

Volume

2

PROTOCOLS FOR THE CONVERGED NETWORK

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Table of Contents

Executive Summary	i
1. Converging Legacy Networks	1
2. Enabling Systems for the Converged Network	2
3. ITU-T Multimedia Transport Protocols	4
4. IETF Multimedia Transport Protocols	6
5. Comparing Protocol Flows for PSTN and VoIP Calls	9
6. Looking Ahead	11
7. Acronyms and Abbreviations	13
8. About the Author and Sponsor	14

Executive Summary

This is the second of six technical briefing papers that examine the concepts, operation and analysis of *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, plus the widespread acceptance of the Internet Protocol (IP), provide the driving factors that enable these infrastructures.

Voice and data networks have very different objectives, and therefore differ in their infrastructure and operation. Legacy voice networks are likely to be based on Private Branch Exchange (PBX) architecture, with connections into the Public Switched Telephone Network (PSTN). Data networks are likely to be based on the Internet Protocol (IP), connecting terminals, hosts, and other processors. Because voice networks are connection-oriented and data networks are connectionless, gateways are required to process calls between these two dissimilar systems. Other devices known as gatekeepers are deployed to assist with network addressing and management functions.

Two protocol suites have been developed in support of converged networks. The International Telecommunications Union — Telecommunications Standard Sector (ITU-T) has developed the H.323 standard, an architecture that includes many other protocols that provide call management and information transfer functions. The Internet Engineering Task Force (IETF) has developed a multimedia transport suite that includes a number of protocols, including the Session Initiation Protocol (SIP). In addition, vendor products that are based on the ITU-T and IETF work, such as Cisco Systems, Inc.'s Skinny Client Control Protocol (SCCP) have been developed to optimize the cost and overhead of these protocol implementations.

A call through a converged network involves a number of subsystems including the end stations, gateways and gatekeepers, and therefore involves a number of protocol processes. A good understanding of these processes is essential for a successful converged network implementation.

1. Converging Legacy Networks

Volume One of this series of technical briefing papers explored the concepts and challenges of a *converged network*, one that has the capabilities to transmit voice, data, fax, video, or some combination of these different signals. That document also discussed the differences between connection-oriented networks such as the Public Switched Telephone Network (PSTN) and connectionless networks such as the Internet. For network convergence to occur, the legacy networks that implement these two infrastructures must be combined; an understanding of the differences between these two environments is critically important.

Legacy voice networks are likely to be based upon a Private Branch Exchange (PBX) network, with connections to both private and public (PSTN) wide area network (WAN) links (Figure 1A). Adjunct processors, such as voice mail systems, are likely to be incorporated into that network architecture. In most cases, vendor-proprietary signaling protocols are used for communication between the user stations and the PBX switch, and are also used between PBXs and adjunct processors such as voice mail systems. Industry standard protocols, such as the ITU-T Signaling System 7 (SS7) are used between central offices within the PSTN, thus assuring end-to-end interoperability. Telephone network-based addressing, defined in ITU-T recommendations such as E.164 *Numbering Plan for the ISDN Era*, is typically used.

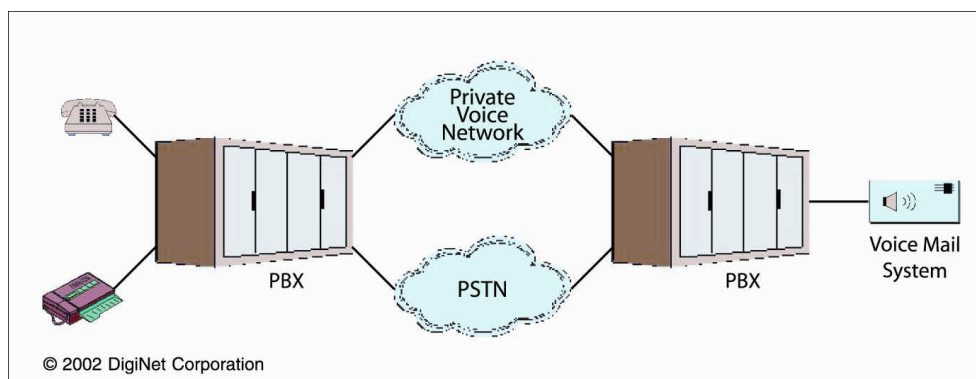


FIGURE 1A. **Legacy Voice Network**

Legacy data networks are likely to be based on the Internet Protocol (IP) and incorporate IP-based addressing schemes. Most host systems, from handheld Personal Digital Assistants (PDAs) to the largest mainframe processors have IP-based connectivity options. As a result, these legacy IP networks are likely to contain a wide

variety of processing and storage capabilities (Figure 1B). In many cases, the legacy data network is also a distributed system, incorporating WAN connectivity such as frame relay circuits between sites. Because of the packet-switched nature of this network, protocols such as the Open Shortest Path First (OSPF) or Border Gateway Protocol (BGP) are deployed to keep the routing information up to date. Note that signaling protocols are not typically required. With the connectionless environment, call setups are not required; furthermore, any WAN links are likely to be permanent connections.

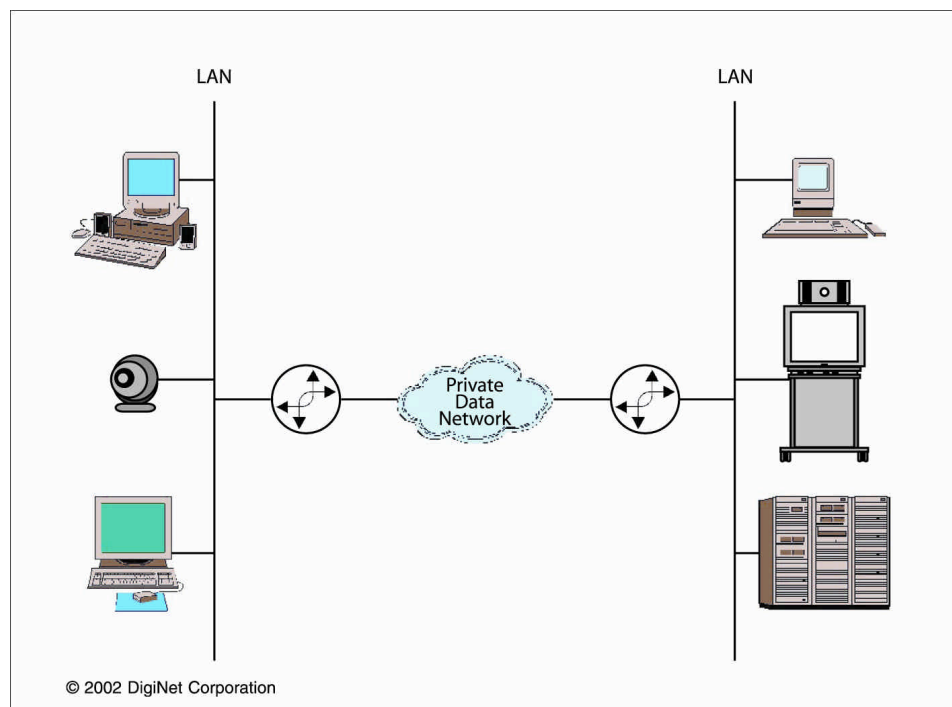


FIGURE 1B. **Legacy Data Network**

2. Enabling Systems for Converged Networks

The converged network incorporates elements from both the voice and data environments, but requires additional systems and protocol operations to enable the internetwork communication (Figure 2). For example, voice networks are connection oriented and require call signaling protocols to establish a call. In contrast, data networks are connectionless (the end stations simply drop packets into the network), but they require routing protocols to update the path information and increase the

likelihood of successful packet delivery. In addition, stations on the voice network are addressed using a telephone number entered from a touch-tone keypad. Stations on the data network are addressed according to the protocol deployed, such as a 32-bit IP address. These address and protocol conversions occur at several enabling systems:

- Gateways, which convert packet-based user information to/from circuit-based (or streaming) information, and also handle any signaling protocol issues, such as call setups and disconnects.
- Gatekeepers, which provide network management functions, such as address translations and bandwidth oversight for the network as a whole. Thus, a gatekeeper could prevent a video conference from consuming all available bandwidth, and leaving network users without basic connectivity.
- Domain Name System Servers, which work with the gatekeepers to provide address translation and management functions.
- Terminals, which provide the multimedia interface and signal processing functions such as converting the analog voice to a digital bit stream. Much of this signal processing is done in a codec chip, short for coder/decoder.

The protocols that operate in these enabling systems have been derived from two different organizations: the International Telecommunication Union — Telecommunication Standards Sector (ITU-T) and the Internet Engineering Task Force (IETF). We will look at these development efforts in the next two sections.

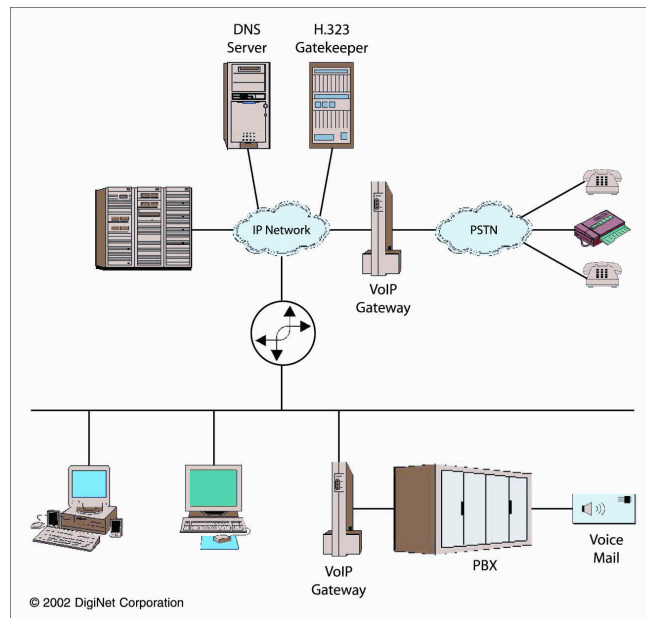


FIGURE 2. **Converged Voice/Data Network**

3. ITU-T Multimedia Transport Protocols

Many of the standards developed by the ITU-T have focused on transport technologies over WANs, including leased line, ISDN, frame relay, and ATM environments. Video conferencing became part of those efforts, with the early work focused on a standard for multimedia communications over ISDN environments, published by ITU-T Study Group 16 as Recommendation H.320. Based on this work, the research was expanded to add other packet-based network infrastructures, including those, such as Ethernet, which did not offer Quality of Service (QoS) guarantees like ISDN. These ITU-T multimedia architectures include:

- H.320: Narrowband visual telephone systems and terminal equipment, typically used with ISDN service.
- H.321: Adaptation of H.320 terminals to broadband ISDN (ATM) environments.
- H.322: Visual telephone systems and terminal equipment for guaranteed QoS LANs.
- H.323: Packet-based multimedia communication systems, typically used for non guaranteed QoS LANs.
- H.324: Terminal for low bit rate multimedia communication, typically used with PSTN and wireless applications.

Of the architectures noted above, H.323 is the most widely used for applications in current networking environments. H.323 is actually an umbrella architecture that incorporates work from other standards, such as ISDN signaling, plus work of other standards organizations, including the Internet Engineering Task Force (IETF). There are several key elements that are incorporated into the H.323 umbrella architecture:

- H.225.0 Signaling: Terminal-to-gatekeeper signaling functions, which establish, maintain and terminate connections between H.323 devices. H.225.0 signaling is based on the ITU-T Q.931 ISDN signaling standard.

- H.225 RAS: Registration, Admission and Status messages that are sent between the H.323 endpoint and the gatekeeper for network management.
- H.245: Terminal control functions, such as those required for the negotiation of channel usage, encoding algorithm selection, and so on between end stations.
- RTP: Transport of the digitally encoded voice or video information using the IETF Real-time Transport Protocol.
- Audio/video encoding: Transfers analog voice or video into a digitally encoded representation of the original signal, which is then suitable for transmission over a digital network, such as IP. Many encoding algorithms have been defined, such as the ITU-T G.723.1, G.728, G.729 standards, which will be discussed in future papers in this series.

An overview of the various ITU-T multimedia standards is illustrated in Figure 3. Note that a single network may contain elements based on both the ITU-T H.323 and IETF standards, since the underlying infrastructure for both is based on IP. Details on the ITU-T multimedia standards can be obtained at <http://www.itu.ch>.

H.323 is an extensive protocol designed to support a variety of multimedia applications. For some applications, such as simple voice communication within a PBX environment, the rigors of H.323 add overhead that are not required. Cisco Systems, Inc. has developed an alternative to this H.323 complexity with the Skinny Client Control Protocol (SCCP) or simply *Cisco Skinny*.

With the SCCP architecture, the vast majority of the H.323 processing power resides in an H.323 proxy known as the *Cisco Call Manager*. The end stations (telephones) run what is called the *Skinny Client*, which consumes less processing overhead. The Client communicates with the Call Manager using connection-oriented (TCP/IP-based) communication to establish a call with another H.323-compliant end station. Once the Call Manager has established the call, the two H.323 end stations use connectionless (UDP/IP-based) communication for audio transmissions. Costs and overhead are thus reduced by confining the complexities of H.323 call setup to the Call Manager, and using the Skinny protocol for the actual audio communication into and out of the end stations.

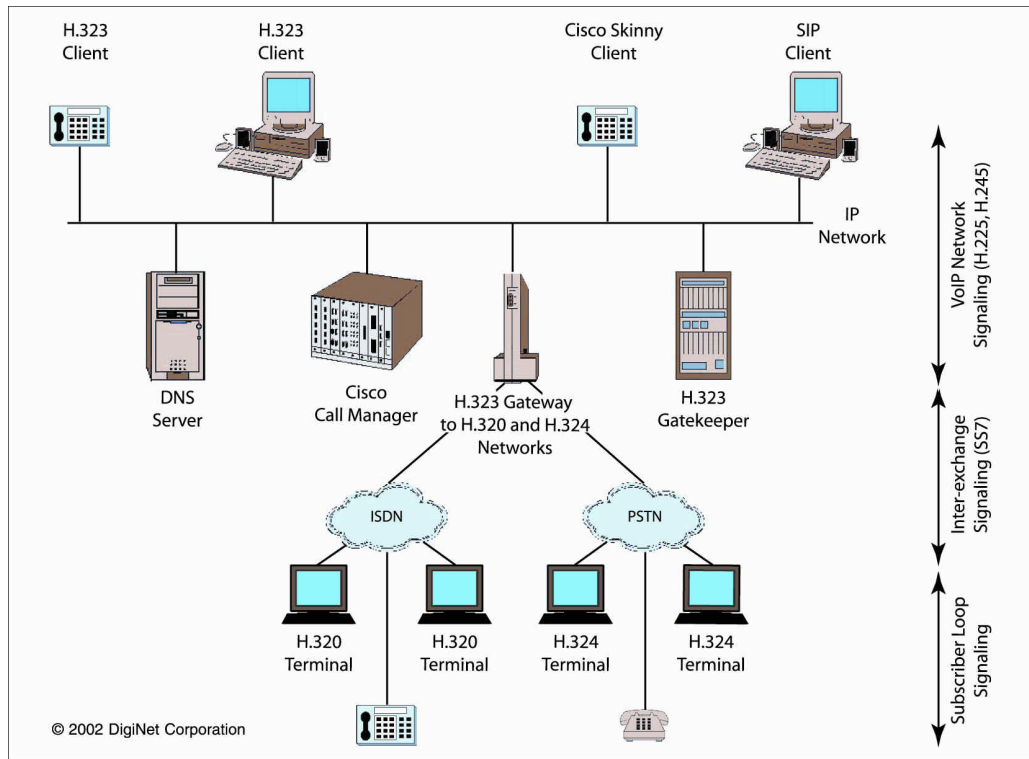


FIGURE 3. ITU-T and IETF Multimedia Protocol Implementations

4. IETF Multimedia Transport Protocols

The Internet Protocol (IP) was developed in the late 1960s, and designed to support data — not voice or video — communication. As new Internet-related technologies such as the World Wide Web emerged, it became evident that extending the use and capabilities of IP beyond the typical email, file transfer and remote terminal applications would be beneficial. As a result, the Transport Area of the Internet Engineering Task Force (IETF) has chartered a number of working groups that are investigating specific areas of multimedia technology. Details on the work of these individual groups can be found at <http://www.ietf.org/html.charters/>.

Figure 4 illustrates the individual protocols that comprise much of the results from the IETF working groups mentioned above. Note from the figure the protocol dependencies illustrated with the arrows. For example, audio/video applications are encoded with one of a number of coding standards, which then flows through the Real

Time Protocol (RTP), the User Datagram Protocol (UDP), the Internet Protocol (IP), and finally to the LAN, MAN, or WAN network interface.

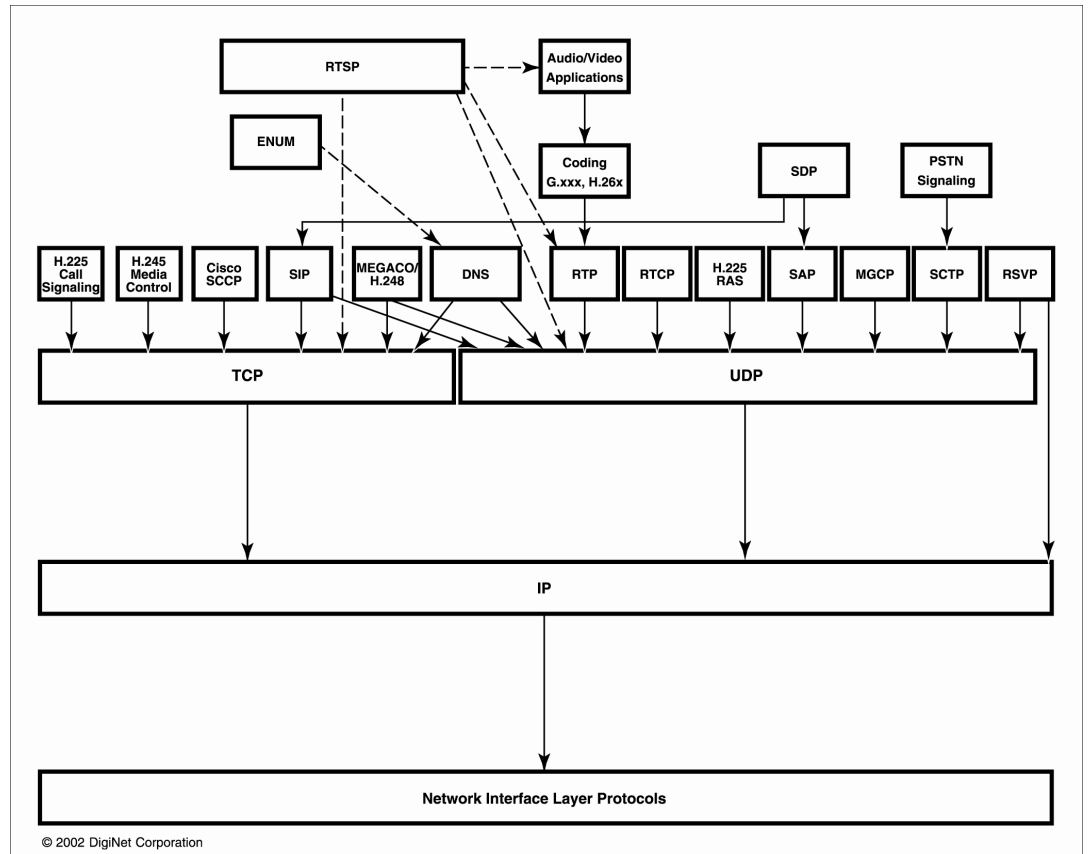


FIGURE 4. ITU-T and IETF Multimedia Transport Protocol Relationships

The protocols that comprise the IETF multimedia architecture include:

- IP: Internet Protocol, which is responsible for addressing and message fragmentation/reassembly within the packet-switched environment.
- UDP: User Datagram Protocol, which provides for connectionless end-to-end transport reliability.
- TCP: Transmission Control Protocol, which provides for connection-oriented end-to-end reliability, but requires more overhead than UDP.

- SIP: Session Initiation Protocol, which initiates end-user connections. SIP is similar in function to the ITU-T H.323 protocol, but designed with less overhead than the earlier versions of H.323.
- MEGACO/H.248: a joint effort from the IETF and ITU-T to develop a gateway control protocol.
- DNS: Domain Name Systems, which provides a method of associating well-known host names with IP addresses.
- RTP: Real-time Transport Protocol, which adds timestamps, sequencing and other parameters for transport of time-sensitive information.
- RTCP: RTP Control Protocol, which periodically transmits control packets to provide feedback on the status of the RTP transmission.
- SAP: Session Announcement Protocol, which periodically announces parameters of a conference session.
- MGCP: Media Gateway Control Protocol, which defines messages and procedures to control gateways connecting dissimilar transport media.
- RSVP: Resource Reservation Protocol, which reserves bandwidth along a path for improved application support.
- SCTP: Stream Control Transmission Protocol, which is used for connection-oriented transport of PSTN signaling over IP networks.
- SDP: Session Description Protocol, which defines a standard method of describing the characteristics of a multimedia session.
- ENUM: Electronic Numbers, which define mapping algorithms between telephone numbers and IP addresses.
- RTSP: Real Time Streaming Protocol, an application protocol that controls the delivery of streaming information, such as voice and video.
- Audio/Video Encoding: standard algorithms that digitize and compress/decompress an audio or video stream for packet transmission.

Many of these protocols are used to complete a call over a converged network, as described in the next section.

5. Comparing Protocol Flows for PSTN and VoIP Calls

As the previous sections have described, a number of systems and protocol processes are involved in completing telephone calls. Let's compare the processes used in both PSTN (circuit-switched) and VoIP (packet switched) voice call scenarios.

A circuit-switched telephone call through the PSTN involves the five components shown in Figure 5A. These include: end user equipment, such as telephones and fax machines; connections to the local exchange central office, typically provided on copper pairs; local switching offices; a signaling network and a transport network. A typical telephone call through the PSTN would involve five major steps:

1. An analog telephone goes off-hook. This signal is recognized at the local central office, which returns dial tone and accepts the destination telephone number, typically transmitted using Dual Tone Multi-frequency (DTMF) tones.
2. The local central office generates a call setup packet that passes through the signaling network, identifying a path to the desired destination. Once that path has been identified, transport network resources are reserved in support of the connection.
3. The destination central office signals the destination telephone of an incoming call using ringing tone.
4. When the destination party answers, the signaling network starts billing, and the voice network connection becomes active.
5. The two parties carry on a conversation, and when completed, hang up their telephones. The on-hook signals are recognized at the local and destination central offices. Billing functions are then completed, and the reservations of the transport network facilities are released so that those resources can be devoted to another call.

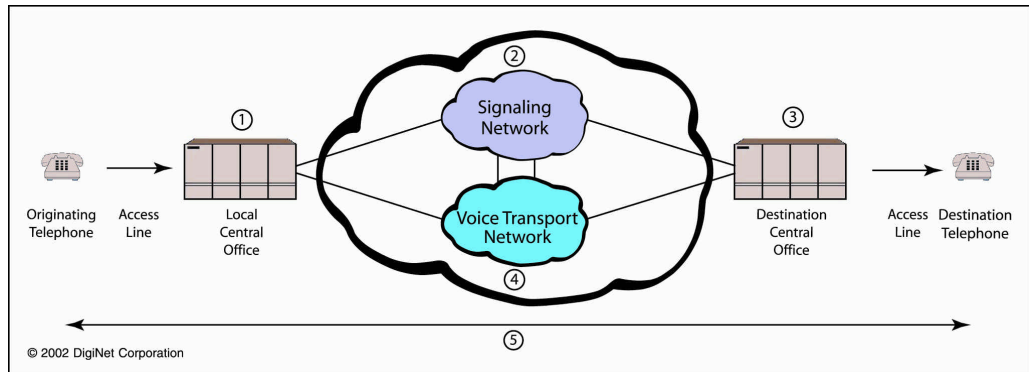


FIGURE 5A. **PSTN Call Processing and Protocol Flows**

Placing a call via a packet network is similar to the PSTN case. Figure 5B illustrates an example of a call using H.323 protocols, and the involvement of other processors such as gatekeepers and gateways (note that processes for other protocol architectures, such as SIP, MGCP or Cisco SCCP would be different). Five major steps would also be involved in this scenario:

1. An analog telephone goes off hook and places a call to a remote telephone connected to an IP network. The signaling network within the PSTN receives the calling user's destination number, which is then passed to the VoIP gateway.
2. Call signaling information is passed from the gateway to the gatekeeper requesting admission to the network. This signaling information is sent using TCP/IP for greater reliability.
3. Since the destination telephone resides on another network, the respective gatekeepers communicate signaling information requesting call completion.
4. The destination client and remote gatekeeper exchange signaling information.
5. Once all call signaling is completed, the two end stations exchange media information (voice, video, etc. samples) using RTP/UDP/IP for greater efficiency. When the information transfer is completed, additional signaling messages are used to disconnect the call.

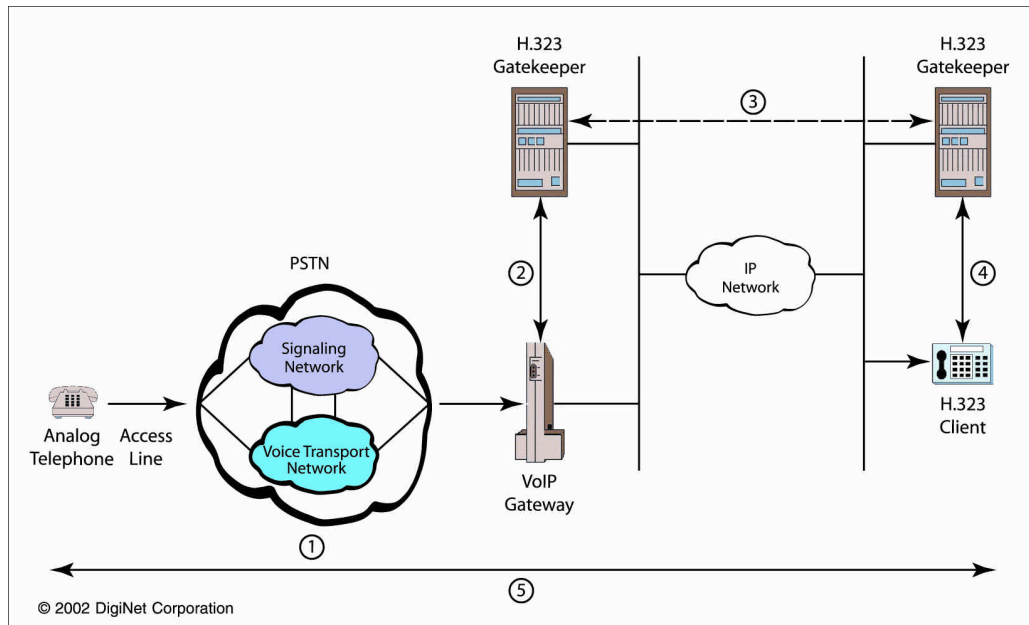


FIGURE 5B. **VoIP Call Processing and Protocol Flows**

Comparing the PSTN and VoIP call scenarios, note that two network operations are involved in each case: signaling, for call establishment, management and termination, plus information transport from source to destination. The analysis of these two operations will be central themes of the future papers in this briefing series.

6. Looking Ahead

This is the second of six technical briefs on Converged Networks sponsored by Sniffer Technologies. Titles of current and future volumes in the series include:

- 1. Introduction to Converged Networking:** A description of concepts and challenges of converged networks, including business, technical and operational issues.
- 3. Implementing the Voice over IP Network:** Issues to consider before you jump in, including existing network utilization, planning for new applications, network design, and interoperability testing.
- 4. 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.

5. 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.

6. Supporting the Converged Network: This concluding paper will deal with ongoing support requirements, including: multivendor interoperability, traffic prioritization, WAN bandwidth optimization and quality of service optimization

7. Acronyms and Abbreviations

CCITT	Consultative Committee for International Telephony and Telegraphy
CON	Connection-oriented network service
CNLS	Connectionless network service
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System
ENUM	Electronic Numbers
ETSI	European Telecommunications Standards Institute
IETF	Internet Engineering Task Force
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ITSP	Internet Telephony Service Provider
ITU-T	International Telecommunication Union — Telecommunications Standard Sector
LAN	Local Area Network
MGCP	Media Gateway Control Protocol
PBX	Private Branch Exchange
PDA	Personal Digital Assistant
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RAS	Registration, Admission, and Status
RSVP	Resource Reservation Protocol
RTCP	Real-time Control Protocol
RTP	Real-time Transport Protocol
RTSP	Real-time Streaming Protocol
SAP	Session Announcement Protocol
SCCP	Skinny Client Control Protocol
SCTP	Stream Control Transmission 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

8. 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 nineteen books on network analysis, design, and management. His latest book is titled *Voice over IP Technologies, Strategies for the Converged Enterprise*, published in 2002 by M&T Books, Inc., a division of John Wiley (Indianapolis, Indiana). He is a frequent presenter at industry events and has taught at the ComNet, CT Expo, Internet Telecom Expo, Network+Interop, Comdex, and 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 <http://www.dignet.com>.

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