VoIP Primer: Introduction to Voice over IP

In order to understand and appreciate some of the business and technical issues driving the popularity of carrying voice over the Internet Protocol (IP), you need a basic framework of how telephony works. This article provides a brief overview of the basics so that you can explore why Voice over IP (VoIP) is a smart solution in today's network environment.

As communications has evolved over time, we've reach the point at which we can pick up the phone and call someone on the other side of the world. The delivery of information, or an idea, is nearly instantaneous. Our culture has shaped us to expect things to happen very quickly, and we have become impatient about delays in delivering information in any form.

Alexander Graham Bell made the first telephone call on March 10, 1876, but at that point, nobody fathomed the impact this technology would have on us all. AT&T put the first transcontinental phone lines in service in 1915, and communications again changed forever.

To transmit a telephone conversation, you must send sound over long distances. In a phone call, this sound is voice, but it could just as well be music or some other sound. The public switched telephone network (PSTN) has been optimized to carry voice traffic for more than 100 years. Today, separate voice and data networks may be too costly. VoIP allows integration of voice and data onto a single network.

> ☞ *IP Telephony Demystified* (McGraw Hill) contains more detailed explanations of many of the concepts touched on in this article.

Telephony Basics: Transmitting Voice

The PSTN is a circuit-switched network. Every time you make a phone call, telephone switches set up temporary connections between parties involved and establish a circuit that is used for a voice conversation. The Internet is a packet-switched network. Information, in digital form, is carried in packets over many diverse paths, or routes. The information in these packets can be anything from email messages to voice telephone calls. All that is required is that the message be encoded in some binary form for transmission.

Our vocal chords are used to produce sounds. Sound is a series of vibrations in air pressure. These sound waves travel through the air, much like ripples on a lake travel through water. Just like ripples in water, the farther they get from the source, the weaker they become.

Early telephones converted the physical energy of voice sound waves into electrical impulses that could be sent over copper wires. As electrical energy can use repeaters or amplifiers to regenerate or boost the signal, telephone calls can be carried over great distances. At the other end, the electrical signal is converted back to sound waves at the telephone handset. This function in the telephone set is referred to as a transducer. For a basic telephone call, the network establishes a connection between parties, encodes the signals, then transmits them. It decodes the signal at the receiver. When the call is finished, the network then disconnects and updates usage and billing records to bill for the call (see Figure 1.)

Figure 1: Phone calls on the PSTN.

Signals that are continuously variable are commonly thought of as being analog, whereas discrete signals are considered digital. For example, water flowing in a river is a form of analog signal. Flow meters can measure the variable rate at which the river flows. A typical light switch, in contrast, controls a digital signal. The light has only two states—on or off. Digital signals typically have predefined levels that represent specified conditions.

Analog signals are represented by a waveform or sine wave as Figure 2 shows. A digital signal can only represent predefined values. You frequently deal with a binary system in networking. There are two predefined levels. These are represented in binary math as a 0 or 1. A digital system can have a large set of predefined values. The alphabet is made up of 26 predefined values represented by the letters. A digital system doesn't necessarily have to be a binary system.

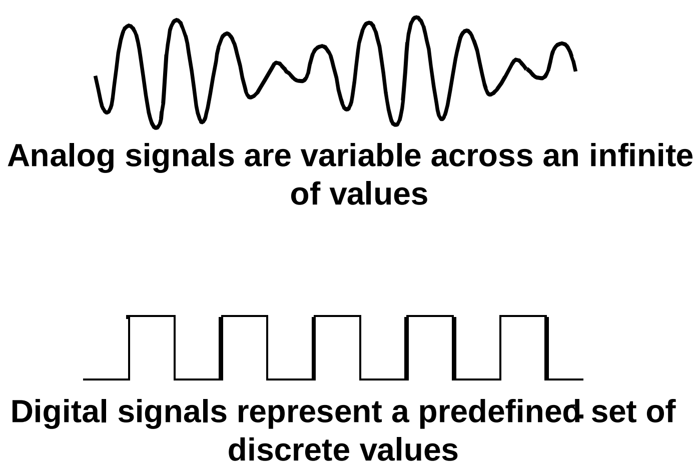

Figure 2: Analog and digital signals.

The square wave is not a sine wave but has enough similarities to suggest a relationship between the two. The relationship was completely described by mathematician Jeanne-Baptiste Fourier.

Packetizing Voice

To transmit voice over a data network, the voice conversation must be packetized. To transmit the conversation in packets, you need to digitize it. Voice digitization is not a new technology. In the 1960s, the phone companies began using a digital carrier system called T-carrier. Initially used as a trunking technology between phone company central offices, it's what is commonly called a T1 circuit today.

As your voice represents an analog signal, you must first convert it into digital format. This conversion is commonly accomplished through a technique known as pulse code modulation (PCM), using a coder and decoder (or codec). Using PCM, analog voice conversations are sampled 8000 times per second. A technique called pulse amplitude modulation (PAM) is used to convert each sample into one of 255 possible 8-bit words. Through a process of compressing and expanding (called companding), noise is reduced to help eliminate background hiss and changes in volume.

There are several voice processing standards that were developed by the International Telecommunication Union (ITU, formerly the CCITT) to standardize encoding of digital speech. In the PSTN, the G.711 standard is used worldwide.

> 📖 More information about encoding standards is available at the ITU Web site at http://www.itu.int/.

Understanding these encoding schemes isn't crucial to the process of packetizing voice for VoIP service, but provides useful background to understand the telephony processes that let you merge voice and data onto a single network infrastructure.

Connection Oriented vs. Connectionless

The PSTN is a circuit-switched network and is oriented to connections. Each telephone call represents a connection. Packet networks can be connection oriented, but a connectionless network requires no setup. Connectionless networks can't necessarily guarantee that packets will be delivered in the order they were transmitted. As packets can take different routes through the network, they can arrive at different times. In a VoIP network, this means the network nodes and end devices need resources to store packets until enough have been delivered that they can be reassembled into a message for delivery. This action all happens in milliseconds so that people don't notice a difference in call quality.

Packet Switching—Store-and-Forward

Although the telephone network uses circuit switching, there are other switching technologies that are quite effective. One example of store-and-forward switching is airline travel. Passengers travel between cities that have airports and can switch en route to reach their final destination. This network topology is referred to as "hub and spoke." The system has switching points, or nodes, around the world. To get from the starting point to a final destination, a passenger may need to switch routes from one plane to another at a node site along the way. In some cases, passengers are delayed (or buffered) while waiting for the next flight. With luck, all passengers are forwarded to their final destinations.

In data networks, links between switches are shared on demand. Routing calculations determine the best path for packets to take toward their destinations. Data networks are described as "bursty" in nature. That means that traffic patterns and volumes are sporadic and often unpredictable. This unpredictability doesn't pose a problem to Web or email traffic. These types of traffic are not real-time, and slight delays don't cause problems. Voice traffic is real-time in nature. Delays of more than 250 milliseconds (ms) or variations in timing degrade voice quality and can make the network incapable of carrying voice traffic.

In packet networks, you overcome this circumstance by breaking large chunks of information, such as a telephone call, into smaller IP packets that can take different paths through the network. In a store-and-forward network, each packet must carry the source and destination address of the information contained in the packet payload. Store-and-forward networks use statistical multiplexing techniques, such as first in, first out (or FIFO), so they've always been well suited for traditional, bursty LAN data traffic.

Today, packet networks carry huge volumes of data with far greater reliability and performance guarantees than in the past. Quality of service (QoS) to support call quality can be implemented through assessing the network requirements before implementing VoIP, adding bandwidth capacity where necessary, and implementing widely accepted methods for quality assurance such as Multi-Protocol Label Switching (MPLS) across a wide area network (WAN).

Replacing the Enterprise PBX with VoIP

Companies have long realized that maintaining separate voice and data networks is an expensive proposition. The telephone system, or PBX, is a large financial investment. As LAN and WAN technologies have become simpler to manage and VoIP solutions more widely available, it makes good business sense to integrate the two networks and consolidate resources. The benefits of this consolidation are numerous:

- A single wiring scheme can eliminate separate voice and data wires. Although most companies long ago standardized on Category-5 cabling, voice and data still often compete for cable resources.
- The PBX can be a very expensive investment. Integrating voice services into multi-function routers and VoIP gateway servers can reduce investment cost significantly.
- Administration skills can be consolidated into a single work group. Large organizations can benefit from consolidating internal support organizations.
- Large companies with many locations can simplify billing by integrating voice and data into a single network. This integration leads to simplified billing from providers, which means fewer company resources are needed to validate bills for payment. For a large enterprise, the impact can be significant.

A VoIP solution for PBX replacement might be as simple as implementing VoIP phones and services internally on the corporate network, with a single gateway to the PSTN to provide a connection for calls outside the company. One advantage to this approach is that simple IP phones can plug directly into the Ethernet LAN to provide telephone service within the corporate network.

This solution provides basic telephone service. Telephone calls are routed over the LAN for internal calls or directed to the gateway to reach parties on the PSTN. Figure 3 shows this simple implementation of a PBX replacement.

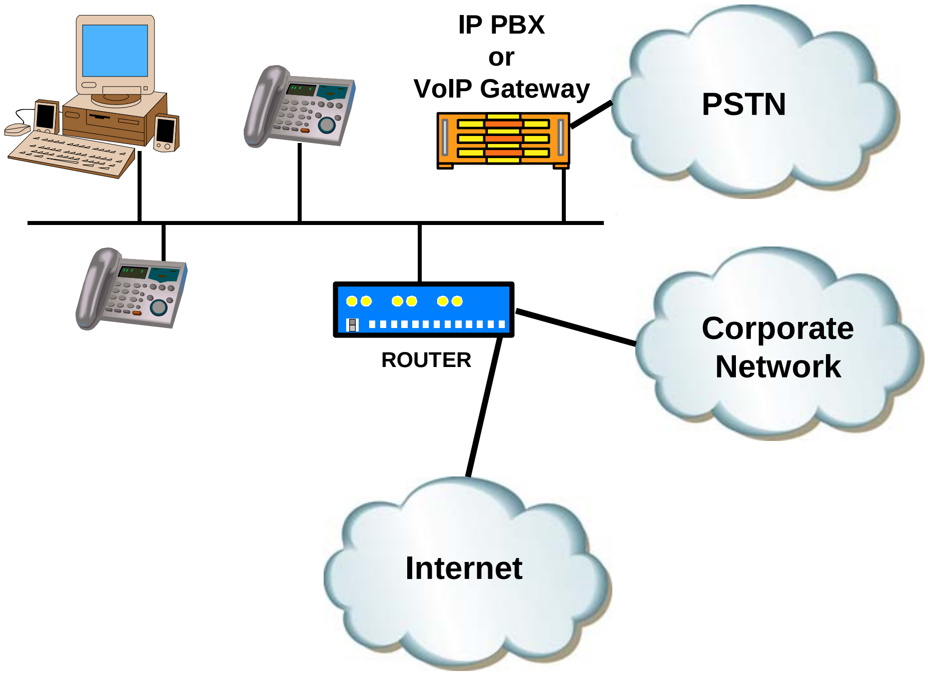

Figure 3: A simple PBX replacement example.

As VoIP systems have matured, standard features such as call waiting, placing calls on hold, conference calling, transferring calls, and music on hold are now common features in most business VoIP solutions. Voicemail systems are now commonly digital systems built on IP networked servers that integrate well with VoIP solutions.

The Converged Network

One of the greatest benefits a company can gain from VoIP is an evolution well beyond the simple consolidation of the network onto a single wiring infrastructure. When networks converge, integrated services can be realized using new approaches. In a fully converged network, the integration of computers and telephony leads to customer service opportunities.

Customer data, collected over both telephone and computer systems, can enable new approaches to customer service. Call centers can be geographically dispersed, even integrating telecommuters working from home. A single customer service center can handle both telephone calls and Web services.

Imagine customers clicking a Help button on a Web site that sets up a VoIP phone call to a customer service representative. With VoIP in a fully integrated environment, the customer service representative might actually get a copy of the Web screen that the customer last visited, complete with history of what this customer had been trying to do. How would you react as a customer if you clicked Help and the service representative came online and said "It looks like you're trying to find information about our new product *X*. I'm sorry the Web information wasn't more helpful. How can I help?" (see Figure 4).

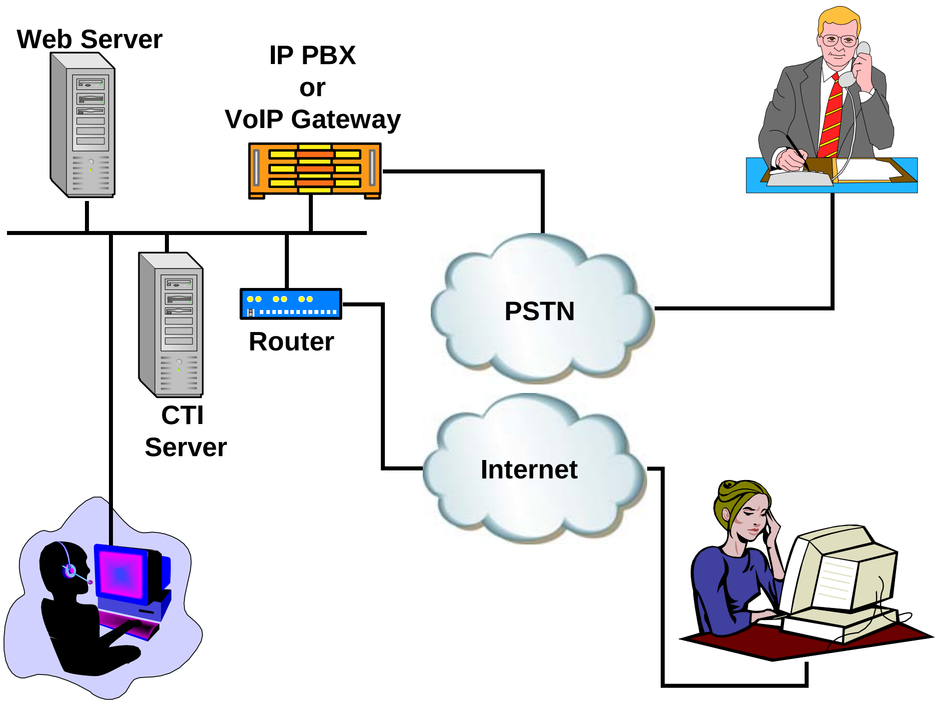

Figure 4: Computer telephony integrated into a converged network.

Computer telephone integration at its finest and most developed can provide a powerful competitive edge to customer service organizations.

Summary

Network evolution and corporate culture take time and effort to change. They evolve continually. Although no company can jump from legacy telephone systems to a fully integrated voice and data network overnight, VoIP provides a point of entry for your company to begin moving into a competitive converged network.