**Problem Diagnosis from User Description**

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| |  |  |  |  | | --- | --- | --- | --- | | **Problem** | **Problem Occurs** | | | | **Intermittently** | **Periodically** | **Continuously** | | **Conversational difficulty** | [High jitter level](http://www.voiptroubleshooter.com/problems/jitter.html) | [Route flapping](http://www.voiptroubleshooter.com/problems/routeflap.html) | [Echo problem](http://www.voiptroubleshooter.com/problems/echo.html) | | [Route flapping](http://www.voiptroubleshooter.com/problems/routeflap.html) | [High delay](http://www.voiptroubleshooter.com/problems/delay.html) | | **Gaps in speech** | [Gaps in speech](http://www.voiptroubleshooter.com/problems/gaps.html) | [Route flapping](http://www.voiptroubleshooter.com/problems/routeflap.html) | [Start or end of words missing](http://www.voiptroubleshooter.com/problems/clipping.html) | | [Link failures](http://www.voiptroubleshooter.com/problems/linkfail.html) | [RED in Router](http://www.voiptroubleshooter.com/problems/red.html) | | [Start or end of words missing](http://www.voiptroubleshooter.com/problems/clipping.html) | [Wireless LAN handoffs](http://www.voiptroubleshooter.com/problems/wlan.html) | | **Tick or Pop Sounds** | [Access link congestion](http://www.voiptroubleshooter.com/problems/access.html) | [Route flapping](http://www.voiptroubleshooter.com/problems/routeflap.html) | [Access link congestion](http://www.voiptroubleshooter.com/problems/access.html) | | [Grounding problem](http://www.voiptroubleshooter.com/problems/ground.html) | [Softphone timing](http://www.voiptroubleshooter.com/problems/softphone.html) | [LAN congestion](http://www.voiptroubleshooter.com/problems/lancongest.html) | | [LAN congestion](http://www.voiptroubleshooter.com/problems/lancongest.html) | [Timing drift](http://www.voiptroubleshooter.com/problems/drift.html) |  | | **Audio quality poor or noisy, level too low or high** | [Access Link Congestion](http://www.voiptroubleshooter.com/problems/access.html) |  | [Amplitude clipping - "buzzy"](http://www.voiptroubleshooter.com/problems/ampclip.html) | | [Distortion](http://www.voiptroubleshooter.com/problems/distortion.html) | | [Hum on call](http://www.voiptroubleshooter.com/problems/hum.html) | | [Noisy call](http://www.voiptroubleshooter.com/problems/noise.html) | | [Grounding problem](http://www.voiptroubleshooter.com/problems/ground.html) | [Voice sounds dead](http://www.voiptroubleshooter.com/problems/dead.html) | | [LAN Congestion](http://www.voiptroubleshooter.com/problems/lancongest.html) | [Voice sounds hollow](http://www.voiptroubleshooter.com/problems/hollow.html) | | **Speech broken up or distorted** | [Access Link Congestion](http://www.voiptroubleshooter.com/problems/access.html) | [RED in Router](http://www.voiptroubleshooter.com/problems/red.html) | [Access Link Congestion](http://www.voiptroubleshooter.com/problems/access.html) | | [Bad Ethernet cable](http://www.voiptroubleshooter.com/problems/badcable.html) | | [Distortion](http://www.voiptroubleshooter.com/problems/distortion.html) | | [LAN Congestion](http://www.voiptroubleshooter.com/problems/lancongest.html) | [Route flapping](http://www.voiptroubleshooter.com/problems/routeflap.html) | [LAN Congestion](http://www.voiptroubleshooter.com/problems/lancongest.html) | |

**Problem: Jitter**

Excessive jitter can result from congestion on LANs, [Access Links](http://www.voiptroubleshooter.com/problems/access.html), low bandwidth WAN links/trunks or the use of [load sharing](http://www.voiptroubleshooter.com/problems/loadshare.html).

[**In depth discussion of jitter sources**](http://www.voiptroubleshooter.com/indepth/jittersources.html)

**Impact**

High levels of jitter cause large numbers of packets to be discarded by the jitter buffer in the receiving IP phone or gateway. This may result in severe degradation in call quality or large increases in delay.

[**Example audio file - 5% packet discard rate:**](http://www.voiptroubleshooter.com/sound_files/5pct_rand_plc.wav)

[**Example audio file - 10% packet discard rate**](http://www.voiptroubleshooter.com/sound_files/10pct_rand_plc.wav)**:**

[**Example audio file - 25% packet discard rate:**](http://www.voiptroubleshooter.com/sound_files/25pct_rand_plc.wav)

**Resolution**

Jitter measurements can be difficult to interpret. [RFC3550 (RTP/RTCP)](http://www.voiptroubleshooter.com/tools/voiptr_rtcp.htm) provides a simple running average of the packet to packet delay variation - this provides useful information about the 500 milliseconds before the RTCP report was sent and hence only reports on about 5-10% of the call. The Jitter Envelope measurement provided by [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm) is more accurate and measures jitter in a way that correlates better with the impact of jitter on VoIP systems.

For jitter levels under 100 milliseconds then it may be acceptable to increase the [jitter buffer size](http://www.voiptroubleshooter.com/problems/jbconfig.html) in end-systems or to enable adaptive [jitter buffer](http://www.voiptroubleshooter.com/problems/jitterbuffer.html) operation.

For jitter levels over 100 milliseconds then increasing the jitter buffer size to avoid packet discards will introduce significant delay and cause [conversational problems](http://www.voiptroubleshooter.com/problems/delay.html); in this case it is preferable to take steps to isolate the source of jitter and eliminate the problem at source.

**Tool Description: RTCP - RFC3550**

RTCP forms part of the RTP protocol used to carry Voice over IP. RTCP provides simple packet loss, jitter and endpoint "wallclock" time information.

The newer RFC3611 RTCP XR protocol has a helpful Voice over IP Metrics Payload that contains much useful information

**Applications**

RTCP is not widely implemented and hence information may not be available from all endpoints. When available, RTCP packets can be captured and decoded by a network analyzer. Note that the jitter metric in RTCP is not particularly easy to relate to packet discard rate - it is quite possibly to have a low RTCP jitter metric but a high rate of discards.

RTCP XR provides a wealth of useful data including packet loss rate, discard rate, burst metrics, delay, signal level, noise level, echo return loss, call quality metrics and jitter buffer statistics

**Problem: Jitter Buffer Configuration**

A [jitter buffer](http://www.voiptroubleshooter.com/problems/jitterbuffer.html) temporarily stores arriving packets in order to minimize [delay variations](http://www.voiptroubleshooter.com/problems/jitter.html). If packets arrive too late then they are discarded. A jitter buffer may be mis-configured and be either too large or too small.

**Impact**

If a jitter buffer is too small then an excessive number of packets may be discarded, which can lead to call quality degradation. If a jitter buffer is too large then the additional delay can lead to [conversational difficulty](http://www.voiptroubleshooter.com/problems/delay.html).

**Resolution**

A typical [jitter buffer](http://www.voiptroubleshooter.com/problems/jitterbuffer.html) configuration is 30 milliseconds to 50 milliseconds in depth. In the case of an adaptive jitter buffer then the maximum size may be set to 100-200 milliseconds. Note that if the jitter buffer size exceeds 100 milliseconds then the additional [delay](http://www.voiptroubleshooter.com/problems/delay.html) introduced can lead to conversational difficulty

**Problem: Jitter Buffer**

A jitter buffer temporarily stores arriving packets in order to minimize [delay variations](http://www.voiptroubleshooter.com/problems/jitter.html). If packets arrive too late then they are discarded. A jitter buffer may be mis-configured and be either too large or too small.

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**Resolution**

A typical [jitter buffer configuration](http://www.voiptroubleshooter.com/problems/jbconfig.html) is 30mS to 50mS in size. In the case of an adaptive jitter buffer then the maximum size may be set to 100-200mS. Note that if the jitter buffer size exceeds 100mS then the additional delay introduced can lead to [conversational difficulty](http://www.voiptroubleshooter.com/problems/delay.html).

**Problem: Route Flapping**

Routers can occasionally enter a state in which they periodically switch route – typically every few minutes. This results in the delay alternately increasing and reducing.

**Impact**

As this usually occurs at a fairly low frequency (say every 30 seconds) then the impact on VoIP call quality is small, generally appearing as an occasional “tick”. .

**Resolution**

Use traceroute to determine where the problem is occurring. Resolution may require reconfiguration of routers.

**Tools:**

[Traceroute](http://www.voiptroubleshooter.com/tools/voiptr_traceroute.htm)

**Problem: Echo**

Echo is often due to a mismatched hybrid (2 to 4 wire convertor) on the analog part of a telephony connection. Another source of echo is acoustic feedback from speaker to microphone of a telephone handset.

Echo becomes a problem when combined with a significant amount of [delay](http://www.voiptroubleshooter.com/problems/delay.html). For example, if an IP phone was connected over wide area IP network to a VoIP Gateway then the delay would be large – echo that occurred on the trunk side of the Gateway would be audible in the IP Phone. If a user reports an echo problem then the source of this problem is likely to be on the other end of the connection.

Talker echo occurs when some proportion of the "talker's" voice is echoed back to the talker - i.e. the person speaking hears their own voice.

Listener echo occurs when some proportion of the talker's voice is echoed from the listener's end of the connection and then a second echo occurs which causes some proportion of this signal to be reflected back to the listener. This results in the listener hearing an echo of the talker's voice.

[Convergence echo](http://www.voiptroubleshooter.com/problems/convergence.html) occurs at the start of a call, and results from the time taken for the echo canceller to "converge".

**Impact**

Talker echo can be very annoying. When coupled with low delay, echo can lead to "[hollowness](http://www.voiptroubleshooter.com/problems/hollow.html)" whereas in the presence of high delay echo sounds like....echo.

[**Example audio file - 25 milliseconds round trip delay:**](http://www.voiptroubleshooter.com/sound_files/echo_200.wav)

[**Example audio file - 200 milliseconds round trip delay:**](http://www.voiptroubleshooter.com/sound_files/echo_1600.wav)

[**Example audio file - 400 milliseconds round trip delay:**](http://www.voiptroubleshooter.com/sound_files/echo_3200.wav)

**Resolution**

Generally Voice over IP Gateways incorporate a line echo canceller to remove or reduce the echo level from analog loops. If this is not functioning correctly, possible due the echo canceller being disabled, to mis-configuration of the signal levels ([loss plan](http://www.voiptroubleshooter.com/problems/lossplan.html)), non-linearity in the speech path or an excessively high echo level then some residual echo may be present. To resolve echo problems it is necessary to identify both the source of the echo (i.e. a particular analog loop or line card) and check its balance or configuration and then to identify why the echo canceller is not adequately compensating for the echo.

**Problem: Loss Plan**

The primary purpose of a Loss Plan is to control the losses of all connections in a telephone network such that:

1. the overall loss rating (OLR) of the end to end connection is approximately 10dB, which represents the optimum level
2. the loudness at intermediate stages is not too low, in which case the signal to noise ratio may be too low and voice activity detectors in echo cancellers/suppressers may not function correctly
3. the loudness at intermediate stages is not too high, in which case echo cancellers may not function correctly and amplitude clipping may occur.

**Impact**

An incorrect Loss Plan can lead to problems ranging from high levels of [echo](http://www.voiptroubleshooter.com/problems/echo.html) to phone calls that sound "[dead](http://www.voiptroubleshooter.com/problems/dead.html)"

**Resolution**

Develop a Loss Plan (see [TIA TSB122 July 2000](http://www.tiaonline.org/standards/sfg/committee.cfm?comm=tr-41&name=User%20Premises%20Telecommunications%20Requirements)) and verify that Gateways, Phones etc. are properly configured to the correct levels.

**Problem: Connection Sounds "Dead"**

Generally if a connections sounds "dead” then the level of background noise is too low. This can be due to an echo canceller being mis-configured or to a poor [loss plan](http://www.voiptroubleshooter.com/problems/lossplan.html). Other causes are mis-configuration of the background noise level in a Voice Activity Detection / Silence Elimination or Line Echo Canceller function.

**Impact**

User report that connection sounds "dead" even though normal speech is possible.

**Resolution**

Check echo canceller settings in Gateway. If VAD is being used then consider disabling.

**Tools:**

Audio Editor

**Problem: Hollow Sound**

This is generally due to a high level of [echo](http://www.voiptroubleshooter.com/problems/echo.html) with a small amount of delay. For example, if an IP phone was connected over a LAN to a VoIP Gateway then the delay would be very small – echo that occurred on the trunk side of the Gateway may cause “hollowness” in the IP Phone.

**Impact**

Users may report that calls sound "hollow", "cavelike" or "tunnel-like".

[**Example audio file:**](http://www.voiptroubleshooter.com/sound_files/echo_200.wav)

**Resolution**

Initially try and locate the source of echo - often on the analog loop connected to a Gateway. The echo canceller in the Gateway should compensate for echo however may not operate correctly if mis-configured or if the loss plan is incorrect.

If the problem seems to be associated with a particular telephone handset then it may be due to internal feedback - consider using a different handset.

**Problem: Gaps in Speech**

This may be due to a high rate of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) or packet discard due to [jitter](http://www.voiptroubleshooter.com/problems/jitter.html), or to a problem, with Voice Activity Detection associated with an echo canceller.

The Voice over IP hardware may be configured to use silence insertion instead of [packet loss concealment](http://www.voiptroubleshooter.com/problems/plc.html), hence during a period of high loss or congestion many short segments of speech can be missing.

Voice Activity Detection is used with some CODECs to determine if speech energy is present. If there is no speech energy present then packets are not transmitted. Echo cancellers use a similar voice energy detection algorithm to mask echo in low level speech signals (echo suppression) and may insert silence.

Another possible cause is a high rate of [link failures](http://www.voiptroubleshooter.com/problems/linkfail.html).

**Impact**

Users report gaps in speech or difficulty in understanding speech.

[**Example audio file - 10% packet loss with silence insertion:**](http://www.voiptroubleshooter.com/sound_files/male1_1_sil_20ms_10.wav)

[**Example audio file - Voice Activity Detection problem**](http://www.voiptroubleshooter.com/sound_files/VAD_clipping.wav)**:**

[**Example audio file - Echo suppression problem:**](http://www.voiptroubleshooter.com/sound_files/temporal_clipping.wav)

**Resolution**

Investigate the source of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) or [packet discard](http://www.voiptroubleshooter.com/problems/jitter.html) problems and verify that the jitter buffer is set to a sufficient depth. Verify that the VoIP hardware is configured to use [packet loss concealment](http://www.voiptroubleshooter.com/problems/plc.html). Check for excessive route changes or link failures in core IP network.

If there is no apparent packet loss or discard then check the operation of Gateway echo cancellers.

**Problem: Packet Loss**

Packet Loss can occur for a variety of reasons including [link failure](http://www.voiptroubleshooter.com/problems/linkfail.html), high levels of congestion that lead to buffer overflow in routers, [Random Early Detection (RED)](http://www.voiptroubleshooter.com/problems/red.html), [Ethernet](http://www.voiptroubleshooter.com/problems/badcable.html) problems, and the occasional misrouted packet

[**In depth discussion of packet loss distribution**](http://www.voiptroubleshooter.com/indepth/burstloss.html)

**Impact**

Packet Loss causes degradation in voice quality. If [packet loss concealment](http://www.voiptroubleshooter.com/problems/plc.html) (PLC) is used then isolated losses may be less noticeable.

[**Example G.711 audio file - 10% packet loss**](http://www.voiptroubleshooter.com/sound_files/male1_3_rep_20ms_10.wav)**:**

Packet Loss typically occurs in bursts of 20-30% loss lasting 1-3 seconds. This may mean that the average packet loss rate for a call appears low although the user reports call quality problems.

Example G.711 with PLC audio file - bursty packet loss

**Resolution**

Try and identify the source of packet loss by examining [packet metrics](http://www.voiptroubleshooter.com/tools/voiptr_stats.htm) available from switches and routers along the voice path.

If packet loss is accompanied by a high level of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) or packet loss is very bursty then the problem may be congestion related.

If packet loss is present when the jitter level is very low or if the problem is specifically associated with one or more users on a specific Ethernet switch then check for [Ethernet problems](http://www.voiptroubleshooter.com/problems/badcable.html).

**Tools:**

[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Bad Ethernet Cable**

A bad or noisy Ethernet cable can cause high rates of corrupted packets. This results in high rates of FCS and Alignment errors being reported by network interface cards or analyzers.

High rates of frame error can often be caused by [Duplex Mismatch](http://www.voiptroubleshooter.com/problems/duplex.html).

Late collisions are symptomatic of an excessively long Ethernet segment resulting in too much delay which impacts the collision detection process.

**Impact**

A high rate of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) causes continuous degradation in call quality throughout a call.

**Resolution**

This class of problem often results in a sustained high packet loss rate without a corresponding high level of jitter.

Replace or reroute bad Ethernet cables. Resolve Duplex Mismatch by configuring both Ethernet Interfaces to full duplex.

**Tools:**

[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm), [Switch Statistics](http://www.voiptroubleshooter.com/tools/voiptr_stats.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Duplex Mismatch**

Ethernet duplex mismatch can occur when two ends of an Ethernet connection attempt to negotiate a full duplex connection, this can occasionally result in a situation where one device thinks the connection is full duplex and the other thinks it is half duplex. This can result in a high rate of packet loss without the high rate of jitter that would be typical of congestion.

This problem can be prevented by ensuring that the Ethernet connections are properly configured.

**Impact**

This problem can result in very high rates of corrupted packets which results in a high rate of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html), hence severe call quality degradation.

**Resolution**

Duplex mismatch can be traced by looking at Switch and Router Ethernet statistics as it will result in high rates of FCS errors, Receive errors and Runt packets. The problem can be resolved by reconfiguring one or both network interfaces on the offending Ethernet segment.

**Tools:**

[Ethernet Switch Statistics](http://www.voiptroubleshooter.com/tools/voiptr_stats.htm), [Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Packet Loss Concealment**

Packet Loss Concealment is a technique used to mask the effects of lost or discarded packets. PLC is generally effective only for small numbers of consecutive lost packets, for example a total of 20-30 milliseconds of speech, and for low packet loss rates.

Packet loss can be bursty in nature - with periods of several seconds during which packet loss may be 20-30 percent. The average packet loss rate for a call may be low however these periods of high loss rate can cause noticeable degradation in call quality.

PLC algorithms typically involve either replaying the last packet received ("replay") or some more sophisticated algorithm that uses previous speech samples to generate speech. Simple replay algorithms tend to lead to "[robotic](http://www.voiptroubleshooter.com/problems/robotic.html)" sounding speech when multiple consecutive packets are lost. More sophisticated algorithms can provide reasonable quality at 20% packet loss rates however can consume DSP bandwidth and hence reduce the number of channels that can be supported in, for example, a high density gateway.

**Impact**

If PLC is not enabled then users may report difficulty in understanding speech due to short gaps. If PLC is used but is not effective then the problem is likely to be bursts of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) or periods of high [discard](http://www.voiptroubleshooter.com/problems/jitter.html) rate.

|  |  |  |  |
| --- | --- | --- | --- |
|  | Silence Insertion | Replay last packet | G.711 Appendix 1 |
| 5% loss rate | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_2_sil_20ms_5.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_4_rep_20ms_5.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_1_itut_20ms_5.wav) |
| 10% loss rate | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_1_sil_20ms_10.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_3_rep_20ms_10.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_2_itut_20ms_10.wav) |
| 20% loss rate | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_5_sil_20ms_20.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_1_rep_20ms_20.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_3_itut_20ms_20.wav) |
| 40% loss rate | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_5_sil_20ms_40.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_2_rep_20ms_40.wav) | [http://www.voiptroubleshooter.com/graphics2/audio.gif](http://www.voiptroubleshooter.com/sound_files/male1_3_itut_20ms_40.wav) |

These example files use G.711

**Resolution**

Ensure that PLC is enabled.

Try and identify the source of packet loss by examining packet metrics available from switches and routers along the voice path. If packet loss rates are low then check for [congestion](http://www.voiptroubleshooter.com/problems/jitter.html) in limited speed links.

**Problem: Link Failures**

A link failure typically appears as a period of consecutive packet loss that can last for many seconds, followed by a change in delay after the link is re-established.

Link failures can be caused by equipment problems (e.g. a failed “blade” in a switch or router, power failure…), a cable being unplugged or cut, a configuration change in the transport network or potentially a denial of service attack. Routers are generally intelligent enough to recognize a link failure and find an alternate route.

**Impact**

Link failure will result in significant [gaps](http://www.voiptroubleshooter.com/problems/gaps.html) in received speech. It is unlikely that link failures will occur frequently however they could potentially last for several seconds.

**Resolution**

Regular occurrence could be symptomatic of equipment or power supply reliability problems. Use traceroute to determine the point at which link failures are occurring.

**Tools:**

[Netwo](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)**[Problem: Random Early Detection (RED)](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)**

[RED is a technique used in routers to apply congestion control. When the queue in the router starts to fill then a small proportion of packets are discarded. This is intended to trigger TCP sources to reduce their window sizes and hence throttle back the data rate. This can cause low rates of](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm) [[packet loss](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)](http://www.voiptroubleshooter.com/problems/packetloss.html) [in Voice over IP streams. There have been reported incidences in which a series of routers apply RED at the same time, resulting in bursts of packet loss.](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)

**[Impact](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)**

[Packet loss due to RED can cause intermittent or periodic call quality problems, including "pops".](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)

**[Resolution](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)**

[Check the packet loss rate and discard rate or](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm) [[jitter](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)](http://www.voiptroubleshooter.com/problems/jitter.html) [level at the receiving VoIP end system. If there is a significant rate of bursty packet loss associated with a high level of jitter then it is possible that the loss could be due to RED. If you are able to examine a trace on an IP network analyzer then you can check to see if the loss occurs at regular intervals - if so then RED is a likely suspect.](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)

**[Tools:](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)**

[[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm),](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm) [[Router statistics](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)](http://www.voiptroubleshooter.com/tools/voiptr_stats.htm)[,](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm) [[VQmon](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

[rk Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm), [Traceroute](http://www.voiptroubleshooter.com/tools/voiptr_traceroute.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Gaps in Speech**

This may be due to a high rate of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) or packet discard due to [jitter](http://www.voiptroubleshooter.com/problems/jitter.html), or to a problem, with Voice Activity Detection associated with an echo canceller.

The Voice over IP hardware may be configured to use silence insertion instead of [packet loss concealment](http://www.voiptroubleshooter.com/problems/plc.html), hence during a period of high loss or congestion many short segments of speech can be missing.

Voice Activity Detection is used with some CODECs to determine if speech energy is present. If there is no speech energy present then packets are not transmitted. Echo cancellers use a similar voice energy detection algorithm to mask echo in low level speech signals (echo suppression) and may insert silence.

Another possible cause is a high rate of [link failures](http://www.voiptroubleshooter.com/problems/linkfail.html).

**Impact**

Users report gaps in speech or difficulty in understanding speech.

[**Example audio file - 10% packet loss with silence insertion:**](http://www.voiptroubleshooter.com/sound_files/male1_1_sil_20ms_10.wav)

[**Example audio file - Voice Activity Detection problem**](http://www.voiptroubleshooter.com/sound_files/VAD_clipping.wav)**:**

[**Example audio file - Echo suppression problem:**](http://www.voiptroubleshooter.com/sound_files/temporal_clipping.wav)

**Resolution**

Investigate the source of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) or [packet discard](http://www.voiptroubleshooter.com/problems/jitter.html) problems and verify that the jitter buffer is set to a sufficient depth. Verify that the VoIP hardware is configured to use [packet loss concealment](http://www.voiptroubleshooter.com/problems/plc.html). Check for excessive route changes or link failures in core IP network.

If there is no apparent packet loss or discard then check the operation of Gateway echo cancellers.

**Tools:**

[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Clipping**

If users report that the start and end of words are being “clipped” then this is typically due to the Voice Activity Detector in the VoIP hardware. Voice Activity Detectors are used for silence suppression in packet voice systems, for echo suppression in echo cancellers and for echo suppression or directional control in speakerphones.

Clipping can be the result of the sound level settings in the VoIP hardware being incorrectly configured (see [Loss Plan](http://www.voiptroubleshooter.com/problems/lossplan.html)).

**Impact**

Users report that words are being clipped - similar in effect to a lower quality speakerphone.

[**Example audio file**](http://www.voiptroubleshooter.com/sound_files/VAD_clipping.wav)**:**

**Resolution**

If VAD or Silence Suppression is being used then you may wish to disable this until the loss plan has been verified. If VAD/Silence Suppression is not being used then the problem may be located in the Echo Canceller in a Voice over IP Gateway.

**Tools:**

Audio Editor (e.g. CoolEdit), VoIP Analyzer

**Problem: Wireless LAN problems**

Wireless LANs can introduce several types of problem to Voice over WLAN/WiFi calls (VoWLAN, VoWiFi). Problems may be due to handoffs between Access Points that introduce gaps in the speech path, areas of low signal strength that result in high rates of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) or [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html), high rates of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) or [delay](http://www.voiptroubleshooter.com/problems/delay.html) due to congestion.

The link layer of a Wireless LAN uses retransmission to increase the reliability of the link. As the delay on a wireless link is very short these retransmissions are generally not noticeable although they can reduce effective throughput. Under conditions of low signal strength this results in increased retransmission and hence an increase in jitter. Delay spikes can also occur during changes in transmission rate.

**Impact**

High rates of packet loss or high levels of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) resulting from congestion (which causes excessive numbers of packets to be discarded by the receiving Voice over IP end system's [jitter buffer](http://www.voiptroubleshooter.com/problems/jitterbuffer.html)) lead to degraded voice quality. Gaps in speech due to handoffs between Access Points will cause occasional gaps in speech or transient call quality problems.

**Resolution**

Emerging IEEE802.11 standards address various aspects of WLAN performance:

(i) Higher speed LANs (IEEE802.11a, g, and later n) provide more bandwidth and capacity

(ii) QoS support (IEEE802.11e) gives voice traffic priority and provides bandwith allocation

(iii) Fast handoffs (IEEE802.11r) minimise the size of gaps

**Problem: Access Link Congestion**

Access links are typically the bottleneck between a high bandwidth LAN and a high bandwidth IP network. An increase in traffic can cause the queue in the edge/access router to fill, which increases [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) and causes a short term increase in delay. High levels of congestion can also introduce [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) due to buffer overflow or [Random Early Detection](http://www.voiptroubleshooter.com/problems/red.html) (RED).

As an example, it takes 8 milliseconds to send a typical data packet over a T1 connection. If two data packets arrive at the access router ahead of a voice packet, then the voice packet would be delayed by 16 milliseconds. If the access link speed is slower than T1 then the delay would be greater - for a 512 kilobit/sec link the delay would be 24 milliseconds per packet.

Access link congestion can be a particular problem for [ADSL and Cable Modem](http://www.voiptroubleshooter.com/problems/adsl.html) connections.

**Impact**

High levels of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) resulting from access link congestion cause excessive numbers of packets to be discarded by the receiving Voice over IP end system's [jitter buffer](http://www.voiptroubleshooter.com/problems/jitterbuffer.html), which leads to degraded voice quality. As the level of congestion varies with traffic then the jitter level will vary, hence users may report that the call becomes [garbled](http://www.voiptroubleshooter.com/problems/distortion.html) intermittently.

**Resolution**

Access link problems can be reduced by

1. Using priority queuing for delay sensitive voice and video traffic
2. Reducing the maximum MTU size on low speed links (512 kbits/s or less)
3. Increasing the capacity of the access link
4. If multiple links are used, then applying load sharing to maximize use of capacity
5. Applying call admission control to limit the number of calls
6. Using fragmentation and interleaving.

Tools: [Traceroute](http://www.voiptroubleshooter.com/tools/voiptr_traceroute.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: ADSL and Cable Modem Congestion**

Although ADSL and Cable Modems are regarded as “Broadband”, they are not fast enough to ensure adequate Voice over IP performance. Very high levels of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) can be seen on many DSL and cable connections due to interference from data traffic.

The chart below shows the serialization delay (time taken to send) of an packet (of size MTU) on links of various speeds. Jitter would be directly proportional to this delay (e.g. may be 2-3 times the serialization delay).



**Impact**

High levels of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) resulting from access link congestion cause excessive numbers of packets to be discarded, which leads to [degraded](http://www.voiptroubleshooter.com/problems/distortion.html) voice quality.

**Resolution**

ADSL and Cable Model congestion can be mitigated by limiting [MTU](http://www.voiptroubleshooter.com/problems/mtu.html) size to 512 bytes or less, using [priority queuing](http://www.voiptroubleshooter.com/problems/priority.html) for real-time traffic and limiting the number of Voice over IP streams that are simultaneously carried.

**Problem: Distortion**

There are a variety of problems that can cause distortion.

**Impact**

Users may report that calls sound "[noisy](http://www.voiptroubleshooter.com/problems/noise.html)", "[dead](http://www.voiptroubleshooter.com/problems/dead.html)", "[hollow](http://www.voiptroubleshooter.com/problems/hollow.html)", "[cavelike](http://www.voiptroubleshooter.com/problems/hollow.html)" or "[tunnel-like](http://www.voiptroubleshooter.com/problems/hollow.html)", have [echo](http://www.voiptroubleshooter.com/problems/echo.html), sound slightly [distorted](http://www.voiptroubleshooter.com/problems/whirlybird.html), sound [robotic](http://www.voiptroubleshooter.com/problems/robotic.html), or be very [choppy](http://www.voiptroubleshooter.com/problems/gaps.html) or [garbled](http://www.voiptroubleshooter.com/problems/gaps.html) (see also [Packet Loss](http://www.voiptroubleshooter.com/problems/packetloss.html))

[**Example audio file - "robotic":**](http://www.voiptroubleshooter.com/sound_files/40pct_rand_plc.wav)

[**Example audio file - "choppy":**](http://www.voiptroubleshooter.com/sound_files/vad_problem.wav)

**Resolution**

See the relevant link above.

Tools: Audio Editor, [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Whirlybird Distortion**

Generally due to the use of specific types of CODEC, notably ACELP.

**Impact**

User reports a high level of distortion through the call.

Example audio file:

**Resolution**

Consider changing to an alternative CODEC type.

**Problem: "Robotic" Speech**

This is generally due to a high rate of [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) or packet discard due to [jitter](http://www.voiptroubleshooter.com/problems/jitter.html). The [packet loss concealment](http://www.voiptroubleshooter.com/problems/plc.html) algorithms used in Voice over IP systems are able to deal with occasional packet loss but when the rate of loss is very high then they can lead to [robotic](http://www.voiptroubleshooter.com/problems/robotic.html) sounding or choppy speech.

"[Robotic](http://www.voiptroubleshooter.com/problems/robotic.html)" sounding speech occurs when a large number of packets are dropped, the [packet loss concealment](http://www.voiptroubleshooter.com/problems/plc.html) algorithm tries to compensate but introduces a synthetic or robotic sound.

"Choppy" speech can occur either when a high rate of packet loss occurs and packet loss concealment is not being used, or when the speech level is low and Voice Activity Detection is missing some parts of the voice signal.

It is important to note that packet loss can vary during a call - although the average loss rate may be low, short term congestion can lead to periods when packet loss can be as high as 30-40 percent.

**Impact**

Users may report "robotic sounding" or garbled speech

[**Example audio file- "robotic" 40% packet loss with PLC**](http://www.voiptroubleshooter.com/sound_files/40pct_rand_plc.wav)**:**

[**Example audio file- "choppy" 10% packet loss - no PLC:**](http://www.voiptroubleshooter.com/sound_files/10pct_rand_silence.wav)

[**Example audio file- "choppy" VAD problem:**](http://www.voiptroubleshooter.com/sound_files/vad_problem.wav)

**Resolution**

Investigate the source of packet loss or jitter problems and verify that the jitter buffer is set to a sufficient depth. If packet loss and jitter are low then check for VAD problems.

**Tools:**

[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm), [Switch and Router statistics](http://www.voiptroubleshooter.com/tools/voiptr_stats.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Noise**

There are a number of conditions that can lead to the voice signal being excessively noisy. One common cause is the signal level being [too low](http://www.voiptroubleshooter.com/problems/lossplan.html) at some point on the voice path, when the signal is amplified then the noise is also amplified. Some equipment problems and bad electrical connections can also lead to excessive noise.

**Impact**

Users may report that calls sound noisy.

**Resolution**

Try to correlate the incidence of noise with specific routes or pieces of equipment (including telephone handsets). If the problem appears to be associated with a specific route that connects to an analog or TDM trunk then check the [loss plan](http://www.voiptroubleshooter.com/problems/lossplan.html). If the problem seems to occur on a specific packet path then check for excessive jitter or packet loss.

**Problem: MTU Size**

The Maximum Transfer Unit (MTU) is the largest IP packet that can be accepted on a path, and is often as much as 1500 bytes in length. The maximum delay introduced by a packet is equivalent to the MTU size divided by the link speed - for example for T1 with a 1500 byte MTU the delay from one packet is 8 milliseconds. On link speeds of T1 rate or less the incremental delay caused by data packets causes [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) in voice packets, for example if two 1500 byte data packets are ahead of a voice packet then the voice will have to wait for 16 milliseconds before being transmitted.

**Impact**

Low link speeds (under T1 rate) and large MTU sizes lead to increased levels of jitter, which in turn can lead to packet discard and degradation in voice quality. The chart below shows the variation of serialization delay (time taken to send a packet) with MTU size and link speed - jitter would generally occur as some multiple of this serialization delay.



**Resolution**

On link speeds of less than 512 kbits per second the MTU size should be limited to 512 bytes or less. Other approaches to resolving this problem include packet fragmentation and interleaving.

**Problem: Prioritization**

Class based queuing routers apply different priority levels to different classes of traffic. Traffic classes may be determined by examining the TOS bits in IP headers or by analyzing the packet payload, the latter approach being more common in traffic shapers.

The TOS field in the IP header may be modified or ignored by routers and hence assigning a high priority to a packet does not guarantee that it will receive priority treatment.

**Impact**

High levels of [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) resulting from access link congestion cause excessive numbers of packets to be discarded, which leads to degraded voice quality.

**Resolution**

If you are using prioritization then examine packet traces from different segments of the network to see if the TOS field is being reset. Check the configuration of routers and also check with IP service providers to make sure that they do support prioritization of traffic.

**Tools:**

[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)

**Problem: Grounding**

Poor electrical grounding can cause intermittent noise to appear on voice connections. This would typically be associated with a specific piece of equipment.

**Impact**

User report sporadic clicks or pops

**Resolution**

Consult an electrician concerning grounding of equipment

**Problem: Softphone Task Conflicts**

Conflicts for PC resources in Windows environments can sometimes lead to PC based soft-phones producing delay spikes every few seconds.

**Impact**

Leads to ticks or brief periods of distortion, with the characteristic that they occur periodically.

**Resolution**

Examine tasks running on PC to see what may be causing conflicts.

**Problem: LAN Congestion**

LAN congestion is often the result of using low speed LANs (10BaseT) and/or passive hubs. This results in a high rate of collisions on the LAN, which increases [jitter](http://www.voiptroubleshooter.com/problems/jitter.html) and [packet loss](http://www.voiptroubleshooter.com/problems/packetloss.html) rates. 10BaseT congestion will lead to jitter levels 10 times that of 100BaseT, and hence it is advisable to use 100BaseT for Voice over IP traffic.

**Impact**

LAN congestion can lead to short term congestion, and hence packet discards. LAN congestion tends to be more transient than [access link congestion](http://www.voiptroubleshooter.com/problems/access.html), hence causes short periods of call quality degradation

**Resolution**

LAN congestion can be reduced by using dedicated 100BaseT links for connections to desktops and servers, using switches instead of hubs and using Gigabit Ethernet for inter-switch connections.

**Tools:**

[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm), [VQmon](http://www.voiptroubleshooter.com/tools/voiptr_vqmon.htm)

**Problem: Timing Drift**

Voice over IP systems commonly experience timing drift. This occurs when the internal clock in the sending system runs at a slightly different speed to that in the receiving system – causing drift of typically up to 60 microseconds per second. Timing drift can also occur in the system being used to observe the packet stream. Timing drift is can be seen on a delay trace as a constant slope or a sawtooth.

**Impact**

Low rates of timing drift may cause an periodic audible “tick”. VoIP systems can sometimes hide this by doing necessary timing adjustments during silence periods. If an NTP timing server is used then VoIP systems may resychronize or adjust their clock speed automatically.

High rates of drift can be much more problematic, and may be symptomatic of hardware problems. These can be caused by high temperatures in end systems such as PCs or due to the use of cheap ceramic resonators instead of crystals in low cost IP phones.

**Resolution**

It can be difficult to resolve timing drift problems without consulting the equipment supplier however initially review the configuration of IP phones and gateways to see what guidance is provided. Use an NTP time server to provide a common reference point. Some operating systems, for example Linux, do provide a method for tuning the internal clock rate of computers which can minimize this effect.

**Tools:**

[Network Analyzer](http://www.voiptroubleshooter.com/tools/voiptr_tools.htm)

**Problem: Amplitude Clipping - "Buzziness"**

If the signal amplitude is too high at some point along the analog voice path, when the voice signal is converted to a digital form amplitude clipping can occur.

**Impact**

Users report that speech may seem excessively loud and potentially "buzzy" or "fuzzy".

[**Example audio file with amplitude clipping:**](http://www.voiptroubleshooter.com/sound_files/amplitude_clipping.wav)

**Resolution**

Verify the [loss plan](http://www.voiptroubleshooter.com/problems/lossplan.html) to see where the signal level may be too high. If you are able to capture and decode voice packet streams then use an audio editor to examine the speech signal for signs of high amplitude parts of the signal being clipped.

**Tools:**

Audio editor (e.g. CoolEdit)

**Problem: Hum**

Hum is generally due to the introduction of AC power supply noise into the non-packet parts of the voice path. Typical causes are wiring running too close to power transformers or power lines, leaky power supplies (typically "wall bricks") and common mode noise due to poor grounding.

**Impact**

Users report a noticeable low frequency hum.

**Resolution**

Try to correlate the incidence of noise with specific routes or pieces of equipment (including telephone handsets). Possible causes include the power supply to the handset, if an IP phone, or the routing of loop wiring if an analog handset.

If the problem appears to be associated with a specific route that connects to an analog or TDM trunk then check the Gateway for possible problems.