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WHITE PAPER

Introduction to Converged Networking

A technical briefing series on VoIP and converged networks

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Executive Summary

This is the first of six technical briefing papers that examine the concepts, operation and analysis of Voice over Internet Protocol (VoIP) and converged networks. The concept of a converged network – one that combines voice, data, and other signal transmissions into a single, higher-speed network interface – has been around for several decades. Advances in Internet-related technologies in general, plus the widespread acceptance of the Internet Protocol (IP), provide the driving factors that enable these infrastructures.

To better understand the concepts of the converged network, one must first see the differences between voice and data network operation. To establish a voice call through the Public Switched Telephone Network (PSTN), the end user sends the network call setup information, which triggers the network to establish a path from source to destination. Once that path has been established, its characteristics should remain static for the duration of the call. This facilitates the transmission of real-time information, such as voice or video. Because a circuit from source to destination has been established, this is called a circuit-switched connection.

In contrast, most data transmissions, such as those over the Internet, are packet based. With packet switching, the end user drops a small unit of information (or a packet) into the network. Within the packet are source and destination addresses, which the network uses to forward that packet to its intended destination. The absence of an established (or static) path, with its circuit-like qualities, makes most packet-switched connections less reliable than circuit-switched connections. Therefore, the transmission of real-time information, such as voice or video, becomes more challenging.

The converged network, which combines voice and data networks into a single, packet-based network, holds the promise of great economic benefit. As a result, there has been tremendous interest among network managers in evaluating and deploying these systems, and market statistics confirm this enthusiasm. Potential benefits include reducing the costs of maintaining two separate networks, reducing the costs of communication circuits through economies of scale, and support for new electronic commerce applications – such as voice-enabled Web sites – that could improve revenue potential.

However, the Internet Protocol, which was designed to support data, is not sufficient in itself to transmit real-time traffic such as voice and video. Additional protocols must be added to the IP-based network infrastructure to fill in these gaps. Fortunately, a number of international standards bodies have collaborated on protocol suites that address these shortcomings. The additional papers in this series will address the operation of converged networks and will illustrate their effectiveness.

1. What is a Converged Network?

The concept of a converged network, one that has the capabilities to transmit voice, data, fax, video, or some combination of these signal types (or media), is not new. Many of these ideas were founded during the development of the Integrated Services Digital Network (ISDN), developed by the International Telecommunication Union – Telecommunications Standard Sector (ITU-T), previously known as the Consultative Committee for International Telephony and Telegraphy (CCITT), in the 1980s. The concepts at that time were focused on integrating voice and data applications into a common transmission system – typically telephone circuits – which were enhanced to support the higher bandwidths that the data transmissions required. Thus, having a single telephone line that could support voice, data, and telemetry services, such as reading your electric meter, was envisioned. But as the ISDN developments migrated to the implementation phase, the service found less than overwhelming acceptance. However, the concept of using one pipe to transport information from multiple, dissimilar sources was still sound.

In the 1990s, network managers witnessed the tremendous growth of the Internet, and new applications, such as the World Wide Web, emerged. These applications were designed around the Internet Protocol, or IP, which had traditionally been used to support file transfers, remote host access, and email. The popularity of this protocol spread, and, before long, the operating systems of virtually all host processing systems, from the smallest Personal Digital Assistant (PDA) to the supercomputer, were enhanced to support IP. Thus, IP became a de facto communication platform.

These technology enhancements from the 1990s propelled the networking industry into the next millennium. Unfortunately much of this enthusiasm became tempered with the economic realities of tightened budgets. Now network managers had another issue to address: how can new IP-centric applications, such as electronic commerce, be deployed while still keeping a keen eye on the budget?

That challenge – reducing the costs of networking infrastructures while at the same time rolling out new revenue-generating applications – has provided network managers with an added incentive to revisit the idea of the converged voice/data network.

But before we look into the benefits of the converged network, let's first examine the technical basis for these two systems, voice and data, that will be combined.

2. Enabling Technologies and Standards

Communication networks can provide one of two different types of network services: connection-oriented service (CON) and connectionless service (CNLS). The typical model for a connection-oriented network is the Public Switched Telephone Network (PSTN). When the end user takes the phone off hook, they notify the network that service is being requested. The network then returns dial tone, and the process of establishing the call through the various switching centers begins. Once the call path has been established, some transport characteristics of the path are also determined. These include the sequential nature of the transmission (first in, first out), fixed delay along that path, and so on. Because a circuit from source to destination has been established, this is called a circuit-switched connection.

Another important characteristic of circuit switching is its reliability. For example, the PSTN has traditionally operated from an industry objective of 99.999% uptime, which is sometimes referred to as the "five nines", or "five nines reliability". This came from a standard used in central office switch design that specified downtime should be less than two hours in forty years of operation. If you work out the math, 40 years equates to 350,400 hours, and 2 hours out of 350,400 hours yields a raw number of 0.00000571. This downtime (or failure) number can be converted to uptime (or reliability) by subtracting from 1, and then converting it to a percentage:

Network reliability = 1 – 0.00000571 = 0.99999429, or 99.999429%

Thus, we see further evidence for why CON networks in general, and the PSTN in particular, are called reliable networks.

In contrast, the typical model for a connectionless network is the postal system. The end user fills out an envelope with both source and destination addresses, and then simply drops it in the mailbox. Through the miracles of routing technologies, that envelope finds its way to the destination — or at least it is supposed to. This is the concept upon which packet switching is based. A large amount of data is broken into smaller units, called packets, for transmission through the network.

Unfortunately, the transport characteristics of a CNLS network are not as reliable as the CON case, and no clear statements about the transport characteristics, such as delivery reliability, packet sequence, or network delay, can be made. This is where the term best effort service is used — the network will diligently attempt to deliver the information that the end user transmitted, but there are no guarantees of success.

These reliability issues must be addressed for implementations to be successful. Many organizations have been hard at work to develop standards and protocol suites that address this challenge. At the forefront of these efforts are the European Telecommunications Standards Institute (ETSI), the Internet Engineering Task Force (IETF), and the ITU-T. Here are a few examples of the work that has been produced:

- **ETSI:** Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON).
- **ITU-T:** H.225.0, H.323, and H.245 standards for multi-media communications.
- **IETF:** the Session Initiation Protocol (SIP) for multi-media communication over IP-based networks, plus the Resource Reservation Protocol (RSVP), Real Time Protocol (RTP), Session Description Protocol (SDP), and others that facilitate real-time communication over IP-based networks.

For network managers, published standards provide greater assurance that products from different vendors will interoperate within an enterprise environment. And as we will see in future papers in this technical briefing series, these new standards and protocols will be required to augment the functions of IP. In addition, extra hardware, such as gateways and gatekeepers, will also be deployed to further meld separate voice and data networks into a cohesive system, as shown in Figure 1.

3. Migrating the Legacy Network

The interest in migrating existing legacy networks to converged networking technology is coming from many directions. The early adopters of VoIP technologies were considered by many to be hobbyists looking to experiment with transmitting voice and video over the Internet. However, their successful accomplishments demonstrated the viability of these technologies and quickly grabbed the attention of both the voice and data networking manufacturers. For example, developers of networking hardware, such as routers and switches, realized that they needed to gain expertise on the voice side of the business, and quickly moved to acquire talent in WAN signaling, trunking, and long-haul circuit operation. In contrast, developers of premises switching systems, sometimes called Private Branch Exchanges, or PBXs, or the smaller Key Telephone Systems, or KTSs, realized a need for greater expertise in packet-switching technologies and moved to acquire talent in Internet and IP-related network operation. Thus, these two giant industries – voice networking and data networking – both moved to establish a beachhead within the other segment's stronghold.

Carriers also identified potential benefits of converged networks within their infrastructures. Many carrier backbones were migrated to high-speed packet-switched or cell-switched technologies during the development of Asynchronous Transfer Mode (ATM) in the late 1990s, making them ideally positioned for the next migration to an IP-centric network. As these carriers encourage their subscribers to use packet technologies for all their media – voice, data and video – fewer format conversions are required and greater overall efficiencies result. In fact, many new carriers, called Internet Telephony Service Providers (ITSPs), have deployed fiber optics throughout their networks with the intention of providing their subscribers with a common interface for all media transmission.

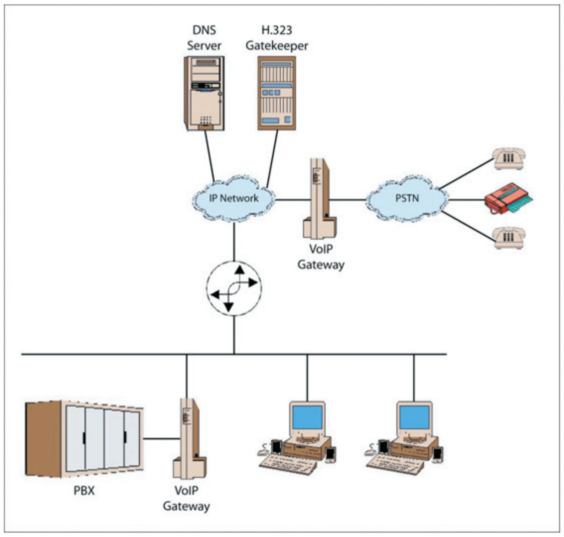

Figure 1. Converged Network Architecture.

Thus, product development interest in converged networks came from many directions – PBXs, LANs, WANs and carriers – and as a result, the vendor market has been growing at a very fast pace. This is good news for network managers, as the breadth of available product options for legacy system replacements continues to expand. Market research firm Infonetics Research, Inc. (Campbell, California) recently studied this market and published their findings in May 2005 in a report titled *User Plans for IP Voice: North America 2005*. Their research indicates that IP-based telephone systems, including PBXs and KTSs will continue to grow in the next two years, while the older Time Division Multiplexing (TDM) systems, analog lines and CENTREX services will decrease, as seen in Figure 2.

Infonetics also found that objections to adopting IP voice are diminishing over time, as the obsolescence of existing systems increase, and the entry price points of new systems decrease.costly to make material changes to the application. The emphasis therefore shifts to maintenance and incremental functional improvement, as opposed to proactive performance management and optimization.

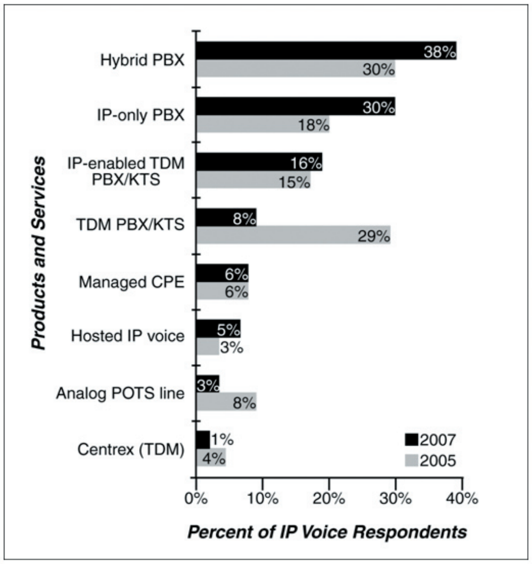

Figure 2. Voice Products and Services at Headquarters Sites. Copyright © 2005, Infonetics Research, Inc.

4. Benefits and Applications for the Enterprise

Imagine that there was only one communication interface from your office to the outside world. This would mean that you would have fewer cables to connect and fewer systems to manage. If you had any difficulties, there would only be one organization, not several, to call for support. The applications that were available on your computer's desktop could integrate both voice and data applications. These might include voicemail or a fax that could be redirected to your email account when you were away on a business trip, or a Web site that was integrated with a live customer response center, making online purchases easier. In addition, economic benefits would likely be realized, as the cost of a single high-speed connection would likely be less than the cost of multiple lower-speed circuits.

If we consider the benefits to a single network user, and extend those to the entire enterprise, we see that a converged network may yield advantages in monthly circuit charges, network management efficiencies, and reduction in associated staff expenses. The Infonetics Research study detailed the applications for IP voice that decision makers are considering, as shown in Figure 3:

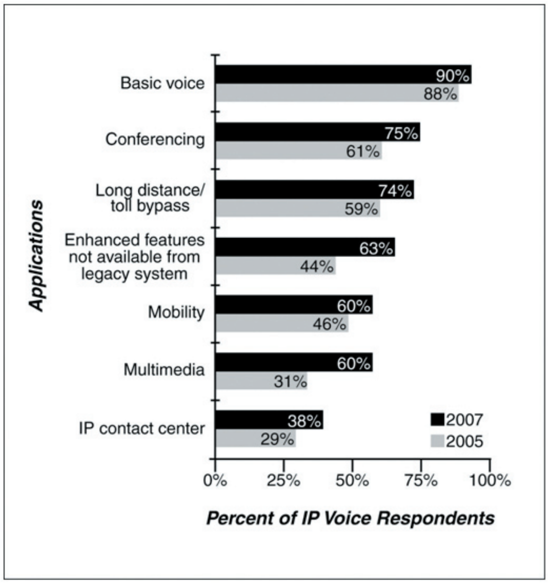

Figure 3. Applications for IP Voice
Copyright © 2005, Infonetics Research, Inc.

Several interesting conclusions can be derived from the above data. First, note that basic voice service is the most popular application, followed by conferencing and long distance/toll bypass. Since these three applications are quite fundamental to most business operations, one could conclude that many network managers are looking to converged networks as a replacement technology as existing PBX, Centrex or KTS systems are retired. But beyond these basic functions, network managers are also looking toward positioning their networks to support enhanced features such as mobility and multimedia applications in the years to come. Thus, the convergence solution can support both the present needs and the future applications, both of which are based upon a IP-centric infrastructure.

For further information on the Infonetics report or their market research services, visit their website at *www.infonetics.com*.

5. Challenges for Converged Networks

The Internet Protocol (IP) which provides the fundamental packet delivery mechanism for the Internet, is a connectionless protocol. It was originally designed to transport data, not voice, and therefore depended upon other higher layer protocols, such as the Transmission Control Protocol (TCP) to enhance its reliability. Reliability results for IP-based LANs and WANs vary widely, but it would be safe to say that these statistics are significantly below the 99.999% reliability found with the PSTN.

Here is the key challenge for converged networks:

How do you take a connection-oriented application (such as voice or video) and transmit it over a connectionless network (such as an IP network, or the Internet), and do so reliably?

This challenge should be considered from three perspectives: network infrastructure, continuing network management, and end-user quality of service.

The network infrastructure has likely been designed in support of voice communications or data communications, but not both. In other words, let us suppose that you are considering adding IP voice services or video conferencing to your existing internetwork. Does the network have the capacity for these additional, bandwidth-intensive, applications?

For the network manager, one of the key challenges is education. Most separate voice and data networks also had separate management staffs and systems. Thus, if you had responsibility for the voice wide area network, you were only peripherally involved in issues concerning the local data networks. As a result, you were not likely to develop expertise in LAN technologies. If you are now responsible for both voice and data elements, you must develop network management and analysis expertise in both these areas.

But perhaps the greatest challenge is keeping the end users satisfied with the service provided by your internetwork. Even though your end users may not be technologists who understand the inter-workings of computer communication systems, they know how to operate the telephone, and they know from their experience how reliable the telephone system is. They have been conditioned to expect a high quality of service. Anything less than meeting their existing expectations is likely to be unacceptable.

These three challenges – infrastructure, management and quality of service – will be examined in detail in the subsequent technical briefs. The following section outlines what is to come.

6. Looking Ahead

This is the first of six technical briefs on Converged Networks sponsored by Network General Corporation. Titles of other volumes in the series to be released include:

- *Protocols for the VoIP and Converged Network:* a look at the components of the converged network, the ITU-T and IETF multimedia protocol suites, and the protocols required by each component.
- *Implementing the VoIP Network:* will examine issues to consider before you jump in, including existing network utilization, planning for new applications, network design, and interoperability testing.
- *Managing Call Flows Using H.323:* the operation of the H.323 family of multimedia protocols, illustrated with case studies and output from the Sniffer protocol analyzer that show converged network operation from the H.323 protocol perspective.
- *Managing Call Flows Using SIP:* the operation of the Session Initiation Protocol (SIP) and the IETF multimedia protocol suite, again illustrated with case studies and output from the Sniffer protocol analyzer.
- *Supporting the VoIP and Converged Network:* this concluding paper will deal with on-going support requirements, including: multivendor interoperability, traffic prioritization, WAN bandwidth optimization and quality of service optimization.

7. Acronyms and Abbreviations

ATM	Asynchronous Transfer Mode
CCITT	Consultative Committee for International Telephony and Telegraphy
CON	Connection-oriented network service
CNLS	Connectionless network service
ETSI	European Telecommunications Standards Institute
IETF	Internet Engineering Task Force
IP	Internet Protocol
ITSP	Internet Telephony Service Provider
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union–Telecommunications Standard Sector
KTS	Key Telephone System
LAN	Local Area Network
PBX	Private Branch Exchange
PDA	Personal Digital Assistant
PSTN	Public Switched Telephone Network
RSVP	Resource Reservation Protocol
RTP	Real-time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TCP	Transmission Control Protocol
TIPHON	Telecommunications and Internet Protocol Harmonization Over Networks
VoIP	Voice over Internet Protocol
WAN	Wide Area Network

About the Author and Sponsor

Mark A. Miller, P.E., is President of DigiNet Corporation, a Denver-based consulting engineering firm providing services in internetwork design, strategic planning, network management and new product development. Mr. Miller is the author of twenty books on network analysis, design, and management. His latest book is titled the *Internet Technologies Handbook*, published by John Wiley & Sons, Inc. (Hoboken, New Jersey). He is a frequent presenter at industry events and has taught at the Networld+Interop, Comdex, and many other conferences. He holds B.S. and M.S. degrees in electrical engineering, and is a registered professional engineer in four states. For more information, DigiNet Corporation may be reached at 303.682.5244 or on the Internet at .

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