The Problem with Faxing over VoIP Channels

"Lower your phone bill!" is one of many slogans used today by popular Voice-over-IP (VoIP) providers. Indeed, you may certainly save money by leveraging an existing broadband internet connection and dropping your traditional phone line. Occasionally you may notice a few glitches in the sound, maybe a little bit of echo now and again, but certainly it sounds as good or better than your cellular phone does, and perhaps you can't even really tell the difference. Maybe your old traditional phone line was sometimes crackly or hissed and the new VoIP line (channel) is even better than that.

Certainly there are some benefits to VoIP, and certainly there will be many people who will be wise to use VoIP instead of traditional phone lines.

However, someday you may try to send or receive a fax over that VoIP channel. Perhaps that first fax session or two will be perfectly fine. But eventually you may start to notice that faxing just doesn't seem to work as well as you remember it working on the old phone line. What happened?

Generally the problem rests in one of two matters. The first possibility is what is called "lossy" audio compression (codec). The second is what is known as "jitter".

Before understanding why lossy compression and jitter can be a problem for faxing, it is important to first understand what happens in fax communication.

How A Good Fax Works

Faxing involves two fax devices communicating generally for the purpose of passing one or more pages of image data from the sender to the receiver. This kind of data, as is most digital data, is ones and zeros. To communicate data over audio channels such as a phone line the data must be "modulated" - meaning that the data is transformed into an audio waveform that will be decodable by the other end to reproduce the ones and zeros that the data really is. Listening to this modulated audio you may describe it as "beeps", "squawks", "screeching", or "static hiss". Although indecipherable by a human ear, to the listening fax machine those noises have significant meaning.

Here is a picture representation (the audio waveform) of both sides of a fax call.
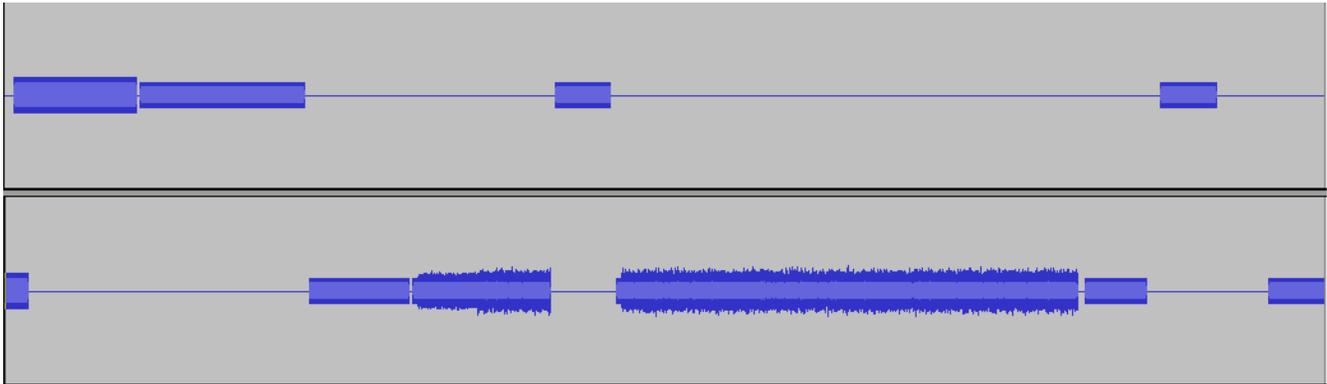
The audio from the fax receiver in the picture is on top, and the audio from the fax sender is on the bottom. Notice that most of the signaling is coming from the sender and that most of the time the receiver spends listening to the sender. Notice also that there are basically three different kinds of audio being represented: message signaling (the thick-bar areas that are smooth on the top and bottom), image data signaling (the rough or fuzzy areas), and silence (the flat line). Lastly, notice that the fax machines do not send signaling at the same time. One sends signaling while the other listens. The listening machine knows that it is time to respond to a signal when silence begins after a period of signaling by the other end.

About Lossy Compression and Jitter

Okay, so what is lossy compression and how does it affect fax calls?

Compression is when a string of source data is taken and then converted in some way into a smaller string. The interpreter on the other end must decompress the string to retrieve the original data. These different ways of compressing audio data are called "codecs". Some codecs are called "lossless" which means that there is no loss in detail of the data when undergoing the compression and decompression exchange. Some codecs are called "lossy" which means that there is a loss of detail when undergoing the compression and decompression exchange; some detail is sacrificed in order to achieve a tighter compression (smaller compressed result).

Faxing cannot occur on lossy codecs because it will consistently remove data from the audio waveform. If you can get some faxes through then this is not the problem you're having. Your VoIP provider is already going to be aware of the need for faxing to occur on lossless codecs, and in most cases if they are providing you with a stated "fax service" on your VoIP channels then undoubtedly they are providing you with lossless channels. You may be required to pay slightly more for this.

A lossless channel does use more bandwidth than a lossy one. Consequently your internet provider will try to up sell you to use more bandwidth if ever you have quality problems with the VoIP service. Perhaps your VoIP provider will even recommend this. However, consider that a normal lossless phone call-quality channel uses 64 kilobits per second of bandwidth. That's only slightly more than what you may have had with 56K dial up internet speeds before you switched to broadband services. A typical broadband internet

connection is at least 2 megabits per second. That's enough to hold 30 simultaneous 64 kilobit channels. Chances are good, too, that your broadband connection is much bigger than that. Your internet provider will be happy to take your money for more bandwidth. But know that it is quite likely that more bandwidth will not resolve your VoIP quality problems. Usually fax problems on VoIP channels occur due to what is known as "jitter", and usually an increase in available bandwidth does not necessarily reduce jitter.

Here are three visual examples of jitter found in VoIP-sent faxes. Each example was sent through a different VoIP provider. Pay close attention as to how they differ from the example above.
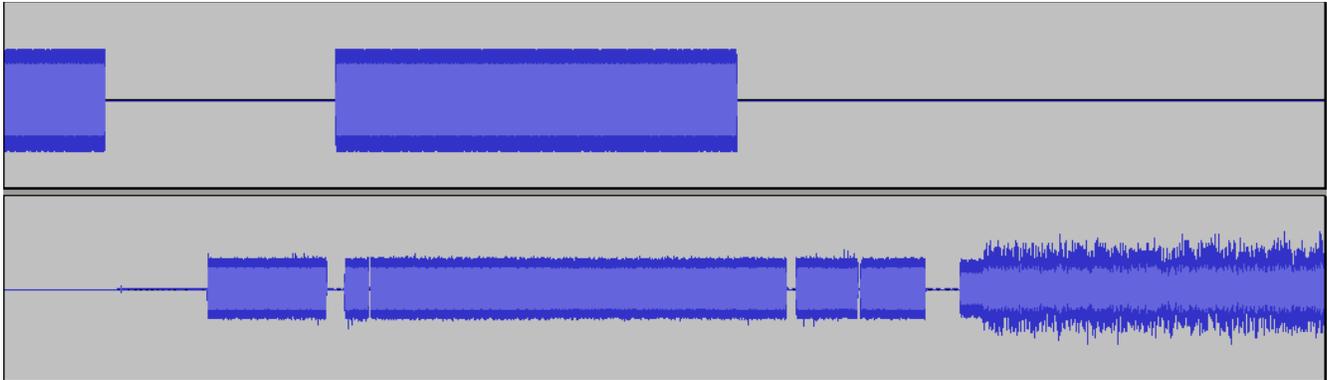
Notice that the message signal is interrupted with small gaps of near-silence. Realize that the largest gap in that signal is only five one-hundredths of one second (5/100 sec). The human ear would not likely detect it. Notice, however, that the receiving fax machine did notice it, and justly assumed that the transition to silence indicated that it was time for it to respond. Consequently both the sender and receiver start signaling at the same time and thus become dis-synchronous.
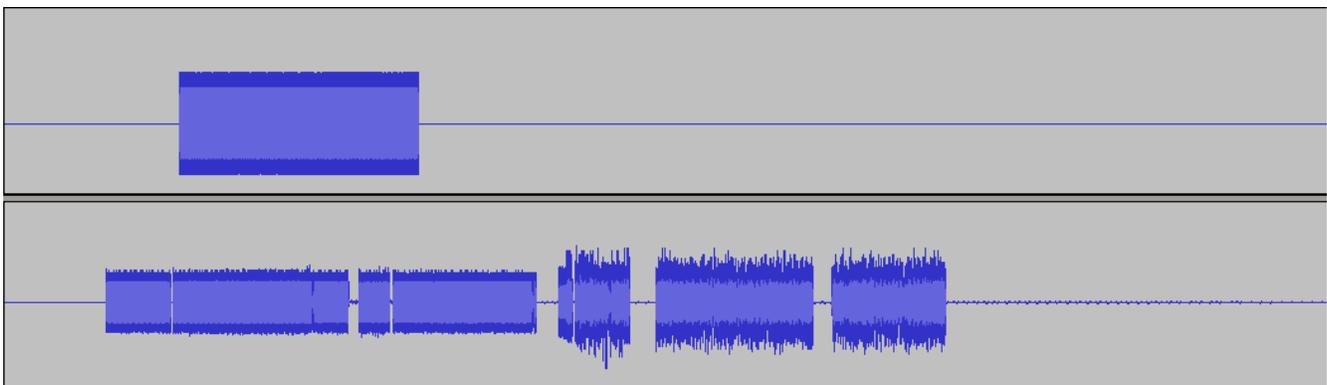
Here is another example of nearly the same thing, except that this time both the message signal and the image data signal are broken. Depending on how the receiving device is programmed a single instance of either of these problems may not necessarily doom a fax session. However, once the two fax endpoints lose synchronization the risk of failure is significantly increased because one or both of the endpoints cannot immediately hear correct signaling from the other. Unfortunately, even if they do resynchronize more jitter is likely to disrupt the communication again in the same call.
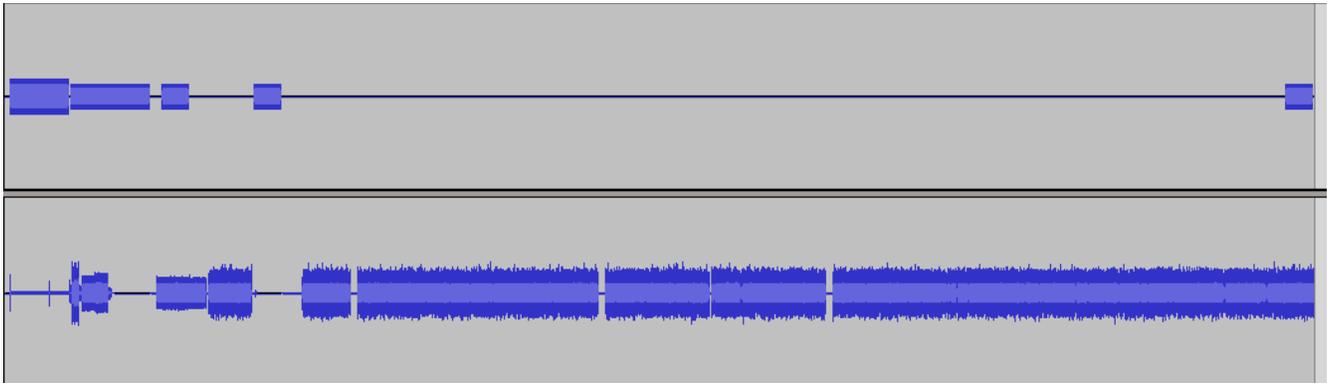
In this example of jitter the gaps in the audio stream appear only during the image data signal. In this case the receiver detected the end-of signal at the first gap and then began patiently waiting for the message signal that should follow the image data signal. After enough time passed without yet detecting the message signal it gave up and disconnected.

What Jitter Is

In a VoIP call the audio is "streamed" between the two endpoints. This varies slightly from how, for example, downloading a music file from the internet works. All data sent across the internet is put into little pieces called "packets", each packet has enough information on it to tell where it came from and where it's going. These packets are sent across the internet network and each of the various "hops" through the internet each station can potentially send the packet through a different route to reach the final destination. On the receiving end the receiver is expected to reassemble all of the packets to reproduce the original data.

Ultimately what happens in that process is that sometimes a station cannot handle a particular packet at a particular time. That station can delay the routing of that packet momentarily if needed. And, depending on the type of packet that the sender used, that station can even ignore and not relay the packet entirely.

Typical internet communications, like browsing an average web page or downloading a music file will not use the type of packet that can be ignored. If the internet stations are unable to relay it then it will be retransmitted. However, VoIP communications, much like most other "streamed" communications, use the type of packet that the internet stations are allowed to ignore if necessary. This is done to keep the audio from pausing at various points and to keep the audio as real-time as possible, like on a phone call with a typical phone line.

This typically works out perfectly fine for voice audio because the human ear does not pick up on the missing audio. And some VoIP equipment, noticing the missing audio, will even synthesize some audio to fill in the gap (this is called a "jitter buffer"). However, both missing and synthesized audio constitute a corruption of the audio data from when it left the sender, and there is no immediate way for the receiver to recover the missing audio data. This is why the jitter you see represented above in the pictures shows up as gaps of silence. An intelligent jitter buffer may close or fill-in the gaps enough to prevent premature detection of a signal-end, however, there still will be missing audio. This missing audio is primarily what makes it difficult for fax to work over VoIP channels.

How This is Different from Traditional Line Behavior

On a traditional phone line audio data comes and goes on the network in a sequential manner that virtually guarantees that the audio sent from one end will be received by the other in the same order and timing as it left. Static and noise does, sometimes still occur on traditional phone lines. This is what it may look like:
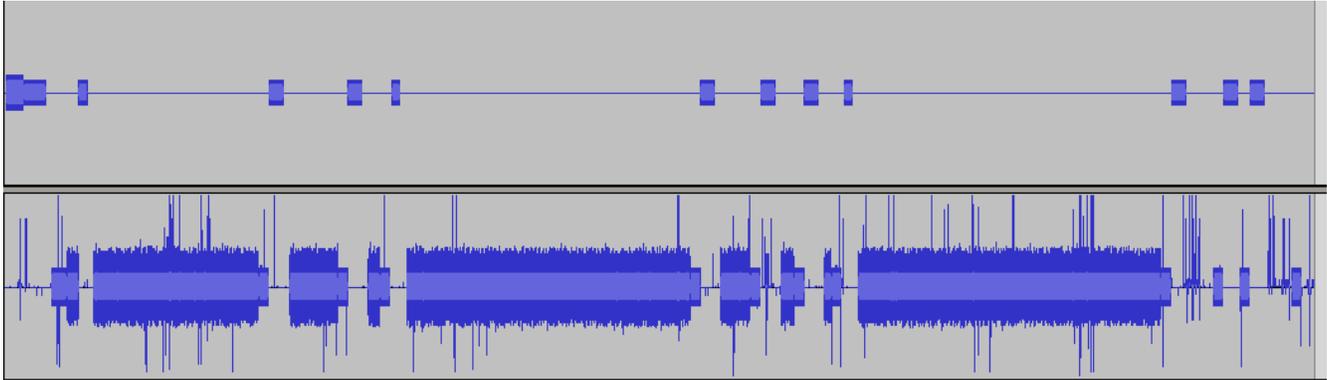
In this case the three pages of this transmitted fax were ultimately received and delivered to the receiver exactly as the sender had sent them with no visual errors whatsoever. They were perfect copies of what the sender transmitted. So, why did the noise clearly visible here not affect the fax reception? There are at least these three reasons:

1) The audio corruption did not involve gaps of silence as commonly seen with VoIP-sent faxes.
2) The moments of noise actually consist of the true audio signal plus some added distortion. Fax machines are typically capable of filtering out many kinds of noises when added to the true audio signal.
3) The fax machines employed an error correction method (ECM) to request, retransmit, and recover any image data that did not make it through.

ECM should even be able to help some instances of VoIP-caused fax trouble. However, that really depends on the frequency and regularity of the occurrence of jitter. If even a small amount of jitter occurs once every second then it will be extremely difficult for the fax session to succeed, even with ECM enabled. In fact, many fax machine manufacturers and VoIP providers will advise you to disable ECM if you experience communications trouble. This advice really isn't a true solution to the problem. Rather, it merely tells the fax machines to accept imperfect page image data and worry more about letting the sender have the chance to transmit all of the pages instead of worrying about getting all of the page image data through perfectly. If imperfect or missing page image data is acceptable to you, then disabling ECM might help. Then again, it may actually make things worse (in the case that ECM was actually helping things).

So, What Can Be Done?

For now, don't try to send or receive faxes over internet VoIP channels. Unless you are anxious to tolerate the headache of unreliable faxing and have an ambition to be on your VoIP provider's support lines to complain about something that they probably cannot resolve, you'll be best-served by simply going back to a traditional phone line for your fax machine. If it's not worth the price to you, then consider purchasing an easy-to-use paper scanner and setting up an account with an internet-based fax service provider for your faxing.

For the long run, encourage your VoIP provider to deploy true Fax-over-IP protocol (FoIP) called "T.38" into their equipment. If they do this, and if they do it correctly, then you should be able to reliably use your fax machine through their FoIP service.