Tips for a VoIP Friendly Network

The cost efficiency, big-business features, and ease of deployment make VoIP a great solution for small- and mid- size businesses, but it is important to have a VoIP friendly network to ensure you have excellent quality. While you may think that your business has plenty of bandwidth, it may not be the bandwidth you need to accommodate VoIP. Fonality wants to make sure that your network is VoIP ready to ensure that you have optimal call quality. There are three very important network-related variables for VoIP; device, bandwidth, and stability.

DEVICE (modem & router)

Modem - Includes a "key" to access your Internet Service Provider's (ISP) network.

Router - Routes traffic to several devices, often includes wireless capabilities.

A VoIP call is only data - your voice, their voice, the ring, touch-tones, any and all noises - passing between callers. Think of this data as a foreign language that needs to be properly translated. A specific translation is needed for VoIP data, so there needs to be a "translator" on your network. This is what a **router** will do - **translate (route) the data.** A router that can't properly translate the VoIP data is like a translator that knows fragments of that language. Your phones end up receiving gibberish, so you end up hearing gibberish.

Sometimes a combo device (**gateway**) is provided by your ISP, which **includes a modem and a wireless router.** The router in a combo-device is only (a) for internet browsing and email or (b) to support the ISP's proprietary phone service. **Gateways are not friendly to VoIP,** as they do not translate the data correctly; however, one of these two changes can easily be made:

Bridge the Gateway - This "builds a bridge" over the routing capabilities of the combo device allowing data to pass through without [incorrect] translation.

Replace the gateway with a "dumb modem" - a modem with NO routing capabilities

Essentially, both methods are a way to setup the modem/gateway to stop trying to translate VoIP data. This, instead, allows the modem to pass the original data to a router that can then translate the VoIP data properly.

BANDWIDTH *(speed of your the network)*

You'll hear, or have heard, the terms "**upload and download speed**". This is the bandwidth, or speed, of your network. It is very important and seldom changes. The rule of thumb is simple: **the more users you have, the faster speed you need**. Without a fast enough speed, only one or two people may be on a call at the same time or the quality of the calls will be bad. **With enough speed, you're in a good place for all users to be on the phone.**

STABILITY *(quality of your network)*

The biggest causes of network instability are **packet-loss and latency.** We'll call this "incomplete data". It's not uncommon to have the fastest bandwidth available and still have a high amount of instability.

The instability is caused by packet-loss. Imagine "packets" as packages on an open-top delivery truck in route from New York to Los Angeles. If there is instability on that road, those packages are likely to fall off the truck. By the time the shipment (data) arrives in LA, it's incomplete. This is no different with data. For voice calls, you need the conversation (data) to be received in full.

A VoIP friendly network should have 0% packet loss.

Instability also leads to latency. Imagine there's traffic on the road and packages aren't delivered on time. Internet VoIP data works the same way. The only difference? There's no HOV or speedy tolls lanes in the internet. Everyone gets equal priority to use the road and, as a result, the latency in sending VoIP data between two points can vary significantly. What complicates matters? If two packages are sent at the same time and they arrive at your door out of order, it's not a big deal. BUT! If you speak two words and they arrive out of order? Well, that's a problem big. Yoda can be fun to imitate but may hinder your business.

A VoIP friendly network should have < 100ms latency

These types of instability may happen once a week, but sometimes network connections are plagued with occurrences several times throughout the day. For most internet functions such as email, browsing, and video streaming, you aren't likely to see many issues with this instability. How and why, you ask?

*You type "google.com" into your web browser. Whether it loads immediately or takes 15 seconds, the page will eventually become complete. An image may take awhile to load, but **the webpage will appear soon enough,** as will the rest of the page.*

*You go to YouTube and look at that funny video your co-workers are passing around, which is only 15 seconds long. Though it may take ten seconds or an hour, **that video will eventually fully load** for your viewing pleasure*

These scenarios can most certainly be annoying, but imagine if that page or video never finished loading? These are not "real-time situations". *That web page will finally load, that email will finally arrive, and that video will finally play all the way through.*

A phone call? **A phone call is a real-time situation.** You talk and the other person responds. If packet-loss or latency are in action, **you're losing bits and pieces, if not all,of the phone call.** Call quality issues will manifest as jitter (alien-like voice and crackle), static, delay, echo, one-way audio ("I can hear them, but they can't hear me."), dropped calls, inability to make or receive calls.

WHAT CAN I DO IF I HAVE THIS TYPE OF PROBLEM!?

Provide your ISP with proof. Calling and saying, "I have a problem. Fix it.", will rarely end with good results. There are a few ways to get evidence showing these interruptions. A ping test is a common method. "Ping" is the same term use in the game Ping-Pong - a signal is sent from your network and bounced back several times. **The tests provide results on the "health" of your network by gauging the length of time that passes (measured in "ms").**

A simple test though the website "**PingTest.net**", which provides readings of jitter, ping, and packet-loss.

An extended test through a company call VoIPSpear (**VoIPSpear.com**) - Their free accounts allow you to run the same ping test consistently on your network. You can login and it will provide graphs showing the state of your network's stability over the prior six hours. (This will require your IP Address. You can get this at IPChicken.com easily.)

Our technicians are able to run tests and provide readings of the results - Sometime this can be done externally, but it may require a "screen-share" session. They are also be able to assist you in setting up the VoIPSpear test mentioned above, if you would like.

None of these tests will interrupt your network or hinder productivity in your business.

A VoIP friendly network should have:

- Ping < 100ms
- Jitter < 10ms
- Packet Loss < 0.1%

A VoIP "unfriendly" network would have:

- Ping > 300ms
- Jitter > 30ms
- Packet Loss > 1%