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"Leading the Conversation"

The Essentials Series

The Business Value of SIP VoIP and Trunking

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by Ken Camp

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VoIP in the Mainstream

Early man used pictographs on cave walls to express ideas. Since then, technologies have evolved to bridge the gap between humans and change how we interact. Communications have evolved as well. Today, we pick up the phone and call someone on the other side of the world. The delivery of our information, or ideas, 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.

Over the past 10 years, Voice over IP (VoIP), once seen as a niche, has become a mainstay of converged services. What began as a quirky technique for communications between PCs on the Internet has grown into a sustainable mainstream technology that fully supports consumers, enterprise business, and carrier-grade operations. Today, VoIP provides the network foundation for the complete integration of voice, video, and data services with enterprise business applications. To appreciate the technologies of today, we need to consider the evolution that brought us here. The development of communications has been filled with wondrous advances in our ability to share information and ideas.

The Evolution of Telephony

On March 10, 1876, Alexander Graham Bell made the first telephone call, and we were given a glimpse of a technology that would spread around the world and change how we communicate. The Public Switched Telephone Network (PSTN) has been growing ever since.

Some of the most significant events in the evolution of the telephone network weren't directly related to technology. When consumers were offered flat-rate pricing for local calling, usage of the telephone rose dramatically. In the mid-1960s, AT&T offered direct-dial long-distance service and for the first time users could call friends or family across the country without the intervention of a long distance operator. Again, this precipitated a dramatic rise in telephone calling patterns.

One of the key technologies that led to where we are today was the digitization of the telephone network. Telco central offices are connected by dedicated circuits called *trunks*. The Bell System led the telecommunications world for years and undertook a massive migration to convert these trunks from older analog circuits to a new digital technology. This migration was driven by the need to increase the density of network carrying capacity while striving to reduce network operating costs. T-1 circuits became a common trunking solution, providing 24 simultaneous voice paths between offices. This also allowed for switching or cross-connecting circuits carrying phone calls through multiple central offices more quickly and efficiently.

This T-1 methodology still exists today, but rather than time division multiplexed (TDM) circuits carrying 24 voice calls, many of these circuits now carry IP-based traffic. Digitizing the network trunking taught the industry a great deal about digitizing and packetizing voice calls. Although VoIP may be a new technology to many, the practice of digitizing and packetizing voice traffic has been in use for many years.

VoIP Collides with Telecommunications

As the lessons learned in digitizing voice traffic advanced, the idea of making a phone call over the Internet continued down a parallel path. Standards began to emerge and overlap. The telecommunications H.323 umbrella protocol standards for voice, video, and collaboration tools were widely adopted by the emerging VoIP community working on Internet technologies. Session Initiation Protocol (SIP) gained wide acceptance over 10 years, and today has become one of the most commonly used protocols.

Early fears that the Internet might reduce telephone network traffic have long since been allayed. Over time, we've seen that the Internet and the PSTN have not been able to replace the other. But the two networks continue to move closer together. Each adopts and incorporates technologies of the other that allow for seamless integration. Today, many callers on the PSTN are using traditional phones and never realize that their call is being carried by a VoIP service somewhere between the caller and called party.

VoIP as a Mainstream Building Block of Voice Service

For the telecommunications carriers, VoIP became a fundamental component of the voice service architecture. IP networks, although not providing the same inherent Quality of Service (QoS) level as a dedicated circuit on the PSTN, provide more resilience, bandwidth capacity, and versatility than the traditional TDM circuits did in the past.

Circuit Switching

Circuit switching is the legacy method used to establish a dedicated electrical (or optical) path between devices. This path or circuit is established for the duration of the telephone call and is dedicated to that call. These resources in the network cannot be used for other calls or by any other user until the call is completed and the resources are released and available.

Because networks cannot be designed to support every possible telephone call at the same time, the switches are designed as "blocking" switches. Thus, when all available resources are in use, callers will experience queuing delay or blockage until resources become available.

The delay through the network once a connection is made is minimal. Most of the telephone network has been designed to provide about 55 milliseconds of delay over the circuits that are established.

Because circuit switching guarantees a dedicated path that cannot be shared, it requires a significant engineering effort to locate, reserve, and connect the necessary resources through the network. This requirement drives up the cost of a connection and causes some delay in the setup process. As a result, circuit switching is more economical for connections of a longer duration such as a voice telephone call where parties may talk for 3 or 4 minutes. It works best when network utilization of the network is high, providing a usage level that keeps the resources busy but not overloaded.

IP networks have enabled the carriers to reduce central office costs, incorporate broadband services such as DSL for Internet access, deliver video, and expand into new services. IP networking over packet switched networks delivers benefits to everyone using the network.

Packet (Store-and-Forward) Switching

There are many ways to provide switching without the use of dedicated facilities. One example of a store-and-forward switching network is the subway system in New York. Passengers can travel between any of the subway stations along the route. The topology of this network is referred to as a “hub and spoke” topology. The subway has many switching points or nodes. To get from one location to another, users might have to transfer from one line to another at one of these nodes. At the hub nodes, passengers (the traffic) might have to wait in a buffer (be stored) until the next available train arrives so that they can move (be forwarded) to their destination. Just as passengers encounter delays in waiting for a train to arrive, and sometimes queuing delays when the trains are fully loaded, a store-and-forward type network provides service that has very different characteristics from a circuit switched network, which would be more similar to a New York taxi; dedicated to a passenger for the duration of the trip

In a data network, even the links between the switches are shared on demand. Switches perform routing calculations to determine which link to send the data onto, and then the data is placed in queue for that link. Resources are allocated on a first-come, first-serve basis, and there are no guarantees that the next leg of the path will be available upon arrival. Delays in queuing can cause data to sit in buffers. In a data network, this means that the delay through the network can be sporadic and unpredictable.

Because of this unpredictability, large blocks of information aren’t well suited to this type of network. Large blocks of information have to be broken into smaller chunks in order to avoid degrading performance of the network. When we think of a 4-minute telephone call, we are really thinking of a very large block of data.

In a store-and-forward network, each block of data has to carry some form of addressing information that the switches can use to determine the location of the final destination. Without this, the information can never be delivered to the recipient.

Data applications are often described as being “bursty in nature,” meaning that there may be lapses or pauses between transmissions. Unlike a voice call, which is a real-time interaction between two people, a data connection is often an interaction between two computers without a person directly involved. Since store-and-forward, or packet, networks use statistical multiplexing, or first in, first out (FIFO) methods, this type of network is better suited to a bursty type of traffic, such as data.

Packet switching is the most common form of store-and-forward switching in use today, with routers being a perfect example of a store-and-forward switch. Packet switching breaks blocks of information into a pre-defined size or size range. This process of packetization creates some overhead, as each packet must have addressing information. Error checking can be performed on a per-packet basis, and if errors occur, only the corrupted packet needs to be retransmitted. This gains some efficiency in the network, as long as messages do not need to be repeated entirely if an error occurs.

This advance from circuit switched voice to VoIP has provided huge benefits to enterprise business. Ten years ago, the term *convergence* referred to the idea of integrating voice and data services onto a single high-capacity circuit from the service provider to the business premise. Today, convergence has brought voice and data together on a single cabling infrastructure and network within the business, all based on IP. This integration of voice and data has set businesses who have embraced the change at the leading edge to now be able to integrate network services with enterprise business applications for improved efficiencies, reduced cost, and strategic business process improvement.

The evolution of convergence encompasses both Unified Communication and Communications-Enabled Business Processes. CEBP is an important way of thinking about the simplification and automation of communications as services on the network in conjunction with enterprise business applications. In a recent paper, Forrester Research defined CEBP as “business processes and applications tightly integrated with unified communications technologies to enable concurrent or consecutive communications among customers, suppliers, and employees within the context of business transactions.” Those organizations that have already migrated to VoIP, or are doing so today, are positioned to take the greatest advantage of complete integration.

VoIP Penetration: Then to Now

VoIP technologies started deploying in the late 1990s. There were many early market efforts from a number of different carriers and solution providers. Some focused on consumers, and some on enterprise business. VoIP didn’t mature or achieve critical mass overnight.

Today, looking at quarter-by-quarter numbers from industry analysts, it appears at first glance that the market is slowing, yet overall penetration is on the rise. The incumbent, traditional telecommunications carriers are clearly losing landline customers at a steady rate. Consumers are migrating to VoIP solutions from cable companies and mobile services (both sectors clearly on the rise).

On the consumer side of the equation, according to iLocus, a Business-to-Business (B2B) market research firm that tracks the VoIP market, a recent bulletin noted the following:

> In 1Q08, vendors shipped a total of about 7.9 million VoIP Subscriber Feature Server licenses for deployment in service provider networks generating $144 million in revenue. The number of lines is down by 19% Q-o-Q. The 3Q07 and 4Q07 quarters however were unusually high growth quarters for VoIP Subscriber Feature Servers. If the 4Q07 seasonality in particular is normalized, there is a nice sequential growth.

iLocus does not track IP upgrades to TDM ports, which encompasses a huge portion of the enterprise market space. They focus on tracking Class 5 next generation network deployments, including:

- VoIP hosted telephony implementations
- New “greenfield” VoIP deployments
- Complete replacement of legacy switches with VoIP
- Extension of existing legacy networks with VoIP equipment in new geographic areas

Telegeography Research recently reported that as of the end of March 2008, 13.8 percent of US households, 27 percent of those with broadband, or 16.3 million consumer lines are using VoIP. When they look at VoIP provided by the cable companies, Telegeography reports the following numbers (descending):

Cable Provider	Number of Subscribers
Comcast	5.1 million
Time Warner	3.17 million
Cox Cable	2.46 million
Cablevision	1.68 million
Charter	1.08 Million

Telegeography's report cites that growth in the cable companies' subscriber base comes from the traditional old Regional Bell Operating Companies (RBOCs), who lost 17.3 million consumer residential phone lines during the period covered.

From a revenue perspective, Infonetics reported the following revenue projections:

Market Forecast	2008	2009	2010
IP Centrex	$153M	$200M	$255M
IP Audio Conferencing	$125M	$165M	$211M
Unified Messaging	$27M	$39M	$54M
Generic SIP Application Server	$34M	$30M	$22M

The VoIP Evolution Simplified

The traditional circuit switched PSTN is a connection-oriented network. The connection is the call setup process that establishes the circuit between the parties on a phone call. Packet networks, particularly those based on IP and Internet technologies, can be either connection-oriented or connectionless. In a connectionless network, no setup is required. Each packet carries sufficient addressing or routing information to allow it to be passed from node to node through the network.

Unless quality mechanisms are put in place, there are no guarantees for QoS, but there are no dedicated network resources. The resources of the network can be shared and used efficiently. Connectionless networks also don't inherently guarantee that packets will be delivered in the order they were transmitted. Packets might take different paths through the network and arrive at different times. Thus, the device at the recipient must have resources to store the packets until enough have arrived to reassemble the message for delivery. As you can see, packet networks provide a good technology for delivering short or bursty messages that don't require the overhead of establishing a circuit connection during call setup.

Summary

VoIP leverages all the knowledge developed by the technology sector over the past century. The legacy telephone service providers understand traffic engineering, busy hour requirements, call blocking algorithms, and every nuance of delivering high-quality voice services. The IP or Internet industry learned about bundling the intelligence of addresses within packets, digitizing content, building QoS guarantees into a network the inherently has none, and building a massive scalable integrated service network. VoIP brings the best of all this mature technology together in a solid, reliable foundation that gives enterprise business the strongest possible architecture to build the foundation of the mission-critical business network.