VoIP Quality of Service:

Now that VoIP -- Voice over Internet Protocol, or "digital telephone," as it is sometimes rebranded -- has gone mainstream, it's important to have a solid understanding of the problems that can crop up. Keep in mind, though, that VoIP is dependent on your broadband Internet connection. Many of the problems that can affect the quality of a VoIP call may be directly due to the quality of the Internet connection itself.

Sound Quality:

The sound quality of a VoIP call is affected by a number of factors outside the VoIP provider's control. On the user end, the quality of the microphone and speakers used will certainly affect how the audio is reproduced. And between you and your intended call recipient, there is a world of Internet hardware that neither you nor the VoIP provider has any control over.

Echo:

Echo can be caused by several factors. One of them, the user can do something about, but others are inherent in the technology. If you are using a microphone and computer speakers, it is pretty easy to understand how an echo can be created: Sound from the speakers is entering the microphone. To minimize this problem, use a headset, preferably one with a noise-canceling microphone.

Echo can also be caused by the quality of the Internet paths that a VoIP call takes. A few VoIP providers try to ensure that they are tied into the highest-quality connections available, technically referred to as Tier 1 carriers, but this goes only so far, because the connections at the user end could be the problem.

Delayed Speech and Long dead spots:

This can be the result of latency or delays on the network or Internet. In some cases, you could experience brief periods where this issue could occur, but if it lasts for more than a day or two, then there may be a problem on the route that your VoIP signals (RTP) are traveling. (Latency is a common issue with satellite connections, so if your broadband connection is from satellite, then this issue could be related to the actual technology itself. In most cases a satellite connection is not adequate for VoIP, even on compressed codecs, because the transmission is burstable which creates significant jitter. This is addition to the latency that these connections create.)

If your router has QoS settings, then enable QoS for your VoIP connection, typically RTP and SIP. This will prevent applications that are running on your LAN from "stepping on" your VoIP (voice) connection. Refer to your routers user guide for instructions on how to do this (not all routers have this feature).

The best test for unusual latency is to measure the ping times between the actual hops (or routers) that the signal is taking. In the case of VoIP, these transmissions are usually UDP and to reproduce the exact paths can be difficult. The next best thing would be to test using a ping test. One utility that not only shows the individual routers, but also shows latency and packet loss, is PingPlotter, which can be used to troubleshoot your VoIP connection over the Internet. If significant latency from your location outward occurs, this would be a significant indication that this latency is causing your voice issues. There is also a very good test designed to give you results designed to measure specifics that would degrade a VoIP connection. The test can be found here. After connecting with the site you can choose a destination city. A simulated call is then measured and you will be presented with a graph indicating a MOS Mean Opinion Score. Several factors go into the calculation for a MOS score, some of which are latency, jitter, time to setup the call and the codec that is chosen. You will find detailed information about their test at the site.

In many cases, latency and packet loss may be the result of issues that are occurring somewhere from the modem out, and the best place to start is right with your particular connection. This would especially be true if day after day you are experiencing the same issue over a period of time. If these problems were somewhere out further on the network, then the chances are that many people would be experiencing them and the ISP would be trying to correct it, to prevent a prolonged issue. As hopefully they would not want to have customers upset at their service. (This is not to say that this does not occur, but rather to troubleshoot close in first.)

It might be the result of a signal issue at your modem or packet loss locally sometimes being caused by a splitter or faulty connector or cable line.

More On VoIP Quality:

VoIP is dependant on both the local and wide area network. If voice quality problems occur, either for brief periods or longer durations the problem can often be the direct result of the network connection. This would include your ISP Internet Service Provider or some other component between you and the party that you are talking with.

Some issues are probably not a network related problem. One would be constant echo on your VoIP line, although a network issue could make this echo worse over certain periods. Another would be the inability not to connect to specific phone numbers. For other consistent voice quality issues we recommend looking at out VoIP tutorials first.

VoIP symptoms of network issues.

Many quality issues with your VoIP connection could be the direct result of network problems. Although these issues could be the result of something else other than network related problems, they are highly indicative of one, especially if your VoIP connection works well most of the time and then experiences problem periods. These symptoms are:

- Bits of voice or words missing.
- Sound (voice) distortion including strange background noises like clicks, ticks, pops or high pitched clangs, etc..
- Delays between the time that you speak and when the other side hears you words, (and vice versa).
- Talking over each other because neither end realizes the other already started speaking.
- Gaps of missing voice.

Causes of poor quality VoIP.

The network related causes of poor quality VoIP as described by the symptoms above, can be broken down into three main categories.

Packet Loss

Data packets carrying parts of a conversation can be lost during transmission. If enough packets are lost, and VoIP is very sensitive to and has a very low threshold to packet loss, then gaps or audible problems will quickly be discernible. The protocols or codecs that are used to transport voice often incorporate Packet Loss Concealment, which will help mask the effects of lost or discarded packets, but they cannot overcome even low percentages of packet loss.

Jitter

The difference in the transit time of packets varies and this difference is called Jitter. As the packets of voice need to arrive in the same order as they were sent, variation will result in a out of sequence part of sound which if not placed in the proper order would make nonsense sound rather than distinguishable words. An IP phone or ATA has a built in Jitter Buffer that introduces a small amount of delay in order to smooth out and sequence these timing variations, but only has a limited capability. (Actually this jitter buffer translates jitter into additional delay and packet loss. Which is another reason VoIP is very sensitive to Jitter.) If some packets arrive too late then they may be entirely discarded producing the bitts of lost voice.

Latency

Latency or delay is the amount of time it takes data to travel from one endpoint to the other endpoint. This travel time depends on several factors including distance, queuing delay, which is the amount of time packets are held in a queue because of congestion on an outbound interface of a device (router), and handling delay which is the result of devices that forward the frame through the network. VoIP depends on RTP traversing in a reasonable amount of time (milliseconds), some people say typically no more than 150ms, but conversations can still be good at the 200-220ms traverse time if other network issues are not a problem (packet loss, jitter). VoIP users will probably notice round-trip delays that exceed 250ms. Above 250ms and callers will begin talking over each other. Around the 500ms range phone calls become impractical and anyone carrying on a conversation with this amount of latency had better end each sentence with "over".

How to network troubleshoot VoIP

To troubleshoot VoIP and the Internet or WAN network that the data is traversing use a good free tool; PingPlotter. Using PingPlotter place the IP address of the providers SIP server as the target address. The resulting graph can be used to locate packet loss at the end targets IP.

Troubleshooting VoIP connections fundamentally are about spotting high latency and packet loss.

Always look at the final destination first, and see if it is registering packet loss. Then look at each upstream hop for the first one (closest to the last hop) that does not show a similar loss. The packet loss problem would then reside between the good hop and the bad hop (hand off).

The bottom of PingPlotter has graphical spikes that will show problematic points.

If you're not seeing any packet loss at the final destination then the other intermediate hops shouldn't be of any concern regarding packet loss. But they could be introducing latency or delay at which point they would be of concern.

An ISP's congestion can affect VoIP quality especially during peak usage hours. Congestion can be recognized by looking at a PingPlotter graph. This event can have detrimental effects on VoIP as the congestion introduces latency and packet loss. Some nodes on cable systems can be over subscribed and during peak hours experience congestion. Typically peak hours would be between 6PM and 9PM.

If packet loss is accompanied by a high level of jitter then congestion could be the problem. Or if packet loss comes and goes in large amounts (known as bursty) then the problem might also be network congestion.

VoIP Faxing

Faxing over VoIP can be a challenge, but with a few configuration changes on your current fax machine, you may find fairly reliable faxing over your VoIP Internet connection.

Faxing over a VoIP connection

First some background on faxing and VoIP

Faxing has been around for years and most of the protocols were written with the intent of sending those signals over traditional phone circuits using sounds. Those sounds were turned back into data by the receiving fax machine, which expects a constant, steady transmission of data, without any loss. If there is some loss of data the receiving fax machine will shut down the transmission.

Faxing over VoIP can be a hit or miss operation. Make the 9600 Baud rate change and disable ECM for the best chance to succeed. The problem is that the codecs used by VoIP ATAs are designed to compress voice, not the analog signals sent and received by modems.

In a VoIP Internet world, voice is first converted into packets and then they are sent over the connections that make up our our vast Internet. They may take slightly different times to arrive at their destination. In doing so some packets may be discarded, but the end result is that the receiving VoIP device has enough packets to make a clear and understandable conversation.

We suggest these settings on a fax machine for faxing over VoIP; slowing the transmission rate down and allowing the machine to continue receiving the transmission even though a few bits of data were lost, then faxing over VoIP can become more consistent. Our suggestions in many cases can resolve issues that prevent faxing over a VoIP connection, but not in all cases. If after trying and making all the VoIP fax changes we suggest you still cannot fax over your connection then try a Internet Fax service.

DSL, Cable And Fixed Wireless Internet:

These are the prefered internet connections that are commonly available. Where cable and DSL are not available, often Fixed Wireless is.

Wireless Internet (3G, 4G, Cellular...):

These types of Internet are much like Satellite Internet (see below). They use slower speed and higher latency connections than other Broadband or Highspeed connections. Please be sure and check a VoIP speed test to see if your provider meets the minimum specifications for VoIP Telephony.

Satellite Internet:

This type of Internet service do not lend itself to VoIP calls. Satellite Internet, regardless of provider, cannot be used for VoIP calls due to the inherent delay involved in carrying your call up to the satellite 22,000 miles above Earth and back down. Low Earth Orbit satellite service may change this, but currently this service is not available.

Common Problems With VoIP:

The marketing around VoIP services point out that they will save you money on both local and long distance phone calls while at the same time providing you with advanced features which are not available from a standard analog phone line. However, there are still some problems with VoIP service that consumers should be aware of while making the decision to switch from standard phone service to Voice Over IP.

The main problem lies in how they handle **911 emergency calls**. When you sign up with a VoIP provider, you typically must provide a valid service address where the VoIP phone will make calls from. Your VoIP provider will then provide you with either basic 911 service or VoIP enchanced 911 service **(E911)**. How E911 service will function varies from provider to provider, so it is important to always read their FAQ's on 911 service prior to choosing them as your VoIP provider. For the most part, E911 will function in the same manner as the 911 service you are used to, but it is best to be aware of your provider's system for handling 911 calls during power outages and over bad Internet connections.

That brings us to the next major problem with VoIP service, which is **power outages**. If you're power goes out, so does your VoIP phone service (in most cases). It is good policy to always have a battery backup in place for your computer and VoIP phone. You should also check with your VoIP provider to make sure they have a **call forwarding** service to send calls to your cell phone at times when your power may be out.

Another major issue with VoIP is that the quality of your phone service will now depend upon a second service, which is your broadband Internet connection. The Clarity and stability of your VoIP calls will be based upon the **reliability** and connection of your high speed Internet service provider. Any problems that arise with your ISP or your Internet connections will also probably affect your VoIP calls, as well. So, if you are not currently using a broadband ISP that you are completely happy with in terms of quality service, then you probably shouldn't consider using VoIP through them since the quality of your VoIP connection will only be as good as the quality of your Internet connection.

What Your Speed Test Tells You:

Download And Upload Speed is a measure of the actual throughput speed for a TCP (which includes HTTP) application including the impact of the connection route latency between the client and the server. When the latency of a connection exceeds the data consumption time for the slowest capacity of the route (usually the clients connection) then the throughput speed will drop below the capacity.

QoS is a measure of how smoothly data packets are moving. If a connection is uncongested or unregulated then every packet should flow at a rate that matches the maximum capacity of the slowest part of the connections route. If regulation of congestion causes this pattern to change then the data flow QoS should drop. Note if there are problems affecting packets but the impact is evenly spread, e.g. all or nearly all of packets are affected then the QoS may still be high.

Round Trip Time is the time it takes for a packet to be sent end-to-end between the client and the server and back. The length and consistency of the trip time ultimately defines the TCP throughput speed. A long trip time will dramatically slow connection throughput speed. An erratic trip time is an early indication of regulation or congestion problems.

TCP Max delay (latency) is a measure of the maximum delay the client was waiting for data to arrive. TCP max delay should normally not exceed the TCP forced Idle time created by the natural latency of the connection. If this is not the case then it indicates that there are problems with the Data Flow QoS of the connection which is causing excessive delays. This is likely to be caused by quality issues such as packet loss or packet regulation.

Max Calls is a measurement of the number of simultaneous phone calls you can make based on your available bandwidth (Download And Upload Speed). This does not tell you how many good calls can be made. If your QoS is bad, every call will be bad. It will not matter if you can support one call or twenty.

The Resolution Process:

The first step in the resolution process is to perform a series of bandwidth speed tests, as a single speed test can be skewed and does not provide representative sample on which to base a complaint. To do this you should take a number of speed tests over a representative period of time at set intervals, such as every 20 minutes for a couple days. A series of tests provides a better picture of network performance and helps identify patterns, such as lower speeds during prime time or rush hour, and makes for a more credible report to provide to your ISP.

It is important that you analyze the data with the purpose of identifying your assessment of the root cause. Taking this extra step does two things to help your case. Firstly it helps the remote support technician to grasp the extent of your problem more quickly, and secondly it helps you to get acceptance of the problem by the ISP. Acceptance of the problem is singularly the most important aspect of getting the problem resolved.