Voice over IP (VoIP)

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Introduction

Dating back over 100 years, traditional voice networks and the telephone have become a integral part of modern society. In fact, it is not unusual, even in remote parts of the world, for people to feel that they are entitled to basic telephone service. Obviously, the telephone and the associated networks are a large part of modern communications and technology. However, in recent years, data networks have been growing at a tremendous rate, largely due to the growing Internet. According to some experts, data traffic is predicted to soon exceed traditional voice traffic. As a result, more and more companies have become interested in implementing VoIP. But what exactly is VoIP and how does it work? Also, what are the benefits of VoIP?

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What is VoIP?

VoIP, or Voice over IP, is an application that enables data packet networks to transport real time voice traffic. It consists of hardware and software that allows companies and persons to engage in telephone conversations over data networks. According to an article written by techguide.com, "VoIP can be defined as the ability to make telephone calls (i.e., to do everything we can do today with the PSTN) and to send facsimiles over IP-based data networks with a suitable quality of service (QoS) and a much superior cost/benefit." It is also known as Internet Telephony. However, the latter term is often used in reference to calls made over the public Internet, and VoIP is often used to refer to calls made on a private network.

The traditional voice network, or POTS (plain old telephone system), uses circuit switching techniques. This means that a particular communications uses a dedicated path for the duration of the call. Although this provides a very reliable connection for voice transmissions, it makes very inefficient use of bandwidth. On the other hand, data networks generally uses packet or cell switching technologies. These use Statistical Time Division Multiplexing (STDM) in order to dynamically allot bandwidth to a particular stream of data, based on its requirements and the requirements and demands of other data on the network. This provides for much more efficient use of available bandwidth but can create problems for voice traffic, which is very sensitive to delay. Because each packet is individually routed across the network, this makes packet switching networks inherently less efficient in dealing with voice traffic and poses a number of challenges to a quality voice transmission. These include: packet loss, delay(echo), jitter(variable delay) and unreliable and out of order packet delivery due to the connectionless nature of packet networks. So, then, how does VoIP work, and how does it overcome these obstacles in order to provide reliable, quality telephone conversations?

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How does it work?

In order to deal with these issues and provide a voice service with a reasonable measure of quality, there are many techniques that are employed in order to deal with network congestion and delay by making better use of bandwidth. These bandwidth saving schemes include prioritization, fragmentation, jitter buffering, voice compression, silence suppression and echo cancellation. This is where the various protocols, such as H.323 come in, as standards are being set to control the quality of voice transmissions on a data network.

Prioritization techniques are related to QoS (quality of service), which is a method of guaranteeing throughput for certain traffic on the network. This can ensure that voice traffic on a data network is given high priority. This prioritization can be based on location, protocol or application type. Protocols used to ensure this QoS are RTP (Real Time Protocol) and RSVP (Resource Reservation Protocol).

Fragmentation divides packets into smaller fragments so that their priority can be ensured. This can help reduce the overall delay of voice delivery. However, on IP based networks, this can create extra overhead because of the large size of IP headers (20 bytes). So although necessary, fragmentation alone cannot ensure the reliable delivery of real time voice applications.

This is why compression is also necessary. Various codecs (coder/decoder) standards have been implemented. ITU G.723, which provides for 3.1 kbps bandwidth over 5.3 and 6.3 kbps channels has been adopted for use with VoIP. The ITU G.729 standard has been adopted for VoFR (Voice Data Convergence Glossary).

In IP-based networks, packets that belong to the same transmission (whether voice or data) do not always arrive with the same amount of delay. For example, packets 1-5 of a given data stream may all arrive with a consistent amount of delay between each packet, but the delay between packet 5 and 6 may be twice as long. This variation in delay is referred to as "jitter." As a result, voice transmissions will sound unnatural. When the next packet in a voice stream does not arrive in time, the previous packet is usually replayed. However, this can create conversations that lack a natural quality. In order to handle this delay variability, a jitter buffer is established. This allows packets to be collected into a buffer and held there long enough for the slower packets to arrive so that they can all be played in proper sequence and in a natural voice flow. Although this can remove packet delay, this creates additional overall delay. According to Gil Biran, Vice President of Research and Development for RAD Data Communications, "the jitter buffer should fit the network's differential delay." This will provide for the necessary balance between packet delay and overall delay, allowing for voice quality transmissions.

In human telephone conversations, generally only about 50% of the full duplex bandwidth is used at any given time. This is because one person is generally listening while the other is talking. When you couple this with the fact that there are natural pauses, pauses for breath and between words, the total required bandwidth for a conversation is reduced an additional 10%. This means that there is between 50-60% of the available bandwidth that is not being used. Silence suppression techniques take advantage of this by detecting when there is a gap and then suppresses the transmission of these silences. This can result in more bandwidth being available for other transmissions. However, because these silences are necessary for the conversation to sound natural, the receiving device must interpret the lack of packets and re insert the silent spots into the output.

When the total end to end delay of a voice transmission is greater than 50 milliseconds, echo becomes a problem that can detract from the quality of the conversation. An echo cancellation unit solve this problem by performing echo cancellation on the signals. ITU G.165 or G.168 provide the standards and requirements for echo cancellation.

When dealing with data transmission on IP networks, TCP (Transmission Control Protocol) handles any packets that may be lost due to congestion or link failures by issuing acknowledgements and requesting retransmittal of lost packets. Although this works well for data, this method is not efficient for time sensitive information such as voice. In order to help ensure a quality voice conversation, packet losses greater than 10% are not tolerable (Techguide). For any packet losses under 10%, interpolation (playback of the last packet) can help maintain a continuous flow of voice with minimal distraction to the quality.

In order for different manufacturers to implement these various techniques and maintain interoperability, various standards have been recommended and approved. The ITU H.323 standard is a sort of umbrella standard that includes in its family the various standards for compression and call control. H.323 describes equipment that provide multimedia communications on networks that do not ensure a guaranteed QoS (such as IP based networks). There are also other protocols designed for VoIP client applications. These include SGCP, SAP, SIP, RTSP and SDP. You can find additional information on these protocols by using the links to Additional Resources on the left. In addition, I plan to do further research and provide a tutorial on these in the near future as well.

Obviously, enhancements to equipment and standards for VoIP are constantly being improved. You may have heard about VoIP but may have wondered whether or not it is a good solution for your company. So, then, what are the benefits of implementing VoIP?

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Benefits of VoIP

Many companies are seeing the value of implementing VoIP in their data networks for many reasons. These include:

- Cost reduction - low cost phone calls
- Convergence of data/voice networks - unification
- Simplification and consolidation - centralized management

As data networks continue to grow, implementing VoIP can be a very appealing option that can allow for reduced costs and provide for greater flexibility. In addition to replacing internal voice networks at large corporate offices, VoIP can be used to connect various branch offices through existing WAN links. This gives companies an alternative to the PSTN that can continue to grow and be scaled to fit their needs.

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Conclusion

As data traffic continues to increase and surpass that of voice traffic, the convergence and integration of these technologies will not only continue to improve, but also will pave the way for a truly unified and seamless means of communication. Implementing VoIP can provide significant benefits and savings to your company. To find out more about this exciting new technology and to see how it can be implemented to fit your needs, please feel free to contact Donald G.W. Lau, an Applications Engineer at ComTest Technologies.

ComTest Technologies proudly represents a number of VoIP manufacturers. These include, but are not limited to, the following:

- Oki
- Alcatel
- Vive
- more...

To learn more about VoIP, visit these links:

- Bandwidth Management and QoS
- Voice over IP Overview at Protocols.com

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References

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