DSP Voice-Quality Statistics in DLCX Messages

The DSP Voice-Quality Statistics in DLCX Messages 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.

These voice quality statistics are sent as part of the MGCP Delete Connection (DLCX) message. By correlating an MGCP call on the Cisco PGW 2200 with the call record on the gateway, additional statistics from the DSP can be understood and debugged for problems related to voice quality.

Feature History for DSP Voice-Quality Statistics in DLCX Messages

Release	Modification
12.3(3)	This feature was introduced.
12.4(4)T	Introduced new voice quality parameters and modified the **mgcp voice-quality stats** command.

Finding Support Information for Platforms and Cisco IOS Software Images

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

Contents

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Prerequisites for DSP Voice-Quality Statistics in DLCX Messages

You must be using Cisco PGW 2200 version 9.4.1 or a later version with a patch level higher than CSCOgs008/CSCOnn008.

Information About DSP Voice-Quality Statistics in DLCX Messages

To configure the DSP Voice-Quality Statistics in DLCX Messages feature, you should understand the following concepts:

Cisco PGW 2200

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 Cisco PGW 2200 in "call control mode" functions as a call agent or softswitch.

Note All voice quality parameters upto Cisco IOS Release 12.4(4)T are supported only on the Cisco PGW 2200 call agent.

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 this 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

MGCP

MGCP defines the call control relationship between call agents (CAs) and VoIP gateways that translate audio signals to and from the packet network. The CAs are responsible for processing the calls.

An MGCP gateway handles the translation between audio signals and the packet network. The gateways interact with a CA, also called a media gateway controller (MGC), which performs signal and call processing on gateway calls. MGCP uses endpoints and connections to construct a call.

Endpoints are sources of, or destinations for data, and can be physical or logical locations in a device. Connections can be either point-to-point or multipoint. The gateway can be a Cisco router, access server, or cable modem, and the CA is a server from a third-party vendor.

Voice Quality Statistics

The Cisco PGW 2200 can capture voice quality statistics sent from MGCP-controlled media gateways and can propagate the statistics into the call detail records (CDRs) at the end of each call. The Cisco AS5x00 media gateways send voice quality statistics to the Cisco PGW 2200.

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 additional CPU load for each call. The additional CPU load 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. Polling for the voice quality statistics, level, and error parameters can be added. 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.

Table 1 Voice Quality Statistics for Cisco IOS Release 12.4(4)T

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

Quality of Service for Voice

The DSP Voice-Quality Statistics in DLCX Messages feature is part of Cisco quality of service (QoS) technologies. QoS is the ability of a network to provide better service to selected network traffic over various technologies, including Frame Relay, ATM, Ethernet and 802.1 networks, and SONET, as well as IP-routed networks that may use any or all of these underlying technologies.

QoS provides the following benefits:

- Control over bandwidth, equipment, and wide-area facilities—As an example, you can limit the bandwidth consumed over a backbone link by FTP or queueing of an important database access.
- More efficient use of network resources—Network analysis management and accounting tools enable you to know what your network is being used for and ensure that you are servicing the most important traffic to your business.
- Customized services—QoS enables Internet service providers (ISPs) to offer carefully customized grades of service differentiation to their customers.

- Coexistence of mission-critical applications—Cisco QoS technologies make certain that bandwidth and minimum delays required by time-sensitive multimedia and voice applications are available and that other applications using the link get their fair service without interfering with mission-critical traffic.
- Foundation for a fully integrated network—Cisco QoS technologies fully integrate a multimedia network, for example, by implementing weighted fair queueing (WFQ) to increase service predictability and IP precedence signaling to differentiate traffic. Also available is Resource Reservation Protocol (RSVP), which allows you to take advantage of dynamically signaled QoS.

To deliver QoS across a network that comprises heterogeneous technologies (for example, IP, ATM, LAN switches), the basic QoS architecture has three components:

- QoS within a single network element (for example, queueing, scheduling, and traffic shaping tools).
- QoS signaling techniques for coordinating end-to-end QoS between network elements.
- QoS policy, management, and accounting functions to control and administer end-to-end traffic across a network.

Voice Quality Parameters for Cisco IOS Release 12.4(4)T

Cisco IOS Release 12.4(4)T introduces these new voice quality parameters:

- DSP/EC : Endpoint Configuration
- DSP/KF : MOS/K-Factor Statistics
- DSP/CS: Concealment Statistics
- DSP/RF—R-Factor Statistics
- DSP/UC: User Concealment Statistics
- DSP/DL: Delay Statistics

DSP/EC : Endpoint Configuration

The following elements describe the configuration of the VoIP endpoint. They are provisioned by the user and are used to debug and log, because they capture the state of the endpoint.

- 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 lower frames per packet results in lower efficiency of IP bandwidth usage. The tradeoff is lower delays and higher robustness of the network.

- VS—VAD enabled flag

  VAD is enabled when VS has a value of one. It results in compression of silent periods leading to reduced or zero packets per second.

VAD is disabled when VS has a value of zero. It results in the transmission of continuous packets per second irrespective of active or silent periods on the transmission path.

- GT—Transmission gain factor (linear)

  Digital gain multiplier applied to transmission on the signal path from the PSTN toward the network. It is applied at the echo canceller *Sout* port. A gain factor less than one indicates a loss pad.

- GR—Reception gain factor (linear)

  Digital gain multiplier applied to reception on the signal path from the network toward the PSTN, applied at the echo canceller *Rin* port. A gain factor less than one indicates a loss pad.

- 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.

DSP/KF : MOS/K-Factor Statistics

K-factor is an endpoint MOS estimation algorithm defined in ITU standard P.VTQ. It is a general estimator and is used to estimate the mean value of a PESQ population for a specific impairment pattern.

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

Mean opinion score (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 inversely proportional to frame loss density. Clarity decreases as more frames are lost or discarded at the receiving end.

K-factor represents a weighted estimate of average user annoyance due to distortions caused by effective packet loss such as dropouts and warbles. It does not estimate the impact of delay-related impairments such as echo.

It is 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).

K-factor 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.

- KF—k-factor MOS-LQO estimate (instantaneous)

  Estimate of the MOS score of the last eight seconds of speech on the reception signal path. If VAD is active, the MOS calculation is suspended during periods of received silence to avoid inflation of MOS scores for calls with higher silence fractions.

- AV—Average k-factor score

  Running average of scores observed since the beginning of a call.

- 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.

- 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.

- 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

It measures packet (frame) loss and its effect on voice quality in an impaired network.

- CR—Concealment ratio (instantaneous)

  An interval-based average concealment rate, and 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 speech silence. During this suspension, it may take more than three seconds for a new value to be generated.

- AV—Average CR

  Average of all CR reports after starting a call.

- MX—Maximum CR

  The maximum concealment ratio observed after starting a call.

- CS—Concealed seconds

  The duration of time during which some concealment is observed.

- CT—Total concealment time in milliseconds

  The total duration of time during which concealment is observed after starting a call.
- TT—Total speech time in milliseconds

  The duration of time during which active speech is observed after starting a call.
- OK—Ok seconds

  The duration of time in seconds during which no 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 is 50 ms.

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.

- 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
- R1—R-factor LQ profile 1
- R2—R-factor LQ profile 2
- IF—Effective codec impairment (Ie_eff)
- ID—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—R0 (default)

  The nominal value at which the signal-to-noise ratio is considered nominal.

DSP/UC: User Concealment Statistics

- 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

- RT—Round trip delay
- ED—End system delay

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. Each procedure is identified as either required or optional.

- Configuring DSP Voice-Quality Statistics in DLCX Messages, page 9 (required)
- Verifying DSP Voice-Quality Statistics in DLCX Messages, page 10 (optional)
- Troubleshooting DSP Voice-Quality Statistics in DLCX Messages, page 12 (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
```

The following sections provide references related to the DSP Voice-Quality Statistics in DLCX Messages 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 12.3 mainline roadmap	*Cisco IOS Release 12.3 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*, Release 12.3
How to use Cisco IOS commands to support voice, video, and fax applications.	*Cisco IOS Voice, Video, and Fax Command Reference,* Release 12.3
Cisco MGC documentation index	Cisco Media Gateway Controllers
How to configure MGCP	*Configuring Media Gateway Control Protocol and Related Protocols*
How to configure QoS for voice applications.	*Configuring Quality of Service for Voice*
How to configure voice ports	*Configuring Voice Ports*, Release 12.2
Enabling basic management protocols on Cisco access platforms	*Enabling Management Protocols: NTP, SNMP, and Syslog*
Cisco IOS Release 12.3	Release notes index, Cisco IOS Release 12.3

Standards

Standards	Title
No new or modified standards are supported by this feature, and support for existing standards has not been modified by this feature.	—

MIBs

MIBs	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

RFCs	Title
No new or modified RFCs are supported by this feature, and support for existing RFCs has not been modified by this feature.	—

Technical Assistance

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

Command Reference

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

New Command

- **mgcp voice-quality-stats**

Modified Command

- **debug mgcp**

debug mgcp

To enable debug traces for Media Gateway Control Protocol (MGCP) errors, events, media, packets, parser, and call admission control (CAC), use the **debug mgcp** command in privileged EXEC mode. To disable debugging output, use the **no** form of this command.

debug mgcp [**all** | **errors** [**endpoint** *endpoint-name*] | **events** [**endpoint** *endpoint-name*] | **media** [**endpoint** *endpoint-name*] | **nas** | **packets** [**endpoint** *endpoint-name* | **input-hex**] | **parser** | **src** | **voipcac**]

no debug mgcp [**all** | **errors** | **events** | **media** | **nas** | **packets** | **parser** | **src** | **voipcac**]

Syntax Description

all	(Optional) Enables debugging traces for MGCP errors, events, media, packets, parser and builder, and CAC.
errors	(Optional) Enables debugging traces for MGCP errors.
endpoint *endpoint-name*	(Optional) Enables debugging traces for MGCP errors, events, media, or packets per endpoint.
events	(Optional) Enables debugging traces for MGCP events.
media	(Optional) Enables debugging traces for MGCP tone and signal events.
nas	(Optional) Enables debugging traces for MGCP network access server (NAS) (data) events.
packets	(Optional) Enables debugging traces for MGCP packets.
input-hex	(Optional) Enables debugging traces for MGCP input packets in hexadecimal values.
parser	(Optional) Enables debugging traces for MGCP parser and builder.
src	(Optional) Enables debugging traces for MGCP system resource check (SRC) CAC information.
voipcac	(Optional) Enables debugging traces for the VoIP CAC process at the MGCP application layer.

Command Modes

Privileged EXEC

Command History

Release	Modification
12.1(1)T	This command was introduced.
12.1(3)T	Additional information was displayed for the gateways.
12.1(5)XM	The output was modified to display parameters for the MGCP channel-associated signaling (CAS) PBX and ATM adaptation layer 2 (AAL2) permanent virtual circuit (PVC) features.
12.2(2)XA	The **media** keyword was added. The **endpoint** *endpoint-name* keyword and argument were added as options for the **errors**, **events**, **media**, and **packets** keywords. The **input-hex** keyword option was added for the **packets** keyword.
12.2(2)T	The output was further modified to display parameters for the MGCP CAS PBX and AAL2 PVC features.

Release	Modification
12.2(2)XB	The **nas** keyword and the **src** and **voipcac** keywords were added. (Refer to MGCP VoIP Call Admission Control in Cisco IOS Release 12.2(2)XB.)
12.2(8)T	This command was integrated into Cisco IOS Release 12.2(8)T. **Note** The **nas** keyword was not integrated into Cisco IOS Release 12.2(8)T.
12.2(11)T	The command was implemented on the Cisco AS5350, Cisco AS5400, and Cisco AS5850.
12.2(13)T	Support for this command was implemented in Cisco 7200 series.
12.3(3)	DSP statistics were reported in the MGCP Delete Connection (DLCX) message.

Usage Guidelines

There is always a performance penalty when using **debug** commands.

Examples

The following is sample output from the **debug mgcp errors**, **debug mgcp events**, **debug mgcp media**, **debug mgcp nas**, **debug mgcp packets**, **debug mgcp parser**, and **debug mgcp src** commands and keywords. The **debug mgcp all** command and keyword would show a compilation of all this output, including the **debug mgcp voipcac** command and keyword output. Note that using the **debug mgcp all** command and keyword may severely impact network performance.

The following is sample output from the debug mgcp errors command and keyword:

```
Router# debug mgcp errors

Unknown network interface type
```

The following is sample output from the debug mgcp events command and keyword:

```
Router# debug mgcp events

Media Gateway Control Protocol events debugging is on
Router#
1w1d: MGC stat - 10.19.184.65, total=44, succ=7, failed=21
1w1d: MGCP msg 1
1w1d: remove_old_under_specified_ack:
1w1d: MGC stat - 10.19.184.65, total=44, succ=8, failed=21
1w1d: updating lport with 2427setup_ipsocket: laddr=172.29.248.193, lport=2427,
faddr=10.19.184.65, fport=2427
1w1d: enqueue_ack: ackqhead=0, ackqtail=0, ackp=1DC1D38, msg=21A037C
```

The following is sample output from the **debug mgcp media** command and keyword:

```
Router# debug mgcp media

Media Gateway Control Protocol media events debugging is on
Router#
DYNAMIC payload type
DYNAMIC payload type
*Jan 1 03:02:13.159:mgcp_verify_supp_reqdet_ev
*Jan 1 03:02:13.159:mgcp_verify_supp_signal_ev
*Jan 1 03:02:13.159:process_request_ev- callp 635368FC, voice_if 6353C1F8
*Jan 1 03:02:13.159:process_detect_ev- callp 635368FC, voice_if 6353C1F8
*Jan 1 03:02:13.159:process_signal_ev- callp 635368FC, voice_ifp 6353C1F8
*Jan 1 03:02:13.159:mgcp_process_quarantine_mode- callp 635368FC, voice_if 6353C1F8
*Jan 1 03:02:13.159:mgcp_process_quarantine_mode- new q mode:process=0, loop=0
```

```
*Jan 1 03:02:13.179:process_deferred_request_events
*Jan 1 03:02:13.479:mgcp_verify_supp_reqdet_ev
*Jan 1 03:02:13.479:mgcp_verify_supp_signal_ev
*Jan 1 03:02:13.479:process_request_ev- callp 6353BCCC, voice_if 638C3094
*Jan 1 03:02:13.479:process_detect_ev- callp 6353BCCC, voice_if 638C3094
*Jan 1 03:02:13.479:process_signal_ev- callp 6353BCCC, voice_ifp 638C3094
*Jan 1 03:02:13.479:mgcp_process_quarantine_mode- callp 6353BCCC, voice_if 638C3094
*Jan 1 03:02:13.479:mgcp_process_quarantine_mode- new q mode:process=0, loop=0
*Jan 1 03:02:13.499:process_deferred_request_events
*Jan 1 03:02:13.827:mgcp_verify_supp_reqdet_ev
*Jan 1 03:02:13.827:mgcp_verify_supp_signal_ev
*Jan 1 03:02:13.827:process_request_ev- callp 635368FC, voice_if 6353C1F8
*Jan 1 03:02:13.827:process_detect_ev- callp 635368FC, voice_if 6353C1F8
*Jan 1 03:02:13.827:process_signal_ev- callp 635368FC, voice_ifp 6353C1F8
*Jan 1 03:02:13.827:mgcp_process_quarantine_mode- callp 635368FC, voice_if 6353C1F8
*Jan 1 03:02:13.827:mgcp_process_quarantine_mode- new q mode:process=0, loop=0
*Jan 1 03:02:13.831:process_deferred_request_events
*Jan 1 03:02:23.163:mgcp_cr_and_init_evt_node:$$$ the node pointer 63520B14
*Jan 1 03:02:23.163:mgcp_insert_node_to_preprocess_q:$$$enq to preprocess, qhead=63520B14,
qtail=63520B14, count 1, evtptr=63520B14
*Jan 1 03:02:23.479:mgcp_cr_and_init_evt_node:$$$ the node pointer 63520BA8
*Jan 1 03:02:23.479:mgcp_insert_node_to_preprocess_q:$$$enq to preprocess, qhead=63520BA8,
qtail=63520BA8, count 1, evtptr=63520BA8
```

The following is sample output for the **debug mgcp nas** command and keyword, with the **debug mgcp packets** command and keyword enabled as well:

```
Router# debug mgcp nas

Media Gateway Control Protocol nas pkg events debugging is on
Router# debug mgcp packets
Media Gateway Control Protocol packets debugging is on
Router#
01:49:14:MGCP Packet received -
CRCX 58 S7/DS1-0/23 MGCP 1.0
X:57
M:nas/data
C:3
L:b:64, nas/bt:modem, nas/cdn:3000, nas/cgn:1000
mgcp_parse_conn_mode :string past nas = data
mgcp_chq_nas_pkg:Full string:nas/bt:modem
mgcp_chq_nas_pkg:string past slash:bt
mgcp_chq_nas_pkg:string past colon:modem
mgcp_chq_nas_pkg:Full string:nas/cdn:3000
mgcp_chq_nas_pkg:string past slash:cdn
mgcp_chq_nas_pkg:string past colon:3000
mgcp_chq_nas_pkg:Full string:nas/cgn:1000
c5400#
mgcp_chq_nas_pkg:string past slash:cgn
mgcp_chq_nas_pkg:string past colon:1000
CHECK DATA CALL for S7/DS1-0/23
mgcpapp_xcsp_get_chan_cb -Found - Channel state Idle
CRCX Recv
mgcpapp_endpt_is_data:endpt S7/DS1-0/23, slot 7, port 0 chan 23
mgcpapp_data_call_hnd:mgcpapp_xcsp_get_chan_cb -Found - Channel state Idle
bw=64, bearer=E1,cdn=3000,cgn=1000
```

The following is sample output from the **debug mgcp packets** command and keyword:

```
Router# debug mgcp packets

Media Gateway Control Protocol packets debugging is on
Router#
1w1d: MGCP Packet received -
```

```
DLCX 408631346 * MGCP 0.1
1w1d: send_mgcp_msg, MGCP Packet sent --->
1w1d: 250 408631346
<---
```

The following is sample output from the **debug mgcp parser** command and keyword:

```
Router# debug mgcp parser

Media Gateway Control Protocol parser debugging is on
Router#
1w1d: -- mgcp_parse_packet() - call mgcp_parse_header
- mgcp_parse_header()- Request Verb FOUND DLCX
- mgcp_parse_packet() - out mgcp_parse_header
- SUCCESS: mgcp_parse_packet()- MGCP Header parsing was OK
- mgcp_val_mandatory_parms()
- SUCCESS: mgcp_parse_packet()- END of Parsing
1w1d: -- mgcp_build_packet()-
1w1d: - mgcp_estimate_msg_buf_length() - 87 bytes needed for header
- mgcp_estimate_msg_buf_length() - 87 bytes needed after checking parameter lines
- mgcp_estimate_msg_buf_length() - 87 bytes needed after checking SDP lines
- SUCCESS: MGCP message building OK
- SUCCESS: END of building
```

The following is sample output from the **debug mgcp src** command and keyword:

```
Router# debug mgcp src

Media Gateway Control Protocol System Resource Check CAC debugging is on
Router#
00:14:08: setup_indication: Set incoming_call flag=TRUE in voice_if
00:14:08: send_mgcp_msg, MGCP Packet sent --->
00:14:08: NTFY 11 aaln/S1/1@Router MGCP 0.1
N: emu@[1.4.173.1]:51665
X: 35
O: hd
<---
00:14:08: MGCP Packet received -
200 11 hello
00:14:08: MGCP Packet received -
RQNT 42 aaln/S1/1 MGCP 0.1
N: emu@[10.40.155.1]:51665
X: 41
R: D/[0-9*#T](d), hu
S: dl
D: (911|xxxx)
00:14:08: send_mgcp_msg, MGCP Packet sent --->
00:14:08: 200 42 OK
<---
00:14:12: send_mgcp_msg, MGCP Packet sent --->
00:14:12: NTFY 12 aaln/S1/1@Router MGCP 0.1
N: emu@[10.40.155.1]:51665
X: 41
O: D/2222
<---
00:14:12: MGCP Packet received -
200 12 phone-number ok
00:14:12: MGCP Packet received -
CRCX 44 aaln/S1/1 MGCP 0.1
N: emu@[10.40.155.1]:51665
C: 3
X: 43
R: hu(n)
M: recvonly
```

```
L: a:G.711u,p:5,e:off,s:off
00:14:12: mgcp_setup_conn_check_system_resource: System resource check successful
00:14:12: mgcp_voice_crcx: System resource is available
00:14:12: mgcp_set_call_counter_control: Incoming call with 1 network leg, flag=FALSE
00:14:12: send_mgcp_msg, MGCP Packet sent --->
00:14:12: 200 44
I: 4
v=0
o=- 4 0 IN IP4 10.20.120.1
s=Cisco SDP 0
c=IN IP4 10.20.120.1
t=0 0
m=audio 16404 RTP/AVP 0
<---
00:14:13: MGCP Packet received -
MDCX 48 aaln/S1/1 MGCP 0.1
N: emu@[10.40.155.1]:51665
C: 3
I: 4
X: 47
M: recvonly
R: hu
L: a:G.711u,p:5,e:off,s:off
v=0
o=- 4 0 IN IP4 1.4.120.3
s=Cisco SDP 0
c=IN IP4 10.4.110.3
t=0 0
m=audio 16384 RTP/AVP 0
00:14:13: mgcp_modify_conn_check_system_resource: System resource check successful
00:14:13: mgcp_modify_connection: System resource is available
00:14:13: send_mgcp_msg, MGCP Packet sent --->
00:14:13: 200 48 OK
<---
00:14:20: MGCP Packet received -
MDCX 52 aaln/S1/1 MGCP 0.1
N: emu@[10.40.155.1]:51665
C: 3
I: 4
X: 51
M: sendrecv
R: hu
L: a:G.711u,p:5,e:off,s:off
00:14:20: mgcp_modify_conn_check_system_resource: System resource check successful
00:14:20: mgcp_modify_connection: System resource is available
00:14:20: send_mgcp_msg, MGCP Packet sent --->
00:14:20: 200 52 OK
<---
00:14:34: MGCP Packet received -
DLCX 56 aaln/S1/1 MGCP 0.1
X: 55
N: emu@[10.40.155.1]:51665
C: 3
I: 4
R: hu
00:14:34: send_mgcp_msg, MGCP Packet sent --->
00:14:34: 250 56
P: PS=1382, OS=110180, PR=1378, OR=109936, PL=63484, JI=520, LA=2
<---
00:14:36: mgcp_reset_call_direction: Reseting incoming_call flag=FALSE in voice_if
00:14:36: send_mgcp_msg, MGCP Packet sent --->
00:14:36: NTFY 13 aaln/S1/1@tlkrgw1 MGCP 0.1
N: emu@[10.40.155.1]:51665
X: 55
```

```
O: hu
<---
```

The following example displays statistics reported in the DLCX message generated at the end of a call:

```
Router# debug mgcp packets

DLCX 311216 s6/ds1-4/1@as5400a MGCP 0.1
C: 48A4B
I: 2
R:
S:
X: 4BFAF
*May 5 10:38:51.643: send_mgcp_msg, MGCP Packet sent to 10.31.1.200:2427 --->
*May 5 10:38:51.643: 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
```

Table 2 describes the significant fields shown in the display.

Table 2 debug mgcp packets Field Descriptions

Field	Description
DSP/TX	DSP transmission (TX) statistics: • PK—Transmission (TX) packets • SG—Signaling packets • NS—Noise packets • DU—Transmission duration • VO—Voice transmission duration
DSP/RX	DSP receive (RX) statistics: • PK—Voice packets • SG—Signaling packets • CF—Comfort noise packets • RX—Receive duration • VO—Voice receive duration • BS—Bad sequence • BP—Bad protocol • LP—Late packets

Table 2 debug mgcp packets Field Descriptions (continued)

Field	Description
DSP/PD	DSP playout delay (PD) statistics: • CU—Current • MI—Minimum • MA—Maximum • CO—clock_offset • IJ—Interarrival jitter
DSP/PE	DSP playout error (PE) statistics: • PC—Predictive concealment • IC—Interpolative concealment • SC—Silence concealment • RM—Retroactive memory update • BO—Buffer overflow • EE—Number of talkspurt endpoint errors
DSP/LE	DSP level (LE) statistics: • 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 • TA—Current transmit activity • RA—Current receive activity
DSP/ER	DSP error statistics: • RD—Receive dropped • TD—Transmission dropped • RC—Receive control • TC—Transmission control
DSP/IC	ICPIF value for measuring voice-quality.

mgcp voice-quality-stats

To enable voice-quality statistics reporting for the Media Gateway Control Protocol (MGCP), use the **mgcp voice-quality-stats** command in global configuration mode. To turn off voice-quality statistics reporting, use the no form of this command.

mgcp voice-quality-stats [**priority**<*value*>] | [**all**]

no mgcp voice-quality-stats [**priority**<*value*>] | [**all**]

Syntax Description

priority<*value*>	Selects numeric parameters 1 or 2 to indicate priority.
all	Selects all VQ parameters.

Command Default

Voice-quality statistics reporting is turned off.

Command Modes

Global configuration

Command History

Release	Modification
12.3(3)	This command was introduced.
12.4(4)T	The **priority** and **all** keywords were introduced.

Usage Guidelines

- The request for digital signal processor (DSP) statistics is controlled by the RTP Control Protocol (RTCP) statistics polling interval. The polling interval is configurable by entering the **ip rtcp report interval** command. Statistics are polled every 5 seconds by default.

Note The Cisco PGW 2200 must have a patch the supports DSP statistics in order to collect data in the call detail records (CDRs).

- This command does not generate any output on the console; it adds additional quality statistics parameters in the MGCP Delete Connection (DLCX) ACK message that is sent to the call agent.

  Cisco IOS Release 12.4(4)T supports only priority levels 1 and 2.
- The keyword **priority** uses a value of 1 or 2 to indicate the priority of the parameters.

Note Choosing priority 2 is similar to using the keyword **all** where all the paramters are selected.

The corresponding set of VQ parameters are sent in the MGCP DLCX message based on the priority selected.

Examples

The following example enables voice-quality statistics reporting for MGCP:

```
Router> enable
Router# configure terminal
Router(config)# mgcp voice-quality-stats
Router(config)# end
```

The following example shows the VQ parameters selected for priority 1:

mgcp voice-quality-stats priority 1

```
16:38:20.461771 10.0.5.130:2427 10.0.5.133:2427 MGCP...... -> 250 1133 OK
P: PS=0, OS=0, PR=0, OR=0, PL=0, JI=65, LA=0
DSP/TX: PK=118, SG=0, NS=1, DU=28860, VO=2350
DSP/RX: PK=0, SG=0, CF=0, RX=28860, VO=0, BS=0, LP=0, BP=0
DSP/PD: CU=65, MI=65, MA=65, CO=0, IJ=0
DSP/LE: TP=0, RP=0, TM=0, RM=0, BN=0, ER=0, AC=0
DSP/IN: CI=0, FM=0, FP =0, VS=0, GT=0, GR=0, JD=0, JN=0, JM=0,
DSP/CR: CR=0, MN=0, CT=0, TT=0,
DSP/DC: DC=0,
DSP/CS: CS=0, SC=0, TS=0,
DSP/UC: U1=0, U2=0, T1=0, T2=0
```

The following example shows all the VQ parameters selected for the keyword **all**:

mgcp voice-quality-stats all

```
16:38:20.461771 10.0.5.130:2427 10.0.5.133:2427 MGCP...... -> 250 1133 OK
P: PS=0, OS=0, PR=0, OR=0, PL=0, JI=65, LA=0
DSP/TX: PK=118, SG=0, NS=1, DU=28860, VO=2350
DSP/RX: PK=0, SG=0, CF=0, RX=28860, VO=0, BS=0, LP=0, BP=0
DSP/PD: CU=65, MI=65, MA=65, CO=0, IJ=0
DSP/PE: PC=0, IC=0, SC=0, RM=0, BO=0, EE=0
DSP/LE: TP=0, RP=0, TM=0, RM=0, BN=0, ER=0, AC=0
DSP/ER: RD=0, TD=0, RC=0, TC=0
DSP/IC: IC=0
DSP/EC: CI=0, FM=0, FP =0, VS=0, GT=0, GR=0, JD=0, JN=0, JM=0, JX=0,
DSP/KF: KF=0, AV=0, MI=0, BS=0, NB=0, FL=0,
DSP/CS: CR=0, AV=0, MN=0, MX=0, CS=0, SC=0, TS=0, DC=0,
DSP/RF: ML=0, MC=0, R1=0, R2=0, IF=0, ID=0, IE=0, BL=0, R0=0,
DSP/UC: U1=0, U2=0, T1=0, T2=0,
DSP/DL: RT=0, ED=0
```

Related Commands

Command	Description
debug mgcp	Enables debug traces for MGCP errors, events, media, packets, parser, and CAC.
ip rtcp report interval	Configures the RTCP statistics polling interval.

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

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