. R-factor score R1 = R0 – Ie_eff.
R2	R-factor for LQ profile2. R-factor score R2 = R0 – Ie_eff - Idd.
IF	Effective codec impairment (Ie_eff) . Includes effect of codec type, codec packet loss robustness, and cumulative concealment ratio.
ID	Delay factors . This parameter is the delay impairment factor and includes talker (Idte) and listener (Idle) echos, and delays (Idd).
IE	Codec baseline score (Ie). The tabulated baseline codec impairment factor.
BL	Codec baseline (Bpl). The packet loss robustness factor for the codec being used.
R0	Nominal value for R0 (default). The nominal value at which the signal-to-noise ratio is considered nominal.

DSP/UC: User Concealment Statistics

The user concealment statistics show the length and threshold level of the concealment.

The following example shows the parameters for user concealment

DSP/UC: U1=0, U2=0, T1=32, T2=46:

Metric	Description
U1	User concealment seconds 1 count (UCS1)
U2	User concealment seconds 2 count (UCS2)
T1	UCS1 threshold in ms
T2	UCS2 threshold in ms

DSP/DL: Delay Statistics

The delay statistics show the length of the delay for round trip and end system measurements.

The following example shows the parameters for delay statistics

DSP/DL: RT=45, ED=5

Metric	Description
RT	Round trip delay.
ED	End system delay.

Statistics Measured by the IP SLAs RTP-Based VoIP Operation

The IP Service Level Agreements (SLAs) Real-Time Transport Protocol (RTP)-based Voice over IP (VoIP) Operation feature provides the capability to set up and schedule a test call and use Voice gateway digital signal processors (DSPs) to gather network performance-related statistics for the call. Available statistical measurements for VoIP networks include jitter, frame loss, Mean Opinion Score for Conversational Quality (MOS-CQ), and Mean Opinion Score for Listening Quality (MOS-LQ).

The IP SLAs RTP-based VoIP operation provides an enhanced capability to measure voice quality by using DSP-based calculations to determine MOS scores. For customer scenarios where the destination gateway does not have DSP hardware, statistical information is gathered only from the DSP of the source gateway. In this case, the RTP data stream is looped back from the destination to the source gateway.

For detailed information about IP SLAs RTP-based VoIP operation, see the *IP SLAs RTP-Based VoIP Operation* feature document.

The statistics gathered by the IP SLAs RTP-based VoIP operation vary depending on the type of DSP module (see Table 3 and Table 4).

Table 3 Statistics Gathered by the RTP-Based VoIP Operation for c549 DSPs

Statistics	Description
Interarrival jitter (destination-to-source and source-to-destination)	Interarrival jitter is the mean deviation (smoothed absolute value) of the difference in packet spacing for a pair of packets. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement. For more information about interarrival jitter, see RFC 3550 (*RTP: A Transport Protocol for Real-Time Applications*).
Estimated R factor (destination-to-source and source-to-destination)	Estimated transmission rating factor R. This value is based on one-way transmission delay and standard default values. No values are obtained from the DSP to calculate the estimated transmission rating factor R. For more information about the estimated R factor, see International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendation G.107 (*The E-model, a computational model for use in transmission planning*).
MOS-CQ (destination-to-source and source-to-destination)	Mean Opinion Score for Conversational Quality. This value is obtained by converting the estimated R factor to Mean Opinion Score (MOS) using ITU-T Recommendation G.107 conversion tables. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement.
Round-trip time (RTT) latency	Round-trip time latency for an RTP packet to travel from the source to the destination and back to the source.
Packet loss (destination-to-source and source-to-destination)	Number of packets lost. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement.
Packets missing in action (source-to-destination)	Number of missing packets. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement.
One-way latency (destination-to-source and source-to-destination)	Average, minimum, and maximum latency values. These values are measured by sending RTP packets to IP SLAs Responder. The RTP data stream is then looped back from the destination to the source gateway.

Table 4 Statistics Gathered by the RTP-Based VoIP Operation for c5510 DSPs

Statistics	Description
Interarrival jitter (destination-to-source and source-to-destination)	Interarrival jitter is the mean deviation (smoothed absolute value) of the difference in packet spacing for a pair of packets. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement. For more information about interarrival jitter, see RFC 3550 (*RTP: A Transport Protocol for Real-Time Applications*).
Estimated R factor (destination-to-source and source-to-destination)	Estimated transmission rating factor R. This value is based on one-way transmission delay and standard default values, as well as values obtained from the DSP. For more information about the estimated R factor, see International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendation G.107 (*The E-model, a computational model for use in transmission planning*).
MOS-CQ (destination-to-source and source-to-destination)	Mean Opinion Score for Conversational Quality. This value is obtained by conversion of the estimated R factor to Mean Opinion Score (MOS) using ITU-T Recommendation G.107 conversion tables. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement.
Round-trip time (RTT) latency	Round-trip time latency for an RTP packet to travel from the source to the destination and back to the source.
Packet loss (destination-to-source and source-to-destination)	Number of packets lost. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement.
Packets missing in action (source-to-destination)	Number of missing packets. The source-to-destination value is measured by sending RTP packets to the IP SLAs Responder. No values are obtained from the DSP for this measurement.
One-way latency (destination-to-source and source-to-destination)	Average, minimum, and maximum latency values. These values are measured by sending RTP packets to IP SLAs Responder. The RTP data stream is then looped back from the destination to the source gateway.
Frame loss (destination-to-source)	Number of DSP frame loss events. A frame loss can occur due to such events as packet loss, late packets, or a jitter buffer error.
MOS-LQ (destination-to-source)	Mean Opinion Score for Listening Quality.

Vendor Specific Attributes

For information about VSAs Supported by Cisco voice products, see “VSAs Supported by Cisco Voice Products.”

Call Agent Support

DSP voice quality metrics can be used with Cisco MGCP call agents. A *call agent* (or *media gateway controller*) and *softswitch* are industry standard terms used to describe the network element that provides call control functionality to telephony and packet networks. The DSP voice quality metrics can report statistics for the following call agents:

- Cisco PGW 2200, page 22
- Cisco Unified CallManager, page 23

Cisco PGW 2200

The Cisco PGW 2200 in “call control mode” functions as a call agent or softswitch.

A PSTN gateway provides the interface between traditional SS7 networks or non-SS7 networks and networks based on Media Gateway Control Protocol (MGCP), H.323, and Session Initiation Protocol (SIP), including signaling, call control, and time-division multiplexing/IP (TDM/IP) gateway functions. The Cisco PGW 2200, coupled with Cisco media gateways, functions as a PSTN gateway.

Caution There is a significant performance degradation on the Cisco PGW 2200 if all connected gateways have the DSP Voice Quality Metrics for MGCP feature enabled.
Enabling voice quality statistics on the gateway should only be performed by Cisco personnel.

The Cisco PGW 2200, in either signaling mode or call control mode, provides a robust, carrier-class interface between the PSTN and IP-based networks. Interworking with Cisco media gateways, the Cisco PGW 2200 supports a multitude of applications, including the following:

- International and national transit networks
- Dial access
- Application service provider (ASP) termination
- Managed business voice applications
- Managed voice virtual private networks (VPNs)
- PSTN access for hosted and managed IP telephony
- Residential voice applications
- PSTN access for voice over broadband networks
- Network clearinghouse applications
- Centralized routing and billing for clearinghouse of IP-based networks

Cisco Unified CallManager

Cisco Unified CallManager serves as the software-based call-processing component of the Cisco Unified Communications family of products. A wide range of Cisco Media Convergence Servers provides high-availability server platforms for Cisco Unified CallManager call processing, services, and applications.

The Cisco Unified CallManager system extends enterprise telephony features and functions to packet telephony network devices such as IP phones, media processing devices, voice-over-IP (VoIP) gateways, and multimedia applications. Additional data, voice, and video services, such as unified messaging, multimedia conferencing, collaborative contact centers, and interactive multimedia response systems, interact through Cisco Unified CallManager open telephony application programming interface (API).

Cisco Unified CallManager provides signaling and call control services to Cisco integrated telephony applications as well as third-party applications. Cisco Unified CallManager performs the following primary functions:

- Call processing
- Signaling and device control
- Dial plan administration
- Phone feature administration
- Directory services
- Operations, administration, maintenance, and provisioning (OAM&P)
- Programming interface to external voice-processing applications such as Cisco IP Communicator, Cisco Unified IP Interactive Voice Response (IP IVR), and Cisco Unified CallManager Attendant Console

How to Configure DSP Voice-Quality Statistics in DLCX Messages

This section contains procedures for configuring the DSP Voice-Quality Statistics in DLCX Messages feature.

- Configuring DSP Voice-Quality Statistics in DLCX Messages, page 23 (required)
- Verifying DSP Voice-Quality Statistics in DLCX Messages, page 24 (optional)
- Troubleshooting DSP Voice-Quality Statistics in DLCX Messages, page 26 (optional)

Configuring DSP Voice-Quality Statistics in DLCX Messages

To configure voice-quality statistics reporting for MGCP, use the following commands beginning in user EXEC mode.

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **mgcp voice-quality-stats**

4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Router> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Router# configure terminal`	Enters global configuration mode.
Step 3	**mgcp voice-quality-stats[priority**<1|2>**]|**[all] **Example:** `Router(config)# mgcp voice-quality-stats 1`	Enables voice-quality statistics reporting for MGCP. The following parameters are sent by default, if the *priority* or *all* keywords are not used: DSP/TX, DSP/RX, DSP/PD, DSP/PE, DSP/LE, DSP/ER, DSP/IC. **Priority 1** parameters are: DSP/TX, DSP/RX, DSP/PD, DSP/LE, DSP/EC, DSP/CS, DSP/DL. **Priority 2** parameters are: DSP/PE, DSP/ER, DSP/IC, DSP/KF, DSP/RF, DSP/UC. Using **priority 2** is similar to using the **all** keyword where the output shows the following parameters: DSP/TX, DSP/RX, DSP/PD, DSP/PE, DSP/LE, DSP/ER, DSP/IC, DSP/EC, DSP/KF, DSP/CS, DSP/RF, DSP/UC, DSP/DL.
Step 4	**end** **Example:** `Router(config)# end`	Completes the configuration.

Verifying DSP Voice-Quality Statistics in DLCX Messages

Use the following **show** commands to check your configuration:

SUMMARY STEPS

1. **show call active voice compact**
2. **show call active voice brief**
3. **show call history voice brief**

Step 1 Obtain the call ID by entering the **show call active voice compact** command in privileged EXEC mode:

```
Router# show call active voice compact
```

```
G<id> A/O FAX T<sec> Codec type Peer Address IP R<ip>:<udp>
Total call-legs: 2
G11D6 ORG T187 g729r8 TELE P
G11D6 ORG T0 g729r8 VOIP P 10.32.1.21:19324
```

Step 2 Check the status of active calls using the call ID obtained from the **show call active voice brief** command:

```
Router# show call active voice brief id 11D6

<ID>: <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
delay:<last>/<min>/<max>ms <codec>
MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
last <buf event time>s dur:<Min>/<Max>s
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
Tele <int>: tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l> i/o:<l>/<l> dBm
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
speeds(bps): local <rx>/<tx> remote <rx>/<tx>
Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
bw: <req>/<act> codec: <audio>/<video>
tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>


Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 0
MGCP call-legs: 1
Total call-legs: 2
11D6 : 37530hs.1 +0 pid:0 Originate active
dur 00:03:21 tx:1472/29003 rx:1405/27682
Tele 6/4:15 (1): tx:201530/37000/0ms g729r8 noise:-65 acom:90 i/0:-87/-24 dBm

11D6 : 37531hs.1 +-1 pid:0 Originate connecting
dur 00:00:00 tx:1403/27642 rx:1472/29003
IP 10.32.1.21:19324 rtt:0ms pl:36000/0ms lost:0/0/0 delay:100/90/110ms g729r8

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 0
MGCP call-legs: 1
Total call-legs: 2
```

Step 3 Verify your configuration with the **show call history voice brief** command:

```
Router# show call history voice brief

<ID>: <start>hs.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
  dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes> <disc-cause>(<text>)
 IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
  delay:<last>/<min>/<max>ms <codec>
  MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
   last <buf event time>s dur:<Min>/<Max>s
 FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
 ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
  sig:<on/off> <codec> (payload size)
 Telephony <int>: tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm acom:<lvl>dBm
```

Troubleshooting DSP Voice-Quality Statistics in DLCX Messages

Use the **debug mgcp packets** command and keyword to display statistics reported in the DLCX message generated at the end of the call. The following is sample debug output:

```
Router# debug mgcp packets

DLCX 311216 s6/ds1-4/1@as5400a MGCP 0.1
C: 48A4B
I: 2
R:
S:
X: 4BFAF
*May 5 10:20:51.643: send_mgcp_msg, MGCP Packet sent to 10.31.1.200:2427 --->
*May 5 10:20:51.643: 250 311216 OK
P: PS=1469, OS=28943, PR=1518, OR=29923, PL=0, JI=100, LA=0
DSP/TX: PK=1448, SG=0, NS=23, DU=206450, VO=39000
DSP/RX: PK=1449, SG=0, CF=23, RX=206450, VO=38000, BS=0, BP=0, LP=0
DSP/PD: CU=100, MI=90, MA=110, CO=69352809, IJ=0
DSP/PE: PC=0, IC=0, SC=0, RM=6, BO=0, EE=0
DSP/LE: TP=-24, TX=-440, RP=-87, RM=-870, BN=0, ER=50, AC=90, TA=-24, RA=-87
DSP/ER: RD=0, TD=0, RC=0, TC=0
DSP/IC: IC=0
```

Additional References

The following sections provide references related to the <<Feature Name>> feature.

Related Documents

Related Topic	Document Title
How to configure QoS for Cisco features.	*Cisco IOS Quality of Service Solutions Configuration Guide*
Cisco IOS Release 12.4 mainline roadmap	*Cisco IOS Release 12.4 Configuration Guides and Command References*
How to configure your Cisco router or access server to support voice, video, and fax applications.	*Cisco IOS Voice Configuration Library*
How to use Cisco IOS commands to support voice, video, and fax applications.	*Cisco IOS Voice Command Reference, Release 12.4*
Cisco MGC documentation index	Cisco Media Gateway Controllers
How to configure MGCP	*Configuring Media Gateway Control Protocol and Related Protocols*
How to configure the digital signal processor farm	*Configuring Enhanced Conferencing and Transcoding for Voice Gateway Routers*
How to configure QoS for voice applications.	*Configuring Quality of Service for Voice*
How to configure voice ports	*Configuring Voice Ports*, Release 12.3
Enabling basic management protocols on Cisco access platforms	*Enabling Management Protocols: NTP, SNMP, and Syslog*

Standards

Standard	Title
International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendation G.107	The E-model, a computational model for use in transmission planning

MIBs

MIB	MIBs Link
No new or modified MIBs are supported by this feature, and support for existing MIBs has not been modified by this feature.	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

RFCs

RFC	Title
RFC 3550	RTP: A Transport Protocol for Real-Time Applications

Technical Assistance

Description	Link
The Cisco Technical Support & Documentation website contains thousands of pages of searchable technical content, including links to products, technologies, solutions, technical tips, tools, and technical documentation. Registered Cisco.com users can log in from this page to access even more content.	http://www.cisco.com/techsupport

Command Reference

This section documents modified commands only.

For voice quality stats, some existing CLI are modified as follows:

- **gw-accounting**

The following show commands are modified as follows:

- **show call active voice**
- **show call history voice**

gw-accounting

To enable the accounting method for collecting call detail records, use the **gw-accounting** command in global configuration mode. To disable the accounting method, use the **no** form of this command.

Cisco IOS Release 12.4(11)XW and Later Releases

gw-accounting {**aaa** | **syslog** [**stats**] }

no gw-accounting {**aaa** | **syslog** [**stats**]}

Cisco IOS Release 12.2(11)T and Later Releases

gw-accounting {**aaa** | **syslog**}

no gw-accounting {**aaa** | **syslog**}

Cisco IOS Release 12.2(8)T and Earlier Releases

gw-accounting {**h323** [**vsa**] | **syslog** | **voip**}

no gw-accounting {**h323** [**vsa**] | **syslog** | **voip**}

Syntax Description

aaa	Enables accounting through the AAA system and sends call detail records to the RADIUS server in the form of vendor-specific attributes (VSAs).
syslog	Enables the system logging facility to output accounting information in the form of a system log message.
stats	Enables voice quality statistics to be sent to the system log.
h323	Enables standard H.323 accounting using Internet Engineering Task Force (IETF) RADIUS attributes.
vsa	(Optional) Enables H.323 accounting using RADIUS VSAs.
voip	Enables generic gateway-specific accounting.

Command Default

No accounting method is enabled.

Command Modes

Global configuration

Command History

Release	Modification
11.3(6)NA2	This command was introduced on the following platforms: Cisco 2600 series, Cisco 3600 series, and Cisco AS5300.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T. The **vsa** keyword was added.
12.1(1)T	The **voip** keyword was added.
12.2(2)XA	This command was implemented on the Cisco AS5350 and Cisco AS5400.

Release	Modification
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T and was implemented on the Cisco 7200 series.
12.2(11)T	The **h323**, **vsa**, and **voip** keywords were replaced by the **aaa** keyword.
12.4(11)XW	The **stats** keyword was added.

Usage Guidelines

This command enables you to output accounting data in one of three ways:

Using RADIUS Vendor-Specific Attributes

The IETF draft standard specifies a method for communicating vendor-specific information between the network access server and the RADIUS server by using the vendor-specific attribute (VSA) attribute 26. VSAs allow vendors to support their own extended attributes that are not suitable for general use. The Cisco RADIUS implementation supports one vendor-specific option using the format recommended in the specification. The Cisco vendor ID is 9, and the supported option has vendor-type 1, which is named "cisco-avpair." The value is a string of the format:

```
protocol: attribute sep value *
```

"Protocol" is a value of the Cisco "protocol" attribute for a particular type of authorization. "Attribute" and "value" are an appropriate attribute-value (AV) pair defined in the Cisco TACACS+ specification, and "sep" is "=" for mandatory attributes and "*" for optional attributes. This allows the full set of features available for TACACS+ authorization to also be used for RADIUS. For a list of VSA fields and their ASCII values, see the *Cisco IOS Security Configuration Guide* for your Cisco IOS release.

Use the **gw-accounting aaa** command to configure the VSA method of applying H.323 gateway-specific accounting.

Note Releases earlier than Cisco IOS Release 12.2(11)T use the **gw-accounting h323 vsa** command.

Overloading the Acct-Session-ID field

Attributes that cannot be mapped to standard RADIUS are packed into the Acct-Session-ID field as ASCII strings separated by the character "/". The Acct-Session-ID attribute is defined to contain the RADIUS account session ID, which is a unique identifier that links accounting records associated with the same login session for a user. To support additional fields, the following string format is defined for this field:

```
<session id>/<call leg setup time>/<gateway id>/<connection id>/<call origin>/
<call type>/<connect time>/<disconnect time>/<disconnect cause>/<remote ip address>
```

Table 5 shows the field attributes that are used with the overloaded session-ID method and a brief description of each.

Table 5 Field Attributes in Overloaded Acct-Session-ID

Field Attribute	Description
Session-Id	Standard RADIUS account session ID.
Setup-Time	Q.931 setup time for this connection in Network Time Protocol (NTP) format: hour, minutes, seconds, milliseconds, time zone, day of week, month, day of month, and year.

Table 5 Field Attributes in Overloaded Acct-Session-ID (continued)

Field Attribute	Description
Gateway-Id	Name of the underlying gateway in the form “gateway.domain_name.”
Call-Origin	Origin of the call relative to the gateway. Possible values are **originate** and **answer**.
Call-Type	Call leg type. Possible values are **telephony** and **VoIP**.
Connection-Id	Unique global identifier used to correlate call legs that belong to the same end-to-end call. The field consists of 4 long words (128 bits). Each long word is displayed as a hexadecimal value and is separated by a space character.
Connect-Time	Q.931 connect time for this call leg in NTP format.
Disconnect-Time	Q.931 disconnect time for this call leg in NTP format.
Disconnect-Cause	Reason that a call was taken offline as defined in the Q.931 specification.
Remote-Ip-Address	Address of the remote gateway port where the call is connected.

Because of the limited size of the Acct-Session-ID string, it is not possible to include many information elements in it. Therefore, this feature supports only a limited set of accounting information elements.

Use the **attribute acct-session-id overloaded** command to configure the overloaded session ID method of applying H.323 gateway-specific accounting.

Note Releases earlier than Cisco IOS Release 12.2(11)T use the **gw-accounting h323** command.

Using syslog Records

The syslog accounting option exports the information elements associated with each call leg through a system log message, which can be captured by a syslog daemon on the network. The syslog output consists of the following:

```
<server timestamp> <gateway id> <message number> : <message label> : <list of AV pairs>
```

Table 6 lists the syslog message fields.

Table 6 Syslog Message Output Fields

Field	Description
server timestamp	Time stamp created by the server when it receives the message to log.
gateway id	Name of the gateway that sends the message.
message number	Number assigned to the message by the gateway.
message label	String used to identify the message category.
list of AV pairs	String that consists of <attribute name> <attribute value> pairs separated by commas.

Use the **gw-accounting syslog** command to configure the syslog record method of gathering H.323 accounting data.

If you enable both **aaa** and **syslog** simultaneously, call detail records are generated in both methods.

Examples

The following example shows accounting using RADIUS VSA attributes:

```
gw-accounting aaa
```

The following example shows basic accounting using the syslog method:

```
gw-accounting syslog
```

Related Commands

Command	Description
attribute acct-session-id overloaded	Overloads the acct-session-id attribute with call detail records.
dial-peer voice	Enters dial-peer configuration mode and specifies the method of voice-related encapsulation.
inband-alerting	Enables in-band alerting so that the originating gateway can open an early media path.
radius-server vsa send	Enables the voice gateway to recognize and use VSAs.

show call active voice

To display call information for voice calls in progress, use the **show call active voice** command in user EXEC or privileged EXEC mode.

show call active voice [**brief** [**called-number** *number* | **calling-number** *number*]] | **compact** [**duration** {**less** *seconds* | **more** *seconds*}] | **echo-canceller** *call-id* | **id** *identifier* | **media-inactive** [**called-number** *number* | **calling-number** *number*] | [**long-dur-call**] | [**redirect** {**rtpvt** | **tbct**}] | [**stats**]

Syntax Description

brief	(Optional) Displays a truncated version of call information.
called-number *number*	(Optional) Displays a specific called number pattern.
calling-number *number*	(Optional) Displays a specific calling number pattern.
compact	(Optional) Displays a compact version of call information.
duration	(Optional) Displays active calls that are longer or shorter than a specified *seconds* value. The argument and keywords are as follows: • **less**—Displays calls shorter than the *seconds* value. • **more**—Displays calls longer than the *seconds* value. • *seconds*—Elapsed time, in seconds. Range is from 1 to 2147483647. There is no default value.
echo-canceller *call-id*	(Optional) Displays information about the state of the extended echo canceller (EC). To query the echo state, you need to know the hex ID in advance. To find the hex ID, enter the **show call active voice brief** command or use the **show voice call status** command. Range is from 0 to FFFFFFFF.
id *identifier*	(Optional) Displays only the call with the specified *identifier*. Range is a hex value from 1 to FFFF.
media-inactive	(Optional) Displays information about inactive media that have been detected.
long-call-dur	(Optional) Displays information only for calls with long_duration detected and notified.
redirect	(Optional) Displays information about active calls that are being redirected using Release-to-Pivot (RTPvt) or Two B-Channel Transfer (TBCT). The keywords are as follows: • **rtpvt**—Displays information about RTPvt calls. • **tbct**—Displays information about TBCT calls.
stats	(Optional) Displays informationabout DSP voice quality metrics.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 2600 series and Cisco 3600 series.
12.0(3)XG	Support for VoFR was added.
12.0(4)XJ	This command was implemented for store-and-forward fax on the Cisco AS5300.
12.0(4)T	This command was implemented on the Cisco 7200 series.
12.0(7)XK	This command was implemented on the Cisco MC3810.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(3)T	This command was implemented for modem pass-through over VoIP on the Cisco AS5300.
12.1(5)XM	This command was implemented on the Cisco AS5800.
12.1(5)XM2	The command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support is not included for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(11)T	Support was added for the Cisco AS5300, Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(13)T	The **echo-canceller** keyword was added. The command output was modified with an extra reflector location when the extended EC is present; the largest reflector location is shown.
12.3(1)	The **redirect** keyword was added.
12.3(4)T	The **called-number**, **calling-number**, and **media-inactive** keywords were added.
12.3(14)T	New output relating to Skinny Client Control Protocol (SCCP), SCCP Telephony Control Application (STCAPP), and modem pass-through traffic was added.
12.4(2)T	The LocalHostname display field was added to the VoIP call leg record and command ouput was enhanced to display modem relay physical layer and error correction protocols.
12.4(4)T	The **long-dur-call** keyword was added to generate **show** command output that displays only calls with long_duration detected and notified.
12.4(11)XW	The **stats** keyw ord was added.

Usage Guidelines

Use this command to display the contents of the active call table. This command displays information about call times, dial peers, connections, quality of service, and other status and statistical information for voice calls currently connected through the router.

When the extended EC is present, the **show call active voice** command displays the contents of the Ditech EC_CHAN_CTRL structure. Table 7 contains names and descriptions of the fields in the EC_CHAN_CTRL structure. Table 7 also provides a listing of the information types associated with this command.

Table 7 EC_CHAN_CTRL Field Descriptions

Symbol	Field	Description
BYP0	Channel bypass	1 = Transparent bypass; EC is disabled. 0 = Cancel; EC is enabled.
TAIL3	Max tail	0 = 24 milliseconds. 1 = 32 milliseconds. 2 = 48 milliseconds. 3 = 64 milliseconds. **Note** This field should be set just greater than the anticipated worst round-trip tail delay.
REC3	Residual echo control	0 = Cancel only; echo is the result of linear processing; no nonlinear processing is applied. 1 = Suppress residual; residual echo is zeroed; simple nonlinear processing is applied (you might experience “dead air” when talking). 2 = Reserved. 3 = Generate comfort noise (default).
FRZ0	h-register hold	1 = Freezes h-register; used for testing.
HZ0	h-register clear	Sending the channel command with this bit set clears the h-register.
TD3	Modem tone disable	0 = Ignore 2100 Hz modem answer tone. 1 = G.164 mode (bypass canceller if 2100 Hz tone). 2 = R. 3 = G.165 mode (bypass canceller for phase reversing tone only).
ERL0	Echo return loss	0 = 6 decibel (dB). 1 = 3 dB. 2 = 0 dB. 3 = R. Worst echo return loss (ERL) situation in which canceller still works.
HLC1	High level compensation	0 = No attenuation. 1 = 6 dB if clipped. On loud circuits, the received direction can be attenuated 6 dB if clipping is observed.
R0	Reserved	Must be set to 0 to ensure compatibility with future releases.

Use the **show call active voice redirect** command to monitor any active calls that implement RTPvt or TBCT.

When a call is no longer active, its record is stored. You can display the record by using the **show call history voice** command.

Examples

The following is sample output from the **show call active voice** command for modem relay traffic:

```
Router# show call active voice

Modem Relay Local Rx Speed=0 bps
Modem Relay Local Tx Speed=0 bps
Modem Relay Remote Rx Speed=0 bps
Modem Relay Remote Tx Speed=0 bps
Modem Relay Phy Layer Protocol=v34
Modem Relay Ec Layer Protocol=v14
SPRTInfoFramesReceived=0
SPRTInfoTFramesSent=0
SPRTInfoTFramesResent=0
SPRTXidFramesReceived=0
SPRTXidFramesSent=0
SPRTTotalInfoBytesReceived=0
SPRTTotalInfoBytesSent=0
SPRTPacketDrops=0
```

The following is sample output from the **show call active voice** command:

```
Router# show call active voice

Total call-legs:2

 GENERIC:
SetupTime=7587246 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=7587506
CallDuration=00:00:11
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=2
TransmitPackets=101
TransmitBytes=1991
ReceivePackets=550
ReceiveBytes=11000
VOIP:
ConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
RemoteIPAddress=172.29.248.111
RemoteUDPPort=17394
RoundTripDelay=4 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=cisco
SessionTarget=
OnTimeRvPlayout=10300
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
```

```
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=70 ms
LoWaterPlayoutDelay=69 ms
ReceiveDelay=69 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
SignalingType=ext-signal
CallerName=
CallerIDBlocked=False
 GENERIC:
SetupTime=7587246 ms
Index=2
PeerAddress=133001
PeerSubAddress=
PeerId=133001
PeerIfIndex=8
LogicalIfIndex=7
ConnectTime=7587505
CallDuration=00:00:56
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=2
TransmitPackets=2801
TransmitBytes=56020
ReceivePackets=162
ReceiveBytes=3192
TELE:
ConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
IncomingConnectionId=[0x7F8D82A4 0x928E11D5 0x8094FCFB 0x1C38F0FA]
TxDuration=56030 ms
VoiceTxDuration=3210 ms
FaxTxDuration=0 ms
CoderTypeRate=g729r8
NoiseLevel=-44
ACOMLevel=-13
OutSignalLevel=-45
InSignalLevel=-45
InfoActivity=2
ERLLevel=7
EchoCancellerMaxReflector=64
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=Fals
```

Note Table 7 on page 35 describes the significant fields shown in the display.

The following is sample output from the **show call active voice** command for voice traffic over call-agent controlled call legs. Note that call legs for SCCP telephony endpoints, that is, phones controlled by STCAPP, are displayed under the "Call agent controlled call-legs." (“SCCP call-legs” displays call legs for devices that are not telephony endpoints, for example, transcoding and conferencing).

```
Router# show call active voice

Telephony call-legs: 2
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 2
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 4

 GENERIC:
SetupTime=1557650 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=999100
PeerIfIndex=14
LogicalIfIndex=10
ConnectTime=1562040 ms
CallDuration=00:01:01 sec
CallState=4
CallOrigin=2
ChargedUnits=0
InfoType=speech
TransmitPackets=3101
TransmitBytes=519564
ReceivePackets=3094
ReceiveBytes=494572
TELE:
ConnectionId=[0x11B1860C 0x22D711D7 0x8014E4D4 0x8FD15327]
IncomingConnectionId=[0x11B1860C 0x22D711D7 0x8014E4D4 0x8FD15327]
CallID=25
TxDuration=59670 ms
VoiceTxDuration=59670 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-12
ACOMLevel=22
OutSignalLevel=-12
InSignalLevel=-11
InfoActivity=1
ERLLevel=22
EchoCancellerMaxReflector=2
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x80
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x80
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
DSPIdentifier=1/1:1

 GENERIC:
SetupTime=1559430 ms
Index=1
```

```
PeerAddress=7702
PeerSubAddress=
PeerId=999100
PeerIfIndex=14
LogicalIfIndex=11
ConnectTime=1562020 ms
CallDuration=00:01:03 sec
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=3151
TransmitBytes=528900
ReceivePackets=3158
ReceiveBytes=503876
TELE:
ConnectionId=[0x0 0x0 0x0 0x0]
IncomingConnectionId=[0x0 0x0 0x0 0x0]
CallID=26
TxDuration=60815 ms
VoiceTxDuration=60815 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-12
ACOMLevel=28
OutSignalLevel=-12
InSignalLevel=-11
InfoActivity=1
ERLLevel=28
EchoCancellerMaxReflector=2
SessionTarget=
ImgPages=0
CallerName=
CallerIDBlocked=False
AlertTimepoint=1559430 ms
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=7701
TranslatedCallingOctet=0x0
TranslatedCalledNumber=7702
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
GwOutpulsedCalledNumber=7702
GwOutpulsedCalledOctet3=0x0
GwOutpulsedCallingNumber=7701
GwOutpulsedCallingOctet3=0x0
GwOutpulsedCallingOctet3a=0x0
DSPIdentifier=1/1:2

 GENERIC:
SetupTime=1562040 ms
Index=1
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=0 ms
CallDuration=00:00:00 sec
```

```
CallState=2
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=3215
TransmitBytes=512996
ReceivePackets=3208
ReceiveBytes=512812
VOIP:
ConnectionId[0x0 0x0 0x0 0x0]
IncomingConnectionId[0x0 0x0 0x0 0x0]
CallID=27
RemoteIPAddress=10.10.0.0
RemoteUDPPort=17718
RemoteSignallingIPAddress=10.10.0.0
RemoteSignallingPort=0
RemoteMediaIPAddress=10.2.6.10
RemoteMediaPort=17718
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=FALSE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=other
ProtocolCallId=
SessionTarget=
OnTimeRvPlayout=60640
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=105 ms
LoWaterPlayoutDelay=105 ms
TxPakNumber=3040
TxSignalPak=0
TxComfortNoisePak=0
TxDuration=60815
TxVoiceDuration=60815
RxPakNumber=3035
RxSignalPak=0
RxDuration=0
TxVoiceDuration=60690
VoiceRxDuration=60640
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
PlayDelayCurrent=105
PlayDelayMin=105
PlayDelayMax=105
PlayDelayClockOffset=-1662143961
PlayDelayJitter=0
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=-12
```

```
InSignalLevel=-11
LevelTxPowerMean=0
LevelRxPowerMean=-115
LevelBgNoise=0
ERLLevel=28
ACOMLevel=28
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
PlayoutMode = undefined
PlayoutInitialDelay=0 ms
ReceiveDelay=105 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
Media Setting=flow-around

Modem passthrough signaling method is nse:
Buffer Fill Events = 0
Buffer Drain Events = 0
Percent Packet Loss = 0
Consecutive-packets-lost Events = 0
Corrected packet-loss Events = 0
Last Buffer Drain/Fill Event = 0sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec

CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
Username=

 GENERIC:
SetupTime=1562040 ms
Index=2
PeerAddress=
PeerSubAddress=
PeerId=0
PeerIfIndex=0
LogicalIfIndex=0
ConnectTime=0 ms
CallDuration=00:00:00 sec
CallState=2
CallOrigin=1
ChargedUnits=0
InfoType=speech
```

```
TransmitPackets=3380
TransmitBytes=540332
ReceivePackets=3386
ReceiveBytes=540356
VOIP:
ConnectionId[0x0 0x0 0x0 0x0]
IncomingConnectionId[0x0 0x0 0x0 0x0]
CallID=28
RemoteIPAddress=10.0.0.0
RemoteUDPPort=18630
RemoteSignallingIPAddress=10.10.0.0
RemoteSignallingPort=0
RemoteMediaIPAddress=10.2.6.10
RemoteMediaPort=18630
RoundTripDelay=0 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=FALSE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=other
ProtocolCallId=
SessionTarget=
OnTimeRvPlayout=63120
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=105 ms
LoWaterPlayoutDelay=105 ms
TxPakNumber=3158
TxSignalPak=0
TxComfortNoisePak=0
TxDuration=63165
TxVoiceDuration=63165
RxPakNumber=3164
RxSignalPak=0
RxDuration=0
TxVoiceDuration=63165
VoiceRxDuration=63120
RxOutOfSeq=0
RxLatePak=0
RxEarlyPak=0
PlayDelayCurrent=105
PlayDelayMin=105
PlayDelayMax=105
PlayDelayClockOffset=957554296
PlayDelayJitter=0
PlayErrPredictive=0
PlayErrInterpolative=0
PlayErrSilence=0
PlayErrBufferOverFlow=0
PlayErrRetroactive=0
PlayErrTalkspurt=0
OutSignalLevel=-12
InSignalLevel=-11
LevelTxPowerMean=0
LevelRxPowerMean=-114
LevelBgNoise=0
```

```
ERLLevel=22
ACOMLevel=22
ErrRxDrop=0
ErrTxDrop=0
ErrTxControl=0
ErrRxControl=0
PlayoutMode = undefined
PlayoutInitialDelay=0 ms
ReceiveDelay=105 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
SRTP = off
VAD = disabled
CoderTypeRate=g711ulaw
CodecBytes=160
Media Setting=flow-around

Modem passthrough signaling method is nse:
Buffer Fill Events = 0
Buffer Drain Events = 0
Percent Packet Loss = 0
Consecutive-packets-lost Events = 0
Corrected packet-loss Events = 0
Last Buffer Drain/Fill Event = 0sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec

CallerName=
CallerIDBlocked=False
OriginalCallingNumber=
OriginalCallingOctet=0x0
OriginalCalledNumber=
OriginalCalledOctet=0x0
OriginalRedirectCalledNumber=
OriginalRedirectCalledOctet=0x0
TranslatedCallingNumber=
TranslatedCallingOctet=0x0
TranslatedCalledNumber=
TranslatedCalledOctet=0x0
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0x0
MediaInactiveDetected=no
MediaInactiveTimestamp=
MediaControlReceived=
Username=
Telephony call-legs: 2
SIP call-legs: 0
H323 call-legs: 0
Call agent controlled call-legs: 2
SCCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 4
```

The following is sample output from the **show call active voice** command for fax-relay traffic:

```
Router# show call active voice

Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 1
MGCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 1
```

```
 GENERIC:
SetupTime=1049400 ms
Index=2
PeerAddress=52930
PeerSubAddress=
PeerId=82
PeerIfIndex=222
LogicalIfIndex=0
ConnectTime=105105
CallDuration=00:00:59
CallState=4
CallOrigin=1
ChargedUnits=0
InfoType=10
TransmitPackets=1837
TransmitBytes=29764
ReceivePackets=261
ReceiveBytes=4079
VOIP:
ConnectionId[0xEB630F4B 0x9F5E11D7 0x8008CF18 0xB9C3632]
IncomingConnectionId[0xEB630F4B 0x9F5E11D7 0x8008CF18 0xB9C3632]
RemoteIPAddress=10.7.95.3
RemoteUDPPort=16610
RemoteSignallingIPAddress=10.7.95.3
RemoteSignallingPort=1720
RemoteMediaIPAddress=10.7.95.3
RemoteMediaPort=16610
RoundTripDelay=13 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=TRUE

SessionProtocol=cisco
ProtocolCallId=
SessionTarget=ipv4:10.7.95.3
OnTimeRvPlayout=1000
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=110 ms
LoWaterPlayoutDelay=70 ms
ReceiveDelay=70 ms
LostPackets=0
EarlyPackets=1
LatePackets=0
VAD = enabled
CoderTypeRate=t38
CodecBytes=40
Media Setting=flow-through
AlertTimepoint=104972
CallerName=
CallerIDBlocked=False
OriginalCallingNumber=4085550130
OriginalCallingOctet=0x0
OriginalCalledNumber=52930
OriginalCalledOctet=0xE9
OriginalRedirectCalledNumber=
```

```
OriginalRedirectCalledOctet=0x7F
TranslatedCallingNumber=4085550130
TranslatedCallingOctet=0x0
TranslatedCalledNumber=52930
TranslatedCalledOctet=0xE9
TranslatedRedirectCalledNumber=
TranslatedRedirectCalledOctet=0xFF
GwReceivedCalledNumber=52930
GwReceivedCalledOctet3=0xE9
GwOutpulsedCalledNumber=52930
GwOutpulsedCalledOctet3=0xE9
GwReceivedCallingNumber=4085452930
GwReceivedCallingOctet3=0x0
GwReceivedCallingOctet3a=0x80
GwOutpulsedCallingNumber=4085550130
GwOutpulsedCallingOctet3=0x0
GwOutpulsedCallingOctet3a=0x80
Username=
FaxRelayMaxJitterBufDepth  = 0 ms
FaxRelayJitterBufOverFlow = 0
FaxRelayHSmodulation = 0
FaxRelayNumberOfPages = 0
Telephony call-legs: 0
SIP call-legs: 0
H323 call-legs: 1
MGCP call-legs: 0
Multicast call-legs: 0
Total call-legs: 1
```

Note Table 7 on page 35 and Table 8 on page 47 describe fields in the display.

The following is sample output from the **show call active voice brief** command:

```
Router# show call active voice brief

<ID>: <CallID> <start>hs.<index> +<connect> pid:<peer_id> <dir> <addr> <state>
dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes>
IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
delay:<last>/<min>/<max>ms <codec>
media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>
long_duration_call_detected:<y/n> long duration call duration:n/a timestamp:n/a
MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
last <buf event time>s dur:<Min>/<Max>s
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
Tele <int> (callID) [channel_id] tx:<tot>/<v>/<fax>ms <codec> noise:<l> acom:<l>
i/o:<l>/<l> dBm
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
speeds(bps): local <rx>/<tx> remote <rx>/<tx>
Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
bw: <req>/<act> codec: <audio>/<video>
tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>

Total call-legs:2
1269 :7587246hs.1 +260 pid:0 Answer  active
 dur 00:07:14 tx:590/11550 rx:21721/434420
IP 172.29.248.111:17394 rtt:3ms pl:431850/0ms lost:0/0/0 dela
y:69/69/70ms g729r8
```

```
1269 :7587246hs.2 +259 pid:133001 Originate 133001 active
 dur 00:07:14 tx:21717/434340 rx:590/11550
 Tele 1/0:1 (2):tx:434350/11640/0ms g729r8 noise:-44 acom:-19
i/0:-45/-45 dBm
```

The following is an example of the **show call active voice** command using the **echo-canceller** keyword. The number 9 represents the hexadecimal ID of an active voice call.

```
Router# show call active voice echo-canceller 9

ACOM=-65  ERL=45
Echo canceller control words=6C 0
Bypass=OFF  Tail=64  Residual ecan=Comfort noise
Freeze=OFF  Modem tone disable=Ignore 2100Hz tone
Worst ERL=6  High level compensation=OFF
Max amplitude reflector (in msec)=5
Ecan version = 8180
```

The following is sample output from the **show call active voice echo-canceller** command for a call with a hexadecimal ID of 10:

```
Router# show call active voice echo-canceller 10

ACOM=-15  ERL=7
Echo canceller control words=6C 0
Bypass=OFF  Tail=64  Residual ecan=Comfort noise
Freeze=OFF  Modem tone disable=Ignore 2100Hz tone
Worst ERL=6  High level compensation=OFF
Max amplitude reflector (in msec)=64
```

The call ID number (which is 10 in the previous example) changes with every new active call. When an active call is up, you must enter the **show call active voice brief** command to obtain the call ID number. The call ID must be converted to hex if you want to use the **show call active voice echo-canceller** *x* command (*x* = call ID converted to hex).

The following are call ID examples converted to hex (generally incremented by 2):

Decimal	Hex
2	2
4	4
6	6
8	8
10	A
12	C

Alternatively, you can use the **show voice call status** command to obtain the call ID. The call ID output is already in hex form when you use this command:

```
Router# show voice call status

CallID      CID  ccVdb      Port      DSP/Ch  Called #   Codec    Dial-peers
0x1         11CE 0x02407B20 1:0.1     1/1     1000       g711ulaw 2000/1000
```

The following is sample output from the **show call active voice redirect** command using the **tbct** keyword:

```
Router# show call active voice redirect tbct

TBCT:
        Maximum no. of TBCT calls allowed:No limit
        Maximum TBCT call duration:No limit


Total number TBCT calls currently being monitored = 1

ctrl name=T1-2/0, tag=13, call-ids=(7, 8), start_time=*00:12:25.985 UTC Mon Mar 1 1993
```

Table 8 describes the significant fields shown in the **show call active voice redirect** display.

Table 8 show call active voice redirect Field Descriptions

Field	Description
Maximum no. of TBCT calls allowed	Maximum number of calls that can use TBCT as defined by the **tbct max calls** command.
Maximum TBCT call duration	Maximum length allowed for a TBCT call as defined by the **tbct max call-duration** command.
Total number TBCT calls currently being monitored	Total number of active TBCT calls.
ctrl name	Name of the T1 controller where the call originated.
tag	Call tag number that identifies the call.
call-ids	Numbers that uniquely identify the call legs.
start_time	Time, in hours, minutes, and seconds, when the redirected call began.

Related Commands

Command	Description
show call active fax	Displays call information for fax transmissions that are in progress.
show call history	Displays the call history table.
show call-router routes	Displays the dynamic routes in the cache of the BE.
show call-router status	Displays the Annex G BE status.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice call status	Displays the call status for voice ports on the Cisco router or concentrator.
show voice port	Displays configuration information about a specific voice port.

show call history voice

To display the call history table for voice calls, use the **show call history voice** command in user EXEC or privileged EXEC mode.

show call history voice [**brief** [**id** *identifier*] | **compact** [**duration** {**less** *seconds* | **more** *seconds* }] | **id** *identifier* | **last** *number* | **redirect** {**rtpvt** | **tbct**}] | **[stats**]

Syntax Description

brief	(Optional) Displays a truncated version of the call history table.
id *identifier*	(Optional) Displays only the call with the specified *identifier*. Range is from 1 to FFFF.
compact	(Optional) Displays a compact version of the call history table.
duration *seconds*	(Optional) Displays history information for calls that are longer or shorter than a specified *seconds*. The arguments and keywords are as follows: • **less**—Displays calls shorter than the *seconds* value. • **more—**Displays calls longer than the *seconds* value. • *seconds*—Elapsed time, in seconds. Range is from 1 to 2147483647.
last *number*	(Optional) Displays the last calls connected, where the number of calls that appear is defined by the *number* argument. Range is from 1 to 100.
redirect	(Optional) Displays information about calls that were redirected using Release-to-Pivot (RTPvt) or Two B-Channel Transfer (TBCT). The keywords are as follows: • **rtpvt**—Displays information about RTPvt calls. • **tbct**—Displays information about TBCT calls.
stats	(Optional) Displays informationabout DSP voice quality metrics.

Command Modes

User EXEC
Privileged EXEC

Command History

Release	Modification
11.3(1)T	This command was introduced on the Cisco 3600 series.
12.0(3)XG	Support was added for Voice over Frame Relay (VoFR) on the Cisco 2600 series and Cisco 3600 series.
12.0(4)XJ	This command was modified for store-and-forward fax.
12.0(4)T	The **brief** keyword was added, and the command was implemented on the Cisco 7200 series.
12.0(5)XK	This command was implemented on Cisco MC3810.
12.0(7)XK	The **brief** keyword was implemented on the Cisco MC3810.
12.0(7)T	This command was integrated into Cisco IOS Release 12.0(7)T.
12.1(2)T	This command was integrated into Cisco IOS Release 12.1(2)T.
12.1(5)XM	This command was implemented on the Cisco AS5800.

Release	Modification
12.1(5)XM2	This command was implemented on the Cisco AS5350 and Cisco AS5400.
12.2(2)XA	The output of this command was modified to indicate whether the call in question has been established using Annex E.
12.2(4)T	This command was integrated into Cisco IOS Release 12.2(4)T.
12.2(2)XB1	This command was implemented on the Cisco AS5850.
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. Support is not included for the Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.2(11)T	Support was added for Cisco AS5350, Cisco AS5400, Cisco AS5800, and Cisco AS5850.
12.2(13)T	The ReleaseSource field was added to the Field Description table, and the word **record** was deleted from the command name.
12.3(1)	The **redirect** keyword was added.
12.4(2)T	The LocalHostname display field was added to the VoIP call leg record.
12.4(11)XW	The **stats** keyword was added.

Usage Guidelines

This command displays a call-history table that contains a list of voice calls connected through the router in descending time order. The maximum number of calls contained in the table can be set to a number from 0 to 500 using the **dial-control-mib** command in global configuration mode. The default maximum number of table entries is 50. Each call record is aged out of the table after a configurable number of minutes has elapsed. The timer value is also specified by the **dial-control-mib** command. The default timer value is 15 minutes.

You can display subsets of the call history table by using specific keywords. To display the last calls connected through this router, use the **last** keyword, and define the number of calls to be displayed with the *number* argument.

To display a truncated version of the call history table, use the **brief** keyword.

Use the **show call active voice redirect** command to review records for calls that implemented RTPvt or TBCT.

When a call is active, you can display its statistics by using the **show call active voice** command.

Examples

The following is sample output from the **show call history voice** command:

```
Router# show call history voice

GENERIC:
SetupTime=104648 ms
Index=1
PeerAddress=55240
PeerSubAddress=
PeerId=2
PeerIfIndex=105
LogicalIfIndex=0
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=104964
DisconectTime=143329
CallDuration=00:06:23
```

```
CallOrigin=1
ChargedUnits=0
InfoType=speech
TransmitPackets=37668
TransmitBytes=6157536
ReceivePackets=37717
ReceiveBytes=6158452
VOIP:
ConnectionId[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
RemoteIPAddress=1.14.82.14
RemoteUDPPort=18202
RoundTripDelay=2 ms
SelectedQoS=best-effort
tx_DtmfRelay=inband-voice
FastConnect=TRUE

SessionProtocol=cisco
SessionTarget=ipv4:1.14.82.14
OnTimeRvPlayout=40
GapFillWithSilence=0 ms
GapFillWithPrediction=0 ms
GapFillWithInterpolation=0 ms
GapFillWithRedundancy=0 ms
HiWaterPlayoutDelay=67 ms
LoWaterPlayoutDelay=67 ms
ReceiveDelay=67 ms
LostPackets=0 ms
EarlyPackets=0 ms
LatePackets=0 ms
VAD = enabled
CoderTypeRate=g729r8
CodecBytes=20
cvVoIPCallHistoryIcpif=0
SignalingType=cas

Modem passthrough signaling method is nse
Buffer Fill Events = 0
Buffer Drain Events = 0
Percent Packet Loss = 0
Consecutive-packets-lost Events = 0
Corrected packet-loss Events = 0
Last Buffer Drain/Fill Event = 373sec
Time between Buffer Drain/Fills = Min 0sec Max 0sec

GENERIC:
SetupTime=104443 ms
Index=2
PeerAddress=50110
PeerSubAddress=
PeerId=100
PeerIfIndex=104
LogicalIfIndex=10
DisconnectCause=10
DisconnectText=normal call clearing.
ConnectTime=104964
DisconectTime=143330
CallDuration=00:06:23
CallOrigin=2
ChargedUnits=0
InfoType=speech
TransmitPackets=37717
TransmitBytes=5706436
ReceivePackets=37668
ReceiveBytes=6609552
```

```
TELE:
ConnectionId=[0x4B091A27 0x3EDD0003 0x0 0xFEFD4]
TxDuration=375300 ms
VoiceTxDuration=375300 ms
FaxTxDuration=0 ms
CoderTypeRate=g711ulaw
NoiseLevel=-75
ACOMLevel=11
SessionTarget=
ImgPages=0
```

The following example from a Cisco AS5350 router displays a sample of voice call history records showing release source information:

```
Router# show call history voice

Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 1
Total call-legs: 2

GENERIC:
SetupTime=85975291 ms
.
.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975335
DisconnectTime=85979339
CallDuration=00:00:40
CallOrigin=1
ReleaseSource=1
.

.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975335
DisconnectTime=85979339
CallDuration=00:00:40
CallOrigin=1
ReleaseSource=1
.
.
.
VOIP:
ConnectionId[0x2868AD84 0x375B11D4 0x8012F7A5 0x74DE971E]
.
.
.
GENERIC:
SetupTime=85975290 ms
.
.
.
DisconnectCause=10
DisconnectText=normal call clearing (16)
ConnectTime=85975336
DisconnectTime=85979340
CallDuration=00:00:40
CallOrigin=2
ReleaseSource=1
```

```
.
.
.
TELE:
ConnectionId=[0x2868AD84 0x375B11D4 0x8012F7A5 0x74DE971E]
```

The following is sample output from the **show call history voice brief** command:

```
Router# show call history voice brief

<ID>: <CallID> <start>hs.<index> +<connect> +<disc> pid:<peer_id> <direction> <addr>
dur hh:mm:ss tx:<packets>/<bytes> rx:<packets>/<bytes> <disc-cause>(<text>)
IP <ip>:<udp> rtt:<time>ms pl:<play>/<gap>ms lost:<lost>/<early>/<late>
delay:<last>/<min>/<max>ms <codec>
media inactive detected:<y/n> media cntrl rcvd:<y/n> timestamp:<time>
MODEMPASS <method> buf:<fills>/<drains> loss <overall%> <multipkt>/<corrected>
last <buf event time>s dur:<Min>/<Max>s
FR <protocol> [int dlci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
ATM <protocol> [int vpi/vci cid] vad:<y/n> dtmf:<y/n> seq:<y/n>
<codec> (payload size)
Telephony <int> (callID) [channel_id] tx:<tot>/<voice>/<fax>ms <codec> noise:<lvl>dBm
acom:<lvl>dBm
MODEMRELAY info:<rcvd>/<sent>/<resent> xid:<rcvd>/<sent> total:<rcvd>/<sent>/<drops>
disc:<cause code>
speeds(bps): local <rx>/<tx> remote <rx>/<tx>
Proxy <ip>:<audio udp>,<video udp>,<tcp0>,<tcp1>,<tcp2>,<tcp3> endpt: <type>/<manf>
bw: <req>/<act> codec: <audio>/<video>
tx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
rx: <audio pkts>/<audio bytes>,<video pkts>/<video bytes>,<t120 pkts>/<t120 bytes>
```

The following is sample output from the **show call history voice redirect** command:

```
Router# show call history voice redirect tbct

index=2, xfr=tbct-notify, status=redirect_success, start_time=*00:12:25.981 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=13
index=3, xfr=tbct-notify, status=redirect_success, start_time=*00:12:25.981 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=13
index=4, xfr=tbct-notify, status=redirect_success, start_time=*00:13:07.091 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=12
index=5, xfr=tbct-notify, status=redirect_success, start_time=*00:13:07.091 UTC Mon Mar 1
1993, ctrl name=T1-2/0,  tag=12

Number of call-legs redirected using tbct with notify:4
```

Table 9 describes the significant fields shown in the **show call history voice redirect tbct** display.

Table 9 show call history voice redirect Field Descriptions

Field	Description
index	Index number of the record in the history file.
xfr	Whether TBCT or TBCT with notify has been invoked.
status	Status of the redirect request.
start_time	Time, in hours, minutes, and seconds when the redirected call began.
ctrl name	Name of the T1 controller where the call originated.

Table 9 show call history voice redirect Field Descriptions (continued)

Field	Description
tag	Call tag number that identifies the call.
Number of call-legs redirected using tbct with notify	Total number of call legs that were redirected using TBCT with notify.

Related Commands

Command	Description
dial-control-mib	Set the maximum number of calls contained in the table.
show call active fax	Displays call information for fax transmissions that are in progress.
show call active voice	Displays call information for voice calls that are in progress.
show call history fax	Displays the call history table for fax transmissions.
show dial-peer voice	Displays configuration information for dial peers.
show num-exp	Displays how the number expansions are configured in VoIP.
show voice port	Displays configuration information about a specific voice port.

Feature Information for DSP Voice Quality Metrics for MGCP

Table 10 lists the release history for this feature.

Not all commands may be available in your Cisco IOS software release. For release information about a specific command, see the command reference documentation.

Use Cisco Feature Navigator to find information about platform support and software image support. Cisco Feature Navigator enables you to determine which Cisco IOS and Catalyst OS software images support a specific software release, feature set, or platform. To access Cisco Feature Navigator, go to http://www.cisco.com/go/cfn. An account on Cisco.com is not required.

Note Table 10 lists only the Cisco IOS software release that introduced support for a given feature in a given Cisco IOS software release train. Unless noted otherwise, subsequent releases of that Cisco IOS software release train also support that feature.

Table 10 Feature Information for DSP Voice Quality Metrics for MGCP

Feature Name	Releases	Feature Information
DSP Voice-Quality Statistics in DLCX Messages	12.3(3) 12.4(4)T	This feature provides a way to trace a Media Gateway Control Protocol (MGCP) call between a Cisco PGW 2200 and the Cisco IOS gateway by including the MGCP call ID and the DS0 and digital signal processor (DSP) channel ID in call-active and call-history records.
Voice Quality Enhancements	12.4(11)XW	The Voice Quality Enhancements feature exposes a rich set of voice quality metrics through the syslog and CLI interfaces.

Glossary

AAL2—ATM adaptation layer 2.

ASP—application service provider.

CA—call agent.

CAC—call admission control.

CAS—channel-associated signaling.

CDR—call detail record.

CLI—command-line-interface.

DCLX—MGCP Delete Connection message.

DSP—digital signal processor.

FTP—File Transfer Protocol.

NAS—network access server.

PVC—permanent virtual circuit.

RSVP—Resource Reservation Protocol.

RTCP—RTP Control Protocol. Protocol that monitors the QoS of an IPv6 RTP connection and conveys information about the ongoing session.

RTP—Real-Time Transport Protocol.

SRC—system resource check.

TDM—time-division multiplexing.

VPN—virtual private network.

WFQ—weighted fair queueing.

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