There are many factors to take into account when providing VoIP telephony.

The Definition of a Packet is:

Data sent across the internet is broken down into parts of a certain size (measured in bytes). These are called packets. Each packet carries the information to help it to its destination. The packets are then sent off to their destination by the best available route. When they arrive at their destination they are collected and put back together in the order they were sent. Packets are also referred to as frames, blocks, cells or segments.

The Definition of Latency is:

* The time it takes for a packet to cross a network connection, from the sender to the receiver.

* The period of time that a packet is held by a network device before it is forwarded.

Problems associated with Latency

Due to the way that IP networks are built, specifically the internet, there a many ways in which packets of information can get lost resulting in latency and poor service levels. “Low-Bandwidth” applications, such as email, may not suffer with small delays but VoIP applications will translate these delays into a major nuisance. With VoIP, latency in a two-way conversation is very undesirable. To achieve high quality voice the maximum desired one-way latency is 150ms. Due to Network congestion, packet loss tends to be a key factor in lost voice signals, as well as, unreasonable delay and jitter.

Packet Loss and Packet Delay

Packet Loss is caused within routers. Errors in routers can, and will, occur if a router is overloaded. This will result in the router dropping packets; in IP telephony Packet loss is unacceptable. The performance of an IP call will suffer greatly if a packet loss reaches more than 5 percent. Packet Delay can wreak havoc in latency-sensitive application such as VoIP. Packets transmitted over IP networks are passed through routers. Delays can be introduced in the data network due to router configuration, network capacity and performance and load on the equipment. It is known that each router that handles the Packet will introduce a delay of 10ms.

Jitter Buffer

There are many components for latency, one of which is the Jitter buffer.

The Jitter buffer is used to collect the packets at the gateway to the PSTN or from the receiving VoIP phone for the outbound voice signal. An advantage of utilizing the jitter buffer is that it may reduce packet loss by simply increasing the size of the buffer. However this in itself can lead to extensive latency

Vocoders and Codecs

The vocoder is the device that translates analog and digital voice signals.

Voice encoders/decoders (codecs) are used to compress the voice data before transmission across the IP network. These in themselves will introduce a small delay, often between 15 and 45ms dependant on the Codec chosen for the call.

The Codecs that VoIP Advantage support are:

G711, There are two variations of this standard a/law and u/law. The G711 algorithm encodes non-compressed speech streams running at 64 kbps (approx 80 kbps when IP overhead added). Because the G711a/law codec is used by the European public telephone network, using this codec reduces the amount of transcoding required to pass the call from end point to end point and is therefore the preferred codec for best quality VoIP calls when connected to a UK network. G711u/law is used by the US network.

G729, There are two variations of this standard G.729 and G.729 Annex A. The G729 algorithm encodes non-compressed speech streams running at 32 kbps.

This codec tends to be used in environments where bandwidth is an issue but should not be considered a permanent fix for this problem. In order to measure the quality of the vocoders there is a measuring system called the Mean Opinion Score or MOS. The MOS functions on a scale type measuring system from low to high (0 being low and 5 being the highest). There are many software packages available that will provide MOS scores based on a series of set tests that will use a known connection and a specific sound file to analyse your network performance for VoIP services. For example http://www.testmyvoip.com/ by Brix Networks.

Quality of Service

Quality of Service (QoS) refers to control mechanisms that can provide different priority to different users or data flows, or guarantee a certain level of performance to a data flow in accordance with requests from an application/program. QoS technology is widely used for VoIP as it enables you to allocate priority to your voice data. QoS ensures voice packet delivery without packet loss, within your own network environment. Once the data is outside of your network environment there is no current way to enforce QoS.

Internet service providers

The VoIP service relies heavily on the internet connection. The quality and overall reliability of the phone connection is entirely reliant up the quality, reliability, and speed of the internet connection it is using. Poor internet connections can cause a large amount of latencies and can lead to significantly reduced call quality and cause certain problems such as echoing. It is therefore recommended that where possible the Voice side of your data should be allocated its own internet connection keeping it separate from the data part of your network.

Conclusion

When looking at providing converged VoIP/IP Data networks to your home/company, there are many factors to take into consideration. From your Broadband supplier to the equipment you use to provide your internet connection, and the hardware/software used to provide the VoIP element of your network.

Your goal should be to optimize your VoIP system by providing High Bandwidth, low delay, low packet loss and controlled jitter and latency.

Pre-qualifying your network connection by using some bandwidth/VoIP quality testing tools will help you to eliminate any problems before or after installation.

Introducing QoS should be high on your list of considerations when implementing a converged VoIP/IP data solution.