

# Table of Contents

<b><u>PGW 2200 Softswitch Voice Quality Echo Error Resolution</u></b> .....	<b>1</b>
<u>Introduction</u> .....	1
<u>Prerequisites</u> .....	1
<u>Requirements</u> .....	1
<u>Components Used</u> .....	1
<u>Conventions</u> .....	2
<u>Background Information</u> .....	2
<u>Troubleshoot Voice Quality</u> .....	2
<u>Verify</u> .....	3
<u>NetPro Discussion Forums – Featured Conversations</u> .....	7
<u>Related Information</u> .....	7

# PGW 2200 Softswitch Voice Quality Echo Error Resolution

---

## Introduction

### Prerequisites

Requirements

Components Used

Conventions

### Background Information

### Troubleshoot Voice Quality

Verify

### NetPro Discussion Forums – Featured Conversations

### Related Information

---

## Introduction

This document describes methods to help you troubleshoot voice quality on the Cisco PGW 2200 working in Call Control mode. The methodology that is used in live networks for the Cisco PGW 2200 (Call Control) act on Media Gateway Control Protocol (MGCP) Delete Connection (DLCX) message information during a call disconnect situation. The Cisco PGW 2200 has printed specific bearer related data for calls that meet a specific criteria via the MGCP DLCX message. This methodology should be expanded to include a more complete suite of bearer related, digital signal processor (DSP) data. In an ideal scenario, the bearer related information is written (by the Cisco PGW 2200) in a Call Detail Record (CDR) for each call. For this scenario, a new Cisco IOS® Gateway command **mgcp voice-quality-stats** is created. The Cisco PGW 2200 supports half of the DSP Statistics featurette in the 9.4(1) release with the latest patches and introduces new items in Cisco PGW 2200 release 9.5(1) and later.

## Prerequisites

### Requirements

Readers of this document should have knowledge of these topics:

- Cisco Media Gateway Controller Software Release 9
- DSP Voice Quality Statistics in DLCX Messages
- Part of the DSP Statistics featurette is delivered in the Cisco PGW 2200 from patch 9.4(1)S9
- Recognize and Categorize Symptoms of Voice Quality Problems
- Troubleshoot Echo Problems between IP Phones and Cisco IOS Gateways

### Components Used

The information in this document is based on Cisco PGW 2200 Release 9.4(1) and later (including the latest patches) in combination with Cisco IOS Gateway 12.3 and 12.3T and later.

The information in this document was created from the devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If your network is live, make sure that you understand the potential impact of any command.

## Conventions

For more information on document conventions, refer to the Cisco Technical Tips Conventions.

## Background Information

A DLCX message is sent from the Cisco IOS Gateway to the Cisco PGW 2200 at the end of each call. RFC 2705 allows for the Media Gateway to return Packets Sent (PS), Packets Received (PR), Octets Sent (OS), Octets Received (OR), Jitter (JI), Latency (LA), and Packets Lost (PL) call data, with the MGCP DLCX message to the Cisco PGW 2200.

The information that comes from the MGCP DLCX message identifies standard answers for minor issues, but does not provide all the necessary answers on voice quality complaints per call. Standard parameters do not provide adequate information themselves because they are average data over the entire life of the call. For example, consider MGCP DLCX PL data printed at the end of a call. If you lose 200 packets during a five minute call, there is a significant difference in voice quality if a few packets were lost every 20 seconds or 200 packets all at once. The goal of this document is to provide information on the new MGCP DLCX statistics. The Cisco IOS Gateway sends these statistics to the Cisco PGW 2200 when it is enabled through a command line interface (CLI). By default, no DSP statistics are sent to the Cisco PGW 2200, only the basic items explained for MGCP DLCX in RFC 2705.

## Troubleshoot Voice Quality

When you begin to troubleshoot voice quality complaints from customers, start with the customer definition of the problem. Even if you have hundreds of calls a day, and only ten are voice call complaints reported as echo problems, it is still difficult to determine the exact reason for the customer complaint. For this reason, before you start to troubleshoot a customer complaint (for example, an echo complaint), some questions need to be asked first. Answering the types of questions below help characterize the echo being observed in relation to the start of the call, intermittently, or throughout a call. Depending on the information and the timeframe of the call, you can retrieve the information back from the Cisco PGW 2200 CDR information .

- What was the date and time of the call when the echo was reported?
- What was the phone number from caller and called party?
- Who heard the echo? The caller or the called party? This question is important for the 'echo source direction.'
- Was the echo throughout the duration of the call, and can you link this as 'persistent echo' or 'double-talk echo.' If the echo was constant, we can link this as 'persistent echo' or if the echo fades in and out during the conversation, this is considered 'double-talk echo.'
- Was the echo heard only during the first few words at the start of the conversation, which is part of 'convergence echo'?
- How loud was the echo? Was it very weak, soft, or very loud?

If the echo was loud, it can be linked to persistent echo which may be linked to a hot Echo Return Loss (ERL). For this scenario, you can check the details via the Cisco PGW 2200 CDR information during the time the problem is reported. If echo was soft and persistent, then persistent echo can be linked to a long tail echo.

- Was the caller's speech volume adequate? If not, then the caller has high loss conditions. This scenario is the same for the called party speech volume which also results in the same situation. If not, then the called party has high loss conditions.
- Does the customer hear any static clipping or speech distortion during the call? If not, then double-talk exists, and the source may be from a physical Layer E1/T1 line. If you run into any Layer

1 errors, check this with the Cisco IOS command **show controller**. In this scenario, it can be linked to line and patch code violations.

- What type of phone is the customer using (residential analog, business digital, or GSM )? If the customer reports problems from a residential phone, the line echo source is from the loop conditions. For the business phone, no echo path exists. Therefore, the echo source must be from the digital phone. If the customer is using a GSM phone, the echo source is from a cellular carrier and should have been canceled in the GSM carrier network.

**Note:** If you experience echo calls using headsets on wireless phones, the earpiece design results in these acoustics or hybrid echoes.

- Is the customer using a speaker phone or headset? This type of call connection causes feedback from the loudspeaker into the microphone when the volume is set a 75 percent or higher. If the customer is using a speaker phone, the echo may be acoustic. If the customer is using a headset, the echo may be due to high gain or feedback from the headset volume circuit.
- In some recordings, a man's voice is much louder than a female's. If NLP is enabled on the Cisco IOS Gateway, this causes clipping since the double-talk detector can link this as echo.

## Verify

In this verification example, a call is made via the Cisco PGW 2200 in Call Control mode and the MGCP DLCX information is examined during the time the call is released. Pay attention to the output below highlighted in **bold** text for the echo problems, which is a normal call. In this case, ACOM = 127. If there is no switch (4-wire to 2-wire), there is no echo. Values of 127 in the ERL and ACOM mean that there is no echo.

```
DLCX 207 s3/ds1-0/31@v5xxx-3.cisco.com MGCP 0.1
C: 1A
I: 8
R:
S:
X: CE
```

```
002386: Apr 15 12:34:56.423: send_mgcp_msg, MGCP Packet sent to
10.48.84.25:2427 --->
```

```
002387: Apr 15 12:34:56.423: 250 207 OK
P: PS=460, OS=37348, PR=0, OR=0, PL=0, JI=0, LA=0
DSP/TX: PK=217, SG=0, NS=230, DU=43395, VO=1000
DSP/RX: PK=0, SG=0, CF=0, RX=43395, VO=1000, BS=0, BP=0, LP=0
DSP/PD: CU=0, MI=0, MA=0, CO=0, IJ=0
DSP/PE: PC=0, IC=0, SC=0, RM=0, BO=0, EE=0
DSP/LE: TP=-87, TX=-869, RP=-44, RM=-440, BN=0, ER=127, AC=127, TA=-87, RA=-44
DSP/ER: RD=0, TD=0, RC=0, TC=0
DSP/IC: IC=0
<---
```

```
v5xxx-3#show call active voice
Telephony call-legs: 1
SIP call-legs: 0
H323 call-legs: 0
MGCP call-legs: 1
Multicast call-legs: 0
Total call-legs: 2
```

```
GENERIC:
SetupTime=727150 ms
Index=1
```

PeerAddress=  
PeerSubAddress=  
PeerId=52  
PeerIfIndex=68  
LogicalIfIndex=0  
ConnectTime=72715  
CallDuration=00:00:42 sec  
CallState=4  
CallOrigin=1  
ChargedUnits=0  
InfoType=speech  
TransmitPackets=0  
TransmitBytes=0  
ReceivePackets=394  
ReceiveBytes=26298  
TELE:  
ConnectionId=[0x880E386 0x8E1011D8 0x8008A6F2 0xF5FFFD45]  
IncomingConnectionId=[0x880E386 0x8E1011D8 0x8008A6F2 0xF5FFFD45]  
TxDuration=42190 ms  
VoiceTxDuration=0 ms  
FaxTxDuration=0 ms  
CoderTypeRate=g711alaw  
NoiseLevel=-44  
ACOMLevel=127  
**OutSignalLevel=-87**  
**InSignalLevel=-44**  
InfoActivity=0  
**ERLLevel=127**  
EchoCancellerMaxReflector=16684  
SessionTarget=  
ImgPages=0  
CallerName=  
CallerIDBlocked=False  
OriginalCallingNumber=  
OriginalCallingOctet=0x0  
OriginalCalledNumber=  
OriginalCalledOctet=0x0  
OriginalRedirectCalledNumber=  
OriginalRedirectCalledOctet=0x0  
TranslatedCallingNumber=  
TranslatedCallingOctet=0x0  
TranslatedCalledNumber=  
TranslatedCalledOctet=0x0  
TranslatedRedirectCalledNumber=  
TranslatedRedirectCalledOctet=0x0

GENERIC:  
SetupTime=727150 ms  
Index=2  
PeerAddress=  
PeerSubAddress=  
PeerId=0  
PeerIfIndex=0  
LogicalIfIndex=0  
ConnectTime=0  
CallDuration=00:00:00 sec  
CallState=2  
CallOrigin=1  
ChargedUnits=0  
InfoType=speech  
TransmitPackets=376  
TransmitBytes=25816  
ReceivePackets=0  
ReceiveBytes=0

VOIP:  
ConnectionId[0x880E386 0x8E1011D8 0x8008A6F2 0xF5FFFD45]  
IncomingConnectionId[0x880E386 0x8E1011D8 0x8008A6F2 0xF5FFFD45]  
RemoteIPAddress=0.0.0.0  
RemoteUDPPort=25448  
RemoteSignallingIPAddress=0.0.0.0  
RemoteSignallingPort=0  
RemoteMediaIPAddress=10.48.84.194  
RemoteMediaPort=25448  
RoundTripDelay=0 ms  
SelectedQoS=best-effort  
tx\_DtmfRelay=rtp-nte  
FastConnect=FALSE

AnnexE=FALSE

Separate H245 Connection=FALSE

H245 Tunneling=FALSE

SessionProtocol=other  
ProtocolCallId=1A  
SessionTarget=  
OnTimeRvPayout=1000  
GapFillWithSilence=0 ms  
GapFillWithPrediction=0 ms  
GapFillWithInterpolation=0 ms  
GapFillWithRedundancy=0 ms  
HiWaterPayoutDelay=0 ms  
LoWaterPayoutDelay=0 ms  
TxPakNumber=176  
TxSignalPak=0  
TxComfortNoisePak=218  
TxDuration=42190  
TxVoiceDuration=0  
RxPakNumber=0  
RxSignalPak=0  
RxDuration=0  
TxVoiceDuration=36345  
VoiceRxDuration=1000  
RxOutOfSeq=0  
RxLatePak=0  
RxEarlyPak=0  
PlayDelayCurrent=0  
PlayDelayMin=0  
PlayDelayMax=0  
PlayDelayClockOffset=0  
PlayDelayJitter=0  
PlayErrPredictive=0  
PlayErrInterpolative=0  
PlayErrSilence=0  
PlayErrBufferOverflow=0  
PlayErrRetroactive=0  
PlayErrTalkspurt=0  
**OutSignalLevel=-87**  
**InSignalLevel=-44**  
**LevelTxPowerMean=-869**  
**LevelRxPowerMean=-440**  
LevelBgNoise=0  
**ERLLevel=127**  
**ACOMLevel=127**  
ErrRxDrop=0  
ErrTxDrop=0  
ErrTxControl=0

```

ErrRxControl=0
PlayoutMode = undefined
PlayoutInitialDelay=0 ms
ReceiveDelay=0 ms
LostPackets=0
EarlyPackets=0
LatePackets=0
VAD = enabled
CoderTypeRate=g711alaw
CodecBytes=0
Media Setting=flow-around
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: 1
SIP call-legs: 0
H323 call-legs: 0
MGCP call-legs: 1
Multicast call-legs: 0
Total call-legs: 2

v5xxx-3#

```

**Note:** Delay, echo, and clarity are the three main factors that determine how an individual perceives voice quality. All three must fall within certain bounds for the quality of the received voice signal to be judged acceptable. Another important item involves 'Loss and Level Plans' which is explained in TIA/EIA/TSB-122 . These levels should be checked and recorded at the moment you have performed the acceptance testing with the Local PTT, in case there is interconnection with another PTT – Service Providers.

For echo reporting problems, pay attention to:

- **ERL level:** If it is less than 6 dB, which is less than the G.168 requirement, you will have echo problems. This is because 'Echo Return Loss' is the difference between the transmit level from the E1 test set to the returned echo signal from the 2-wire hybrid. Typically, the echo return loss of a 2-wire/4-wire hybrid is 20 db. There should be a minimum loss of 6 db for echo cancellers to operate correctly. If an ERL is measured between 0 db and 12 db, the cause of the echo should be investigated. This could be due to a faulty hybrid or a phone that has an incorrect impedance.
- **Playout delay,** if playout delay is very large (for example, above 150 ms), then normally, there is either echo or customers may have a hard time when attempting to start a conversation.
- **Background noise level:** If the background noise level is consistently very high (for example, –30 dBm), it may create double-talk scenarios.
- **ACOM level:** If it is very close to the ERL level and the value is not 127, then echo cancellation (ECAN) does not converge or work properly.

# NetPro Discussion Forums – Featured Conversations

Networking Professionals Connection is a forum for networking professionals to share questions, suggestions, and information about networking solutions, products, and technologies. The featured links are some of the most recent conversations available in this technology.

NetPro Discussion Forums – Featured Conversations for Voice
Service Providers: Voice over IP
Voice & Video: Voice over IP
Voice & Video: IP Telephony
Voice & Video: IP Phone Services for End Users
Voice & Video: Unified Communications
Voice & Video: IP Phone Services for Developers
Voice & Video: General

## Related Information

- [Cisco PGW 2200 Softswitch Tech Notes](#)
- [Configuration Examples for the PGW 2200](#)
- [Voice Technology Support](#)
- [Voice and IP Communications Product Support](#)
- [Voice, Telephony and Messaging Technical Support eLearning Solutions](#)
- [Recommended Reading: Troubleshooting Cisco IP Telephony Cisco Press, ISBN 1587050757](#)
- [Technical Support – Cisco Systems](#)

All contents are Copyright © 1992–2004 Cisco Systems, Inc. All rights reserved. Important Notices and Privacy Statement.