

ACHIEVING VOICE QUALITY IN INTERNET PROTOCOL NETWORKS



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According to a recent study, about 50 million minutes of long distance traffic now travel over Internet Protocol networks every day. That's still just a drop in the bucket when compared to the traditional telephone system, but from zero to 50 million in three years is impressive.

Voice over Internet Protocol (VoIP) is beginning to live up to its promise. But the rate of adoption of VoIP has been slower than anticipated.

Why? Time and again, the reason companies and organizations give for delaying VoIP implementation is call quality. They just don't expect VoIP to sound as good as what they experience when they make a "regular" telephone call.

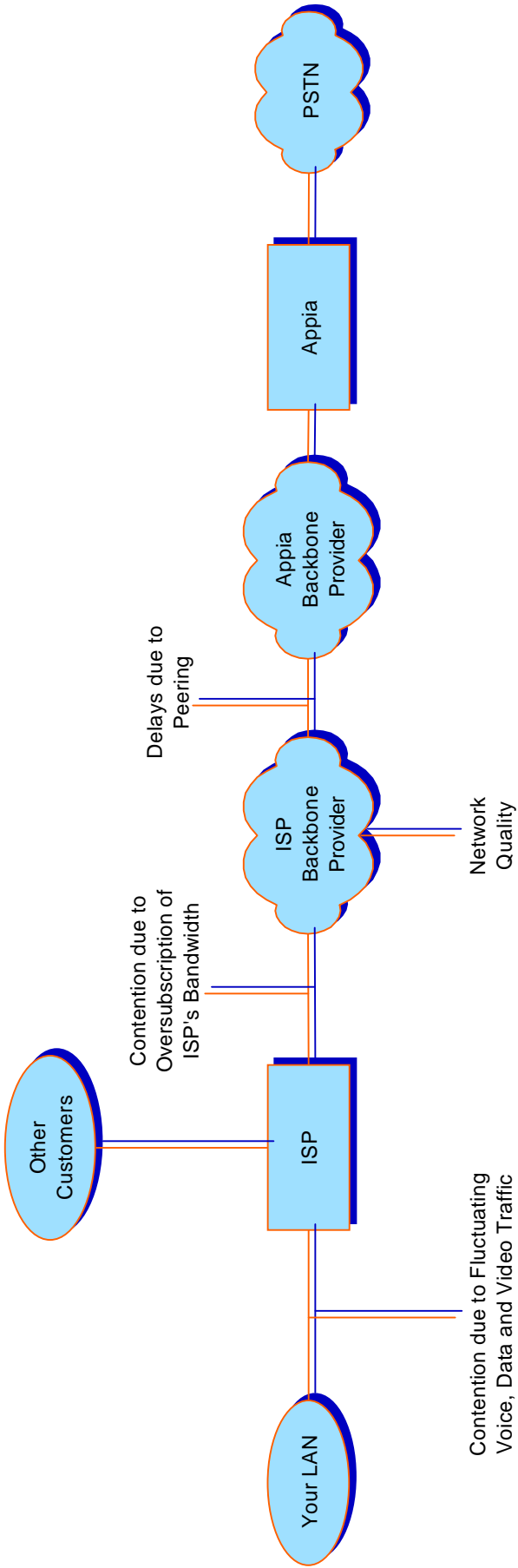
Reasons for poor voice quality

If you have ever used one of the "free" VoIP services that are on the market, you know that the VoIP industry has, to some degree, brought the expectation of poor quality upon itself. Some VoIP providers do things that limit their ability to offer toll quality calling. For example, they use inferior software or gateways, or they use codecs that save on bandwidth but add significantly to latency.

As important as these factors are, they are not likely to affect most companies and organizations - unless, of course, they choose to use the VoIP services offered by the "free" VoIP providers.

The chart that follows this page shows the points at which voice quality can be affected in a typical enterprise.

REASONS FOR DIFFERENCES IN QUALITY OF VOICE CALLS OVER IP NETWORKS



As the chart indicates, the main problems in companies and organizations are:

- Contention for the bandwidth between the customer and the customer's Internet service provider. Specifically, real-time voice traffic can compete (unsuccessfully) with non-real-time traffic such as e-mail, Web browsing, etc.
- Contention for bandwidth between the customers of a given ISP. When an ISP oversubscribes its connection to its backbone provider, its customers do not receive the bandwidth they are paying for. Often, this means small, even imperceptible delays, but even small delays can make all the difference when it comes to voice traffic.
- Reliance on the public Internet. This important factor is discussed at length in the next section.

Why the public Internet stumbles over voice

Right now, almost 70 percent of VoIP calls are routed over the public Internet. In other words, two out of three VoIP calls are being carried over a transport medium that can only confirm VoIP's reputation for inferior sound quality.

The public Internet's inability to handle voice traffic is rooted in its history. First, it was designed for data, not for real-time traffic. So it's perfect for email, downloading Web pages, sending files, and even for such quasi-real time applications as chat. But it is far less effective when packets have to arrive at a particular destination at a particular time - especially when time is measured in milliseconds. As anyone who has dealt with IP backbone providers can attest, this "data centric" approach persists.

Second, even though we use the word "Internet" as if it referred to a single entity, in reality it is a mélange of networks that are owned and operated by a variety of providers who compete with one another. It is hardly surprising, then, that performance varies widely from provider to provider, and that the Internet's performance is also so variable.

A third and related factor is peering between provider networks. No single provider covers the globe, and so virtually all providers peer with one another. In some instances, these arrangements work quite well. But in many cases, they don't. Peering is often the very last item on the "to-do" list of the network engineering staff. As a consequence, delays can be considerable when packets travel from one provider to another.

Finally, thus far the public Internet lacks an accepted way to give priority to certain packets over others. Approaches to doing so are in the works, since it is generally recognized that there is a great need for prioritizing IP traffic. But these have not been agreed upon, and implementation is some time away.

However, a great deal can be done today to make VoIP quality rival the traditional phone system.

Achieving quality in VoIP

Fortunately, there are proven ways to ensure call quality. Here are the key steps.

[Budget for your bandwidth requirements.](#) Be sure to allow enough bandwidth for all of your applications, including voice, video and data. Then allow a comfortable margin to ensure that there is sufficient bandwidth available during peak periods. Appia has a simple tool you can use for bandwidth budgeting. Bandwidth prices are coming down all the time and sometimes the easiest solution is simply to have more bandwidth available than you ever need.

[Choose your ISP carefully.](#) Uptime and the quality of support an ISP provides are important. But be sure you also know what the ISP's bandwidth utilization is at peak traffic times, and most importantly, whether the ISP's peak coincides with the busiest times of your day.

[Base your choice of an ISP also on its backbone provider.](#) The key determinants of call quality are:

- Latency - the time it takes for a call to travel from the originating telephone to the terminating telephone. Unless this time is under about 300 milliseconds roundtrip, call quality will be inferior to the regular telephone network.
- Packet loss - packets contain the information of the conversation. If IP packets are lost, information is lost and conversations will break up.
- Jitter, or packet arrival time variation - latency may average 300 milliseconds or less, but if it varies widely, as it does often on the public Internet, conversations will sound uneven and may even be unintelligible.

Check the latency, jitter and packet loss statistics of the backbone provider. A key indicator of how the network is likely to perform is capacity utilization during peak traffic times. Congestion increases during peak times and congestion contributes most to call quality problems.

[Think "end to end"](#). Unless you prefer to use a private network, it is essential to keep all of your traffic on one backbone. When taken together with the other steps recommended here, this is the best way to avoid delay and packet loss because it avoids peering.

[Implement packet management tools in your LAN.](#) Several good tools are available, and while they are not inexpensive, they can pay for themselves in bandwidth savings and, more importantly, in satisfied users.

These tools enable you to set priorities among packets so that time-sensitive packets for voice and video receive a higher priority than data packets.

[Employ network management tools to monitor and manage the performance of your network at all times.](#) Without these tools, you cannot possibly spot problems and

correct them, either on an immediate or longer-term basis. Such tools are inexpensive, even free (a powerful example is MRTG), and are easy to implement.

Conclusion

All voice traffic will one day travel along with other forms of data on IP networks. This is a reality that is not disputed by even VoIP's most pessimistic detractors.

The idea of convergence is powerful, beautiful and elegant. It has captured the imagination of such visionaries as George Gilder, who speaks poetically about it in his best selling book Telecosm:

"When anyone can transmit any amount of information, any picture, any experience at any time, instantaneously without barriers of convenience or cost, the resulting transformation becomes a transfiguration."

Clearly, convergence is not right around the corner. The transition will be faster once IP networks can reliably prioritize traffic and once IP network operators have gained more experience. Until then, VoIP operators need to recognize the limitations of the transport mechanism they use and take steps to overcome its weaknesses.

What's the result when you do? Our experience is that if VoIP is implemented as explained above, the quality of calls is at least as good as the regular telephone system, and in many instances, even better.