VOICE QUALITY

SOLUTIONS GUIDE

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NOTICE

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Chapter 1

INTRODUCTION

Introduction

This guide presents best practices for ensuring voice quality, from IP network and Ethernet switch configuration to telephone cabling. The following causes of problems with voice quality in IP telephony are discussed:

- Echo
- Crosstalk
- Excessive speech transcoding
- Packet loss
- Latency (delay)
- Jitter
- Incorrect settings in the network

When talking about factors in the provisioning of good quality voice, it is important to discuss the things you can do to prevent problems. It is also important to keep in mind that you will not be able to solve every type of voice quality problem.

For example, here are some voice quality issues that are out of your control:

- Acoustic echo caused by:
  - Handsfree car phone systems
  - Speakerphones or phones in handsfree mode
  - Standalone conference phones
  - Improperly designed phones where vibrations from the loudspeaker transfer to the microphone through the handset casing
- Electrical echo caused by poor or missing echo cancellation:
  - in Customer Premise Equipment (CPE) at the other end of the network.
  - anywhere else in the (Public Switched Telephone Network) PSTN.

This guide does not provide troubleshooting information. For detailed troubleshooting guidance, refer to the *MiVoice Business Voice Troubleshooting Guide*.

Terms and Acronyms

The following table defines some of the terms used in this guide.

Table 1: Terms and Acronyms

TERM	DEFINITION
CODEC	A codec is a generic name used to describe a signal COder and DECoder where the two terms and functions have been combined. Generally a CODEC will involve the translation of voice in one form to another and typically from an analog signal to digital. The most common telephony CODEC is that specified by ITU-T standard G.729.
Compression (Speech Compression)	Speech compression is performed by a CODEC (coder/decoder). A typical compression CODC would be ITU-T Standard G.729.
Crosstalk	Crosstalk occurs when an audio signal is improperly coupled from one channel to another. The most common and noticeable form of crosstalk is between channels, where a different conversation or signaling tones from another channel can be heard. Crosstalk can occur in handset cables and sound similar to echo. Crosstalk in handset cables is normally small and will be cancelled by an echo canceller.
Customer Premise Equipment	The telephone equipment on the customer site. On a Mitel customer site, the MVoice Business system is the CPE. Incoming calls that arrive at the MiVoice Business over the PSTN typically originate at CPE equipment located on the other side of the PSTN.
Double Talk	Occurs when more than one speaker is talking at the same time.
Echo (Acoustic)	When acoustic echo is present on a call, one or both parties hear their own words repeated tens or hundreds of milliseconds later. Acoustic echo originates at the phone, and is caused by sound from the speaker being reflected back into the handset microphone, or by a user's voice being reflected off nearby surfaces and picked up by the microphone. If the round-trip delay of the reflection is less than 50 ms, these reflections result in sidetone. However, if the round-trip delay is greater than 50 ms, these reflections can result in echo. Echo will be heard by the far-end user if the echo suppressor or canceller in the phone fails to remove all acoustic or mechanical reflection, or if the required echo canceller is not present.
Echo (electrical)	Also called line echo. When electrical echo is present on a call, one or both parties hear their own words repeated a few tens or hundreds of milliseconds later. The reflections that cause electrical echo originate at line hybrids. These reflections cause echo if the round trip delay is long (typically greater than 50 ms) and the echo is not removed by an echo canceller.
Echo canceller	A device that removes echo from a signal by adaptively predicting the echo signal and then subtracting the predicted echo from the signal. Since the mathematical model used to predict the amount of echo to be removed is not an exact model and since there are always some non-linearities in the echo path, not all echo will be removed. An echo canceller normally includes an echo suppressor, also called a non-linear processor (NLP), to remove this residual echo.
Hub	A hardware device that connects multiple twisted pair or fiber optic Ethernet devices together and makes them act as a single network segment. Hubs work at the physical layer (layer 1) of the OSI model. Mitel IP phones should not be connected to the network through hubs.
Hybrid	Hybrids are components (typically line cards or circuit modules) that adapt the four-wire digital network to the two-wire analog network used in the PSTN.

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Table 1: Terms and Acronyms (continued)

TERM	DEFINITION
Jitter	The difference between the minimum amount of time and the maximum amount of time required for a packet to propagate through the packet network. Jitter can be considered the variation in latency.
Latency	Latency in a call is the amount of time it takes the caller's voice to reach the other end of the connection. As latency increases in a conversation, it becomes increasingly difficult for users to sustain a normal two-way conversation.
Layer 2 Switch	A network switch that forwards data traffic based on media access control (MAC) addresses.
MCD Mitel Communications Director	Now called MiVoice Business
Public Switched Telephone Network (PSTN)	The world-wide, public, circuit-switched telephone network. It consists of telephone lines, fiber optic cables, microwave transmission links, cellular networks, communications satellites, and undersea telephone cables, all inter-connected by switching centers, allowing any telephone in the world to communicate with any other.
Reflection	A portion of a signal is sent back towards the signal's source. If the round-trip delay in the reflections is short (less than 50 ms), the reflections are perceived by users as sidetone. However, if the round trip delay is long (greater than 50 ms) the reflections are heard as echo. Reflection can be caused by line hybrids or by acoustic or mechanical audio coupling at the phone.
Sidetone	Sidetone is an electrical reflection that has a very short delay. Often, it is purposely transmitted from the handset microphone to the handset speaker to prevent the phone from sounding dead. On 2-wire analog phones, sidetone is a result of the local hybrid. On IP phones, special circuitry or software is usually required to create sidetone. The same reflected signal that creates the sidetone that reassures a user that the phone is not dead will generate annoying echo if the round-trip delay from the mouthpiece to the ear piece increases to more than 50 ms.
Subnet	Subnetting is a technique used by a network administrator to subdivide a network into smaller networks using the same network number assignment. Subnets are connected together through routers. Hubs and gateways are not needed. Layer 2 switches provide connections to devices within the same subnet.
QoS	Quality of Service
Time Division Multiplexing (TDM)	A type of digital multiplexing that involves the combination of numerous signals for transmission on a single communications channel or line, with each signal being broken into different segments of short duration. Multiple signals are interleaved as sub-channels in one communication channel.

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Chapter 2

VOICE QUALITY

Guidelines for ensuring good voice quality

When you are building or adding to a network, knowing the best practices for network deployment will help you prevent voice quality problems.

Note: This document provides a brief overview of common voice quality issues, and how to prevent or resolve them.
For detailed information, see the *MiVoice Business Engineering Guidelines* and the *MiVoice Business Voice Quality Troubleshooting Guide*.

General guidelines when setting up a new network or adding network elements

Follow these general guidelines to minimize voice quality complaints:

- Design the LAN to be fully switched.
- Ensure that IP phones are connected to the IP network only through layer 2 switches. Do not use hubs.
- Use echo cancellers at hybrids (2-4 wire converters).
- The MiVoice Business platform Ethernet ports default to auto-negotiate. Ensure that the Network Layer 2 switch ports are also configured to auto-negotiate.
- Use VLANs to segregate voice packets from data traffic.
- Use L2 and L3 QoS mechanisms for wired LANs.
- Use Wireless Multimedia Quality of Service (QoS) mechanisms for WiFi LANs.
- Use full-duplex connections wherever possible.
- Provision extra bandwidth whenever possible.
- Run Ethernet cabling in its own cable channels. Do not run Ethernet cables in the power cable channels alongside power cables.

Network configuration guidelines

Configuration errors can also cause problems with voice quality. Check through these configuration items to ensure that your system is configured to minimize the possibility of voice quality problems.

MiVoice Business Platform and Gateways

The following settings, when configured incorrectly, can cause voice quality problems:

- Loop Start Trunk impedance settings: Use the Line Measurement Tool to determine the correct settings. Refer to the *MiVoice Business Engineering Guidelines* for information.
- Loss and Level Plan: Check the **Country** setting in the **License and Options Selection** form in the MiVoice Business System Administration Tool. The **Country** setting determines the default language, dialing plan, tone plan, and loss and level plan. Refer to the *3300 ICP Hardware Technical Reference Guide* for loss and level plan details.

- Audio Level Settings: Ensure that **ONS/OPS Station Descriptor** parameters in the MiVoice Business System Administration Tool are set correctly.

  If this issue is occurring at a specific phone, ensure that the volume on the phone is set to a normal level using the volume keys.
- Employ Bandwidth Management when encountering heavily loaded LANs. Refer to “Bandwidth Management” in the System Administration Tool online help for programming instructions.

VLANs

The following guidelines apply for MiVoice Business Releases 4.x up to 6.0. For detailed implementation information and the latest recommendations, refer to the *MiVoice Business Engineering Guidelines* and the Release Notes for the MiVoice Business release running on your controllers.

- With dual-port phones, configure the LAN switches to use tagging for the voice VLAN and no tagging for the default VLAN. This ensures that voice packets are properly prioritized over data applications from the PC.
- Configure ports to pass all active VLANs with tagging from one switch to another (there is no untagged information present in the connection). This maintains priority information between LAN switches.
- Configure ports to:
  - accept tagged information
  - accept untagged information and pass it on to a specified VLAN.
- Do not use VLAN 0. The specification standard for VLAN 0 is not definitive, and this can lead to incompatibility between different vendor units.
- Set Ethernet Priority to a high value for voice. For equipment-specific settings, refer to the Layer 2 and Layer 3 QoS Settings information in the *MiVoice Business Engineering Guidelines*.

Note: Check with the switch vendor; some prefer to use Priority 5, which also works.

- Set Priority for untagged VLAN/native VLAN/default_vlan to 0.
- Hubs do not support priority queuing, so use managed layer 2 switches with 802.1p/Q support.
- The various equipment vendors require different VLAN settings. For best voice quality, follow the vendor-specific VLAN settings specified in the *MiVoice Business Engineering Guidelines*.
- Do not share the voice VLAN with data devices.

Subnets

- Create subnets in your network to reduce the reach of broadcast traffic.

- Include layer 3 devices to improve speed within communities of interest and the overall network, and reduce the burden on the system to all broadcast traffic.
- Configure all network ports for auto-negotiation so that the network devices can select the best speed settings.

WAN Layer 3 Switches

- Ensure that the packet per second (PPS) rate of routers and switches is adequate for the amount of voice traffic.
- Set the router Maximum Transmission Units (MTU) value to divide larger packets into smaller packets, especially for slower speed links (anything less than T1 rates). See the *MiVoice Business Engineering Guidelines* for details.

LAN Quality of Service (QoS) Polices

The MiVoice Business platform can apply different LAN QoS policies to voice packets, signaling packets, and other packets. Use the **LAN Policy (QoS)** form in the MiVoice Business System Administration Tool to set the LAN QoS policy values.

For detailed information and vendor-specific settings, refer to the *MiVoice Business Engineering Guidelines*.

Firewall settings

Ensure that the appropriate firewall ports are correctly configured for the application, signalling, and voice and media connections. Incorrect settings are often the cause of no-audio or one-way audio. For detailed information about port settings, refer to the application documentation or the *MiVoice Business Engineering Guidelines*. Appropriate Quality of Service (QoS) settings should also be configured to ensure correct prioritization for media and signalling, as with any other network router.

Where connections must pass between different networks with different addressing schemes, the MiVoice Border Gateway (MBG) can provide the necessary application-specific Network Address Translation (NAT). A typical example would be users located on the public Internet requiring connection to a private Enterprise network. A standard NAT firewall is not application-specific and may not translate IP addresses included in application protocols, including SIP. Use of a standard NAT often results in signalling connections but one-way audio.

IP Phones

The IP phone Ethernet ports are set to auto-negotiate, by default. Ensure that the network layer 2 ports are also configured to auto-negotiate.

Types of voice quality problems and their causes

Echo on the line is the most commonly reported voice quality problem, but there are several other voice quality issues you may run into. This section discusses the following typical problems:

Echo

When echo is present on a call, one or both parties hear their own words repeated tens or hundreds of milliseconds later. There are two types of echo, based on cause: acoustic echo and electrical echo.

- Acoustic echo: Acoustic echo originates at the phone, and is caused by sound from the speaker being reflected back into the handset microphone, or by a user's voice being reflected by nearby surfaces and picked up by the microphone.

  If the round-trip delay of the reflection is less than 50 ms, these reflections result in sidetone. However, if the round-trip delay is greater than 50 ms, these reflections can result in echo. Echo will be heard by the far-end user if the echo suppressor or canceller in the phone fails to remove all acoustic or mechanical reflection, or if the required echo canceller is not present.

- Electrical echo: Also called line echo. The reflections that cause electrical echo originate at points where the line impedance changes. For example, there may be a change of wiring type somewhere in the loop, or poor termination or corrosion of connection terminals.

  Electrical echo can also be caused by line hybrids. The microphone and loudspeaker, or handset, each need two wires for analog audio connections. The hybrid converts these four signal wires to the more common two wires used to connect analog phones.

  These reflections cause echo if the round trip delay is long (typically greater than 50 ms) and the echo is not removed by an echo canceller.

Who is responsible for cancelling echo?

These sources of echo can occur anywhere in the network, and you will not always be able to fix the problem. In some cases, people maintaining other parts of the global telephone network are responsible for cancelling the echo in their part of the network.

If a reflection occurs in equipment that does not insert a long delay, for instance, that equipment is not responsible for removing the reflection. The reflection needs to be removed only when it is about to enter a long delay path. Examples of long delay path calls include a long distance phone call across the continent, across the ocean, or across a satellite connection.

All Mitel products that are used as PSTN gateways for IP systems include the necessary echo cancellation to ensure the network echo is not reflected back to the end user. The IP phones and applications also include echo cancellation so that local acoustic echo is also not reflected back to the PSTN and the far end station.

The echo cancellation in the phones also ensures that IP Phone to IP Phone calls are not subject to local echo across the IP network.

For a simple test to find out if you are responsible for resolving the echo problem, or if the problem is originating from the Customer Premise Equipment or from somewhere in the PSTN, see “Identifying the Source of Echo” in the *MiVoice Business Voice Quality Troubleshooting Guide*.

Crosstalk

Crosstalk usually occurs in larger wire bundles, especially those that consolidate many analog pairs going to, or from, a central office or local exchange. Often crosstalk occurs because of an excessively loud signal on one pair in the bundle. Most commonly, it occurs between pairs that have poor impedance termination. Often this poor termination is a result of the user end cables being untwisted for large distances, or where the user end has many connections from a central input either with many phones, or with stubs that have no connections. Such connections are also usually poor performers with DSL connections.

Other areas of crosstalk exist between telephone cables and other services at the termination. Again, untwisted cables are more likely to pick up such signals. This type of crosstalk is more often due to telephone cables running alongside power cables, or next to power equipment such as motors and generators, or alongside radio frequency (RF) transmitters, such as WiFi base stations. Follow the local wiring practices to minimize situations like these.

Signal interference, which may be mistaken for crosstalk, may also occur in the end device if it is located next to high powered devices such as motors, high powered lamps, or cell phones.

Typically these impairments affect analog connections, such as LS trunks and ONS phones, more than digital or IP phones.

IP and digital phones are less affected by crosstalk and interference because of the network protocols and digital transport method. However, where the devices are affected, it is likely that the data network does not conform to wiring standards, or additional precautions need to be put into place. Such issues will also manifest themselves with other data devices. One example might be the speed of data transfer between two PCs, or the time it takes to transfer a print job.

Excessive speech transcoding

When you place a VoIP call, your analog voice signals are converted to digital data, compressed into packets, and then transported over the network. The conversion and compression/decompression process is called “transcoding” and is performed by coder/decoders (CODECs).

Note: The terms “transcoding” and “compression” are often used interchangeably, but they are not the same. Transcoding is the direct digital-to-digital conversion of one voice digitization standard to another. It occurs when voice information is changed from one CODEC type to another. Compression simply refers to a reduction in the amount of data.

The amount of voice compression applied by the CODEC determines the quality of the voice signals. The quality of compressed voice is never as good as uncompressed voice, but intelligibility will typically be maintained.

Compression affects other aspects of the MiVoice Business system as well. These include IP phones, the MiVoice Business platform, MiVoice Business devices, IP applications, IP networking routes and trunk routes, and licenses. Voice subjected to excessive compression or too many transcoding hops (for example, from G.711 standard to G.729.a and back to G.711) results in the voice sounding as if it is coming through a drain pipe.

Packet loss

The far-end user's voice may suddenly sound hollow, as if he is talking down a drainpipe, and then stop completely. This is sometimes described as echo, but this effect is a result of excessive packet loss or lost connection. The issue is network-related and should be investigated.

Common causes of packet loss within the network include:

- Congestion at a connection that results in buffer overflow and data loss.
- Highly variable jitter that causes packets to arrive too late at a gateway or IP phone. The packets arrive too late to be used for voice, and are therefore discarded.
- Out-of-sequence packets being discarded on WAN connections,
- Incorrect duplex settings on LAN connections, resulting in data collisions and packet loss.
- Lack of network bandwidth.

A lack of network bandwidth on the LAN or WAN can delay voice packets from reaching their destination, and result in packet loss. If a voice packet is delayed beyond the ability of the receiving gateway to interpolate its content, the gateway leaves a real-time gap in the packet. This gap causes part of the transmitted speech to be lost at the receiving end, resulting in choppy voice.

Although some packet loss can be handled on an ongoing basis, bursts of packet loss will be noticed by users. A network with 0.1% packet loss over time sounds much different than a network with the same loss but occurring in bursts of three or more packets.

Refer to the *MiVoice Business Voice Quality Troubleshooting Guide* for troubleshooting instructions.

Latency (delay)

Latency in a call is the amount of time it takes for the caller's voice to reach the other end of the connection. Latency (and jitter and packet loss) can be caused by incorrect settings in the network.

As latency increases in a conversation, it becomes increasingly difficult for users to sustain a normal two-way conversation. The conversation rapidly deteriorates from an interactive exchange to an "over-to-you" radio-style form of communication. Severe latency in the network can result in dropped calls.

Latency becomes noticeable in a call when the one-way network delay is greater than 80 ms, and is radio-style at 400 ms delay. Assuming that jitter and packet loss are not an issue, end-to-end delay is the most likely cause of voice quality issues on VoIP calls. From International Telecommunication Union - Telecommunication Standardization Sector (ITU-T) recommendations and based on practical experience, the end-to-end one-way network delay for a voice call should not exceed 80 ms.

Jitter

Jitter is the variation in the packet delay. The major cause of jitter is network congestion. This occurs when the amount of data arriving at a node exceeds the capacity of the node to forward the data. In this case, data is stored in a buffer queue until the node is able forward the data. The time that the data spends in this buffer before being forwarded is the major source of jitter.

While high quality network routers, switches, and Network Interface Cards (NICs) can generally forward packets at the network data rate; many routers, switches, and NICs are limited in the number packets they can forward per second. This can result in the users experiencing choppy voice calls.

Common symptoms and causes

The first step in diagnosing a voice problem is completing a full site survey that includes traffic patterns for telephony, what the existing LAN looks like (bandwidth and equipment), legacy telephone wiring, power wiring. For detailed site-survey checklists, and detailed problem, cause, and solution descriptions refer to the *MiVoice Business Voice Quality Troubleshooting Guide*.

For quick reference, Table 2 describes the most common symptoms that are reported, and the most likely causes.

Table 2: Common symptoms and causes

SYMPTOM	MOST COMMON CAUSES
Electrical echo	• Impedance mismatch: Use the LS Measurement Tool to plan and apply the correct settings. • Packet loss or bit errors caused by incorrect QoS settings. • Excessive latency, usually in the VoIP network, either on the Mitel side of the network or in the CPE VoIP on the other side of the PSTN. Add bandwidth or configure Bandwidth Management Zones. **Note:** In MiVoice Business 5.0 SP1, the maximum number of bandwidth management zones was increased from 250 to 999. • Faulty echo canceller in Analog Services Unit, Analog Main Board, or Analog Options Board. Replace the faulty board.
Acoustic echo	• Volume is too high, and received audio is being picked up by the microphone. Reduce the volume and/or use the handset (if the problem occurs only in handsfree mode). • Faulty handset or faulty phone. • Caller is on a cell phone or Bluetooth device.
Hissing Buzzing Hum Motorboating	• A distribution frame, the ASU, or the MiVoice Business controller is not properly grounded. • Ethernet cables are routed alongside power cables. • Ethernet cables are longer than 1.5 meters (5 feet) at the user termination and picking up local noise such as transformer power units. • Power brick is faulty. **Note:** All calls have some idle channel noise.
Crackling Clicking Popping Static	• Packet loss due to lack of network bandwidth. Increase bandwidth. • Incorrect QoS settings. • Packet loss due to contention among devices on the hub. Mitel IP Phones are not supported on hubs. Replace the hub with a Layer 2 or Layer 3 switch. • Faulty cabling or faulty programming. • Excessive collisions caused by auto-negotiate mismatch or cabling errors (for example: 10000Mbps being negotiated across a Cat 5 cable that can only really support 100Mbps). • Insufficient SLAs with WAN link vendor.Make sure the WAN link vendor supports the CODECs being used, and ensure that they are matching the priority bits you are using.

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Table 2: Common symptoms and causes (continued)

SYMPTOM	MOST COMMON CAUSES
Choppy voice	• Too much latency, too much jitter, or lack of network bandwidth. Increase bandwidth. • Incorrect QoS settings. • Distance to wireless base station is too great.
Clipping (truncation when both caller and called talk over each other)	• Too much echo suppression in the network. • Speakerphone volume is too high. • Caller or called is speaking very loudly.
Stutter	• Incorrect QoS settings.
Garbled voice Poor intelligibility Distortion	• Insufficient bandwidth. Add bandwidth or configure Bandwidth Management zones. **Note:** In MiVoice Business 5.0 SP1, the maximum number of bandwidth management zones was increased from 250 to 999. Incorrect QoS settings. • Excessive speech transcoding. Simplify the network to use only G.711 transmission format. Adjust zone settings in bandwidth management. • Incorrect bandwidth settings within bandwidth management allows calls through when insufficient bandwidth exists. Adjust the limits. • Incorrect analog trunk descriptor settings, • Incorrect ONS/OPS station descriptors settings • Caller or called is using a speakerphone and is talking too loudly. Reduce volume and/or switch to handset. • Caller or called is using a non-TIA-compliant phone. • Faulty microphone in handset, headset or speakerphone. • Wireless handset or headset battery is low.
Distorted Music On-Hold	• Music volume too high. • Too many transcoding hops. • Wrong music file format. **Note:** Music files must be in Windows WAV files. The audio format must use CCITT µ-Law (North America) or A-Law (Europe).
Crosstalk	• Faulty cabling or connections. Check the wiring for compliance to the appropriate standards. • TDM switching matrix is faulty.
Dropped Calls	• Insufficient bandwidth. Increase bandwidth. • Bandwidth Management and Call Admission Control is configured incorrectly. • Incorrect QoS settings. • E2T card has not booted properly. Use the **DBMS Save** command to restart the E2T card. • Programming error. Hotel/Motel Call Blocking option may be enabled, preventing calls.
Silence: The caller or called (or both) are not being heard.	• Incorrect firewall settings. • Default MiVoice Business gateway IP address configured incorrectly in MiVoice Business platform. • Default gateway IP address configured incorrectly on IP Phone. • Router maintenance interrupted the call. • Incorrect subnet mask.

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Table 2: Common symptoms and causes (continued)

SYMPTOM	MOST COMMON CAUSES
Tinny audio (drain pipe)	• Call is using compression. Remove call compression and add bandwidth, or configure Bandwidth Management and Call Admission Control to ensure adequate bandwidth. • Incorrect digital link descriptor programming. • Excessive packet loss on network, or unexpected disconnection.

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Resources for help and troubleshooting

Use the following resources to design and troubleshoot the network for the best possible voice quality:

- *MiVoice Business Engineering Guidelines*
- *MiVoice Business Voice Quality Troubleshooting Guide*
- *3300 ICP Hardware Technical Reference Manual*

Powering connections

mitel.com