Not All VoIP is Created Equal

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Until recently, VoIP carried a two-part reputation, lower cost and lower quality. As new technology is deployed and as service providers gain deeper experience, this reputation is changing. All the factors that affect VoIP performance are controllable. With the right planning, you can expect the quality of your voice conversations to actually improve with a VoIP solution. We'll talk about the total solution, VoIP for business as an end-to-end technology. The total solution includes the phone system, the Local Area Network (LAN), the Wide Area Network (WAN) and anyone working virtually outside the wired phone system in the office.

But first, we will set the foundation for measuring voice quality. How do we objectively measure one conversation against the next? We all have opinions on this, but the measurement of quality was standardized long before VoIP was a familiar term. Voice quality is measured using a MOS (Mean Opinion Sore) score. MOS is a legacy telephony acronym that describes the quality of a voice conversation as interpreted by the human ear. This is subjective but it does work.

Mean Opinion Score: Voice Quality Evaluation

MOS	Quality	Impairment
5	Excellent	Imperceptible
4	Good	Perceptible but not annoying
3	Fair	Slightly annoying
2	Poor	Annoying
1	Bad	Very annoying

MOS gives a numerical score to the perceived quality of the media received after being transmitted and compressed using codecs.

Typically, the best score you will see for a telephone conversation is about 4.1 to 4.3. Because of the factors discussed below, your VoIP calls can either be "slightly annoying" to "annoying" or in the "good" to "excellent" range.

The term "toll-quality" came into use about 30 years ago when T1 multiplexers first started transporting voice over private T1 lines. The original idea was that a private WAN could provide voice quality equal to that of the long-distance public switched network. Toll quality was at its

best in the time before we started the shift from analog to digital and much before voice was packetized. Today, VoIP service providers feel they've achieved something noteworthy when their packetized voice service achieves a MOS score of 4 or better.

When we talk in data terminology about things that go wrong with voice calls, we blame latency, packet loss, jitter, signal-to-noise ratio, cross-talk, echo, and other variables. This does not help you much if you're just trying to deliver a quality phone system that does not go down. These are all symptoms of problems that can be fixed.

It makes sense to look at a business VoIP solution in terms of three physical segments where quality and reliability are determined. Some of these are in direct control of the IT staff while others are part of the selection criteria when evaluating a network vendor.

Physical Segments

I. In your building:

Telephone station equipment:
Consider the longevity of the manufacturer and the quality of the equipment they make. This is especially true of speakerphones, which so many people use when sitting at their desk. Also investigate the codec standard being used in the phone (discussed below).

Wire: cabling must be Cat 5 or better, avoid long cable runs (>200 ft). A separate cable run for each phone is preferred but not necessary. *It would be expensive to run a separate wire to each phone.*

On the LAN: remove old bridges (*they won't allow for full duplex communication*), and old switches which are not able to support VoIP needs. WiFi routers cannot be used on the same LAN that is also supporting your telephone system. Set up a voice VLAN to ensure that voice has its own path and has unimpeded priority over all other traffic types. For maximum reliability, use a POE switch to power the phones over the same connection as the LAN. Add uninterrupted power supply (battery) for the WAN router and for the POE switch. *Note, fax lines can be a problem.*

II. On the access network:

Access is the connection from your building to the edge of the carrier's network. For VoIP, the access circuit is usually the biggest choke point - the point in the connection where the bandwidth is most constrained. On the LAN, speeds are usually 100Mb or increasingly, 1Gb. The carrier edge and backbone are usually engineered with the capacity to handle well above the average daily peak traffic of the connected circuits.

For most business customers, this usually means a single T1 (1.5Mb) or bonded T1s, up to a full DS3 (45Mb). Some carriers have also deployed Ethernet over copper technology, offering access speeds up to 20Mb, depending on loop length and conditioning.

Choosing the right amount of bandwidth to support peak voice traffic is a technical discussion we won't address here. Most service providers will have the expertise to advise customers about how much bandwidth is required to support high quality VoIP.

The access component is also the most likely point of failure for a VoIP service. Redundancy options are expensive and require a high level of expertise within the service provider. To protect against the risk of downtime, customers should require their service provider to include repair response intervals and outage credits in their contract.

IIa. In the carrier core network:

Carriers distinguish between the distribution portion of their network and the backbone portion of their network. For distribution, carriers will typically equip speeds of 45Mb to 1Gb. On the backbone, speeds are even faster, usually between 150Mb and 10Gb. With all this bandwidth, the only hitch for VoIP is quality of service (QoS). In VoIP terms, QoS refers to resource reservation control mechanisms. In other words, the carrier's ability to prioritize voice calls over all other traffic types.

Larger organizations will also want to look at the redundancy built into the network, and whether the carrier uses management and monitoring tools to watch VoIP on their network. The tools can be used to measure MOS scores, look at the percent of dropped packets, and other metrics that affect service.

QoS is a critical requirement in all components of a solution. On the LAN, in the form of a prioritized VLAN, and from the CPE all the way through the carrier network, voice traffic has to come first.

III. Codec:

Throughout a VoIP conversation, the codec sets the maximum level of quality. It converts an analog voice signal to digitally encoded version. Codecs vary in the sound quality, and correspond to a bandwidth requirement. Each service, program, phone and gateway typically support several different codecs. When they communicate with each other, they negotiate which codec they will use. Wideband VoIP only works when sent VoIP from end-to-end. With the prevalence of existing PSTN infrastructure in our lives, pure end-to-end VoIP links still make up only a small percentage of communications

traffic. To make sure it is not overly oversimplified; there are many codecs in use today each with their own bandwidth and quality characteristics.

The important point is to ask about the codec being used in your system, and ask about the upgrade path to high definition. As with the migration to HD TV, it will take time for a critical mass of users to be on HD codecs to appreciate the difference. At this point, the migration will accelerate to an all-HD world.

This is a lot to think about when trying to install a VoIP phone system. On the other hand, you have the "care-free" option - throw bandwidth at the problem and run it over the public internet. VoIP will work just fine over the public internet, if.

About the Author

Lou Sommi has more than a decade of executive-level marketing and product development experience in the service provider and telecommunications manufacturing segments. With an extensive understanding of customers, competition and emerging technologies, he is responsible for defining the commercial product lines and marketing direction at Cavalier.