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Voice over IP (VoIP)

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About the Editor...

Jerry Ryan is the Vice President of Editorial Development for the Technology Guides on Communications and Networking. Mr. Ryan is also a principal at ATG. Mr. Ryan has developed and taught many courses in network analysis and design for carriers, government agencies and private industry. He has provided consulting support in the area of WAN and LAN network design, negotiation with carriers for contract pricing and services, technology acquisition, customized software development for network administration, billing and auditing of telecommunication expenses, project management, and RFP generation. He was the president and founder of Connections Telecommunications, Inc., a Massachusetts based company specializing in consulting, education, and software tools which address network design and billing issues. Mr. Ryan is a member of the Networld+Interop Program Committee. He holds a B.S. degree in electrical engineering.

Communicating via packet data networks such as IP, ATM, and Frame Relay has become a preferred strategy for both corporate and public network planners. Experts are predicting that data traffic will soon exceed telephone traffic, if it hasn't already. At the same time, more and more companies are seeing the value of transporting voice over IP networks to reduce telephone and facsimile costs and to set the stage for advanced multimedia applications. Providing high quality telephony over IP networks is one of the key steps in the convergence of voice, fax, video, and data communications services. Voice over IP has now been proven feasible; the race is on to adopt standards, design terminals and gateways, and begin the roll-out of services on a global scale. Needless to say, the technical difficulties of transporting voice and the complexities of building commercial products are challenges many companies are facing today. Adding voice to packet networks requires an understanding of how to deal with system level challenges such as interoperability, packet loss, delay, density, scalability, and reliability. The Internet and the corporate Intranet must soon be voice-enabled if they are to make the vision of "one-stop networking" a reality. This Technology Guide examines recent advances in the infrastructures, equipment, and embedded systems that are needed to successfully enable VoIP and discusses the major issues currently facing product developers. The types of applications that will benefit the most from voice/data convergence are also reviewed.

Introduction

The public telephone network and the equipment that makes it possible are taken for granted in most parts of the world. Availability of a telephone and access to a low-cost, high-quality worldwide network is considered to be essential in modern society (telephones are even expected to work when the power is off). Anything that would jeopardize this is usually treated with suspicion. There is, however, a paradigm shift beginning to occur since more and more communications is in digital form and transported via packet networks such as IP, ATM cells, and Frame Relay frames. Since data traffic is growing much faster than telephone traffic, there has been considerable interest in transporting voice over data networks (as opposed to the more traditional data over voice networks).

Support for voice communications using the Internet Protocol (IP), which is usually just called "Voice over IP" or VoIP, has become especially attractive given the low-cost, flat-rate pricing of the public Internet. In fact, toll quality telephony over IP has now become one of the key steps leading to the convergence of the voice, video, and data communications industries. The feasibility of carrying voice and call signaling messages over the Internet has already been demonstrated but delivering high-quality commercial products, establishing public services, and convincing users to buy into the vision are just beginning.

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. Equipment producers see VoIP as a new opportunity to innovate and compete. The challenge for them is turning this vision into reality by quickly developing new VoIP-enabled equipment. For Internet service providers, the possibility of introducing usage-based pricing and increasing their traffic volumes is very attractive. Users are seeking new types of integrated voice/data applications as well as cost benefits.

Successfully delivering voice over packet networks presents a tremendous opportunity; however, implementing the products is not as straightforward a task as it may first appear. This Technology Guide examines the technologies, infrastructures, software, and systems that will be necessary to realize VoIP on a large scale. Product development challenges such as ensuring interoperability, scalability, and cost/effectiveness will be discussed. The types of applications that will both drive the market and benefit the most from the convergence of voice and data networks will be identified.

Applications and Benefits of VoIP

Voice communications will certainly remain a basic form of interaction for all of us. The PSTN simply cannot be replaced, or even dramatically changed, in the short term (this may not apply to private voice networks, however). The immediate goal for VoIP service providers is to reproduce existing telephone capabilities at a significantly lower "total cost of operation" and to offer a technically competitive alternative to the PSTN. It is the combination of VoIP with point-of-service applications that shows great promise for the longer term.

The first measure of success for VoIP will be cost savings for long distance calls as long as there are no additional constraints imposed on the end user. For example, callers should not be required to use a microphone on a PC. VoIP provides a competitive threat to the providers of traditional telephone services that, at the very least, will stimulate improvements in cost and function throughout the industry.

Figure 1 illustrates one scenario for how telephony and facsimile can be implemented using an IP network. This design would also apply if other types of packet networks (such as frame relay) were being used.

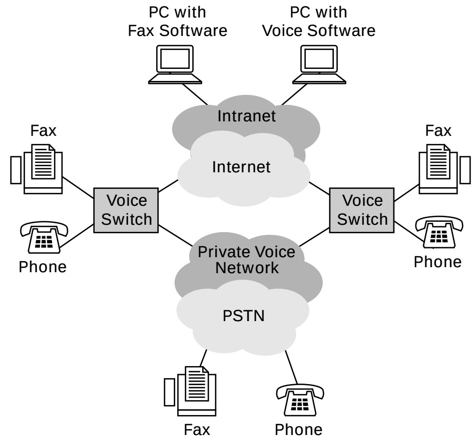

Figure 1: VoIP Infrastructure

VoIP could be applied to almost any voice communications requirement, ranging from a simple inter-office intercom to complex multi-point teleconferencing/shared screen environments. The quality of voice reproduction to be provided could also be tailored according to the application. Customer calls may need to be of higher quality than internal corporate calls, for example. Hence, VoIP equipment must have the flexibility to cater to a wide range of configurations and environments and the ability to blend traditional telephony with VoIP.

Some examples of VoIP applications that are likely to be useful would be:

a) *PSTN gateways:* Interconnection of the Internet to the PSTN can be accomplished using a gateway, either integrated into a PBX (the iPBX) or provided as a separate device. A PC-based telephone, for example, would have access to the public network by calling a gateway at a point close to the destination (thereby minimizing long distance charges).

b) *Internet-aware telephones:* Ordinary telephones (wired or wireless) can be enhanced to serve as an Internet access device as well as providing normal telephony. Directory services, for example, could be accessed over the Internet by submitting a name and receiving a voice (or text) reply.

c) *Inter-office trunking over the corporate intranet:* Replacement of tie trunks between company-owned PBXs using an Intranet link would provide economies of scale and help to consolidate network facilities.

d) *Remote access from a branch (or home) office:* A small office (or a home office) could gain access to corporate voice, data, and facsimile services using the company's Intranet (emulating a remote extension for a PBX, for example). This may be useful for home-based agents working in a call center, for example.

e) *Voice calls from a mobile PC via the Internet:* Calls to the office can be achieved using a multimedia PC that is connected via the Internet. One example would be using the Internet to call from a hotel instead of using expensive hotel telephones. This could be ideal for submitting or retrieving voice messages.

f) *Internet call center access:* Access to call center facilities via the Internet is emerging as a valuable adjunct to electronic commerce applications. Internet call center access would enable a customer who has questions about a product being offered over the Internet to access customer service agents online. Another VoIP application for call centers is the interconnection of multiple call centers.

One of the immediate applications for IP telephony is real-time facsimile transmission. Facsimile services normally use dial-up PSTN connections, at speeds up to 14.4 Kbps, between pairs of compatible fax machines. Transmission quality is affected by network delays, machine compatibility, and analog signal quality. To operate over packet networks, a fax interface unit must convert the data to packet form, handle the conversion of signaling and control protocols (the T.30 and T.4 standards), and ensure complete delivery of the scan data in the correct order. For this application, packet loss and end-to-end delay are more critical than in voice applications.

Most VoIP applications that have been defined are considered to be real-time activities. Store-and-forward voice services will also be implemented using VoIP. For example, voice messages could be prepared locally using a telephone and delivered to an integrated voice/data mailbox using Internet or intranet services. Voice annotated documents, multimedia files, etc. will also become standard within office suites in the near future. The real-time and store-and-forward modes of operation will need to be compatible and interoperable.

Widespread deployment of a new technology seldom occurs without a clear and sustainable justification, and this is also the case with VoIP. Demonstrable benefits to end users are also needed if VoIP products (and services) are to be a long-term success. Generally, the benefits of technology can be divided into the following four categories:

- *Cost Reduction*. Although reducing long distance telephone costs is always a popular topic and would provide a good reason for introducing VoIP, the actual savings over the long term are still a subject of debate in the industry. Flat rate pricing is available with the Internet and can result in considerable savings for both voice and facsimile (at least currently). It has been estimated that up to 70% of all calls to Asia are to send faxes, most of which could be replaced by FoIP. These lower prices, however, are based on avoiding telephony access charges and settlement fees rather than being a fundamental reduction in resource costs. The sharing of equipment and operations costs across both data and voice users can also improve network efficiency since excess bandwidth on one network can be used by the other, thereby creating economies of scale for voice (especially given the rapid growth in data traffic).
- *Simplification*. An integrated infrastructure that supports all forms of communication allows more standardization and reduces the total equipment complement. This combined infrastructure can support dynamic bandwidth optimization and a fault tolerant design. The differences between the traffic patterns of voice and data offer further opportunities for significant efficiency improvements.
- *Consolidation*. Since people are among the most significant cost elements in a network, any opportunity to combine operations, to eliminate points of failure, and to consolidate accounting systems would be beneficial. In the enterprise, SNMP-based management can be provided for both voice and data services using VoIP. Universal use of the IP protocols for all applications holds out the promise of both reduced complexity and more flexibility. Related facilities such as directory services and security services may be more easily shared.
- *Advanced Applications*. Even though basic telephony and facsimile are the initial applications for VoIP, the longer term benefits are expected to be derived from multimedia and multiservice applications. For example, Internet commerce solutions can combine WWW access to information with a voice call button that allows immediate access to a call center agent from the PC. Needless to say, voice is an integral part of conferencing systems that may also include shared screens, whiteboarding, etc. Combining voice and data features into new applications will provide the greatest returns over the longer term.

Although the use of voice over packet networks is relatively limited at present, there is considerable user interest and trials are beginning. End user demand is expected to grow rapidly over the next five years. Frost & Sullivan and other research firms have estimated that the compound annual growth rate for IP-enabled telephone equipment will be 132% over the period from 1997 to 2002 (from $47.3M in 1997 to $3.16B by 2002). It is expected that VoIP will be deployed by 70% of the Fortune 1000 companies by the year 2000. Industry analysts have also estimated that the annual revenues for the IP fax gateway market will increase from less than $20M in 1996 to over $100M by the year 2000. It is clear that a market has already been established and there exists a window of opportunity for developers to bring their products to market.

VoIP Product Development Challenges

The goal for developers is relatively simple: add telephone calling capabilities (both voice transfer and signaling) to IP-based networks and interconnect these to the public telephone network and to private voice networks in such as way as to maintain current voice quality standards and preserve the features everyone expects from the telephone.

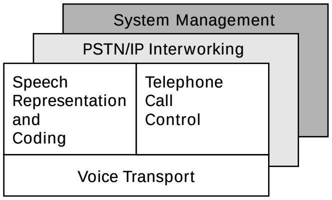

Figure 2: VoIP Architecture

Figure 2 illustrates an overall architecture for VoIP and suggests that the challenges for the product developer arise in five specific areas:

1. Voice quality should be comparable to what is available using the PSTN, even over networks having variable levels of QoS.
2. The underlying IP network must meet strict performance criteria including minimizing call refusals, network latency, packet loss, and disconnects. This is required even during congestion conditions or when multiple users must share network resources.
3. Call control (signaling) must make the telephone calling process transparent so that the callers need not know what technology is actually implementing the service.
4. PSTN/VoIP service interworking (and equipment interoperability) involves gateways between the voice and data network environments.
5. System management, security, addressing (directories, dial plans) and accounting must be provided, preferably consolidated with the PSTN operation support systems (OSSs).

The race to create VoIP products that suit a wide range of user configurations has now begun. Standards must be adopted and implemented, gateways providing high-volume IP and PSTN interfaces must be deployed, existing networks need to be QoS-enabled and global public services need to be established. Adoption of VoIP must also remain economically viable even if PSTN prices decrease. Needless to say, developers often underestimate both the difficulties of adding voice to packet networks and the complexities involved in building products suitable for public networks.

Speech Quality and Characteristics

Providing a level of quality that at least equals the PSTN (this is usually referred to as “toll quality voice”) is viewed as a basic requirement, although some experts argue that a cost versus function versus quality trade-off should be applied. Although QoS usually refers to the fidelity of the transmitted voice and facsimile documents, it can also be applied to network availability (i.e., call capacity, or level of call blocking), telephone feature availability (conferencing, calling number display, etc.), and scalability (any-to-any, universal, expandable).

The quality of sound reproduction over a telephone network is fundamentally subjective, although standardized measures have been developed by the ITU. It has been found that there are three factors that can profoundly impact the quality of the service (see Figure 3):

Delay: Two problems that result from high end-to-end delay in a voice network are echo and talker overlap. Echo becomes a problem when the round-trip delay is more than 50 milliseconds. Since echo is perceived as a significant quality problem, VoIP systems must address the need for echo control and implement some means of echo cancellation. Talker overlap (the problem of one caller stepping on the other talker's speech) becomes significant if the one-way delay becomes greater than 250 milliseconds. The end-to-end delay budget is therefore the major constraint and driving requirement for reducing delay through a packet network.

Jitter (Delay Variability): Jitter is the variation in inter-packet arrival time as introduced by the variable transmission delay over the network. Removing jitter requires collecting packets and holding them long enough to allow the slowest packets to arrive in time to be played in the correct sequence, which causes additional delay. The jitter buffers add delay, which is used to remove the packet delay variation that each packet is subjected to as it transits the packet network.

Packet Loss: IP networks cannot provide a guarantee that packets will be delivered at all, much less in order. Packets will be dropped under peak loads and during periods of congestion (caused, for example, by link failures or inadequate capacity). Due to the time sensitivity of voice transmissions, however, the normal TCP-based re-transmission schemes are not suitable. Approaches used to compensate for packet loss include interpolation of speech by re-playing the last packet, and sending of redundant information. Packet losses greater than 10% are generally not tolerable.

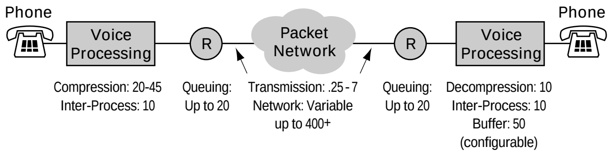

Figure 3: Delay and Jitter

Maintenance of acceptable voice quality levels despite inevitable variations in network performance (such as congestion or link failures) is achieved using such techniques as compression, silence suppression, and QoS-enabled transport networks. Several developments in the 1990s, most notably advances in digital signal processor technology, high-powered network switches, and QoS-based protocols, have combined to enable and encourage the implementation of voice over data networks. Low-cost, high-performance DSPs can process the compression and echo cancellation algorithms efficiently.

Software pre-processing of voice conversations can also be used to further optimize voice quality. One technique, called silence suppression, detects whenever there is a gap in the speech and suppresses the transfer of things like pauses, breaths, and other periods of silence. This can amount to 50-60% of the time of a call, resulting in considerable bandwidth conservation. Since the lack of packets is interpreted as complete silence at the output, another function is needed at the receiving end to add "comfort noise" to the output.

Another software function that improves speech quality is echo cancellation. As was noted earlier, echo becomes a problem whenever the end-to-end delay for a call is greater than 50 milliseconds. Sources of delay in a packet voice call include the collection of voice samples (called accumulation delay), encoding/decoding and packetizing time, jitter buffer delays, and network transit delay. The ITU recommendation G.168 defines the performance requirements that are currently required for echo cancellers.

Engineering a VoIP network (and the equipment used to build it) involves trade-offs among the quality of the delivered speech, the reliability of the system, and the delays inherent in the system. Minimizing the end-to-end delay budget is one of the key challenges in VoIP systems. Ensuring reliability in a "best effort" environment is another. Equipment producers that offer the flexibility to configure their systems to fit the environment and thereby optimize the quality of the voice produced will have a competitive advantage.

IP Network Support for Voice

A key requirement for successful VoIP deployment is the availability of an underlying IP-based network that is capable of supporting real-time telephone and facsimile. As was noted above, voice quality is affected by delay, jitter, and unreliable packet delivery - all of which are typical characteristics of the basic IP network service.

Most of today's data network equipment - routers, LAN switches, ATM switches, network interface cards, PBXs, etc. - will need to be able to support voice traffic. Furthermore, VoIP-specific equipment will either have to be integrated into these devices or work compatibly with them. VoIP equipment must also accommodate environments ranging from private, well-planned corporate Intranets to the less predictable Internet. Three different techniques are used (separately or in combination) to improve network quality of service.

- Providing a *controlled networking environment* in which capacity can be pre-planned and adequate performance can be assumed (at least most of the time). This would generally be the case with a private IP network (an Intranet) that is owned and operated by a single organization.
- Using *management tools* to configure the network nodes, monitor performance, and manage capacity and flow on a dynamic basis. Most internetworking devices (routers, switches, etc.) include a variety of mechanisms that can be useful in supporting voice. For example, traffic can be prioritized by location, by protocol, or by application type, thereby allowing real-time traffic to be given precedence over non-critical traffic. Queuing mechanisms can also be manipulated to minimize delays for real-time data flows. More recent developments, such as tag switching and flow switching, can also improve overall performance and reduce delays.
- Adding *control protocols and mechanisms* that help avoid or alleviate the problems inherent in IP networks. Protocols such as RTP (real-time protocol) and RSVP (Resources Reservation Protocol) are also being used to provide greater assurances of controlled QoS within the network. Other mechanisms such as admission controls and traffic shaping may also be used to avoid overloading a network (this would be comparable to getting a network busy signal on the telephone at peak periods such as Christmas).

VoIP equipment, which can be categorized into client, access/gateway, and carrier class/infrastructure segments, should be configurable to capitalize on these different techniques but must also be sufficiently flexible to add new techniques as they become available. Producers that make use of embedded software should focus on how to best utilize the functions instead of focusing on the problems associated with implementing and testing the objects themselves.

Real-time voice traffic can be carried over IP networks in three different ways:

- *Voice trunks* can replace the analog or digital circuits that are serving as voice trunks (such as private links between company-owned PBXs) or PSTN-access trunks (links between a PBX and the carrier). Voice packets are transferred between pre-defined IP addresses, thereby eliminating the need for phone number to IP address conversions. Fallback to the PSTN (or other private voice circuits) is always an option in this scenario.
- *PC-to-PC voice* can be provided for multimedia PCs (i.e., PCs with a microphone and sound system) operating over an IP-based network without connecting to the PSTN. PC applications and IP-enabled telephones can communicate using point-to-point or multipoint sessions (a form of Internet ham radio). This type of system may emulate a CB radio or an Internet chat group and could be combined with shared data systems such as whiteboards (i.e., multimedia solutions).
- *Telephony (any phone-to- any other phone)* communications (as is illustrated in Figure 1) appears like a normal telephone to the caller but may actually consist of various forms of voice over packet network, all interconnected to the PSTN. Gateway functionality is required when interconnecting to the PSTN or when interfacing the standard telephones to a data network. In the future, IP-enabled telephones will connect directly. For true universality, standards for VoIP (and voice over frame relay or ATM) must be adopted and applied.

Future VoIP networks will include IP-based PBXs (iPBXs), which will emulate the functions of a traditional PBX. These will allow both standard telephones and multimedia PCs to connect to either the PSTN or the Internet, providing a seamless migration path to VoIP. An iPBX can also combine the features of today's switches and routers and could become the gateway into a variety of value-added services such as directories, message stores, firewalls and other network-based servers. Such a VoIP system would also combine real-time and non real-time communications. Voice and facsimile messaging, for example, use functions that are very similar to a telephone call but do not need the same levels of QoS in the underlying network.

Figure 4 illustrates the IP network protocols that are currently being used to implement VoIP.

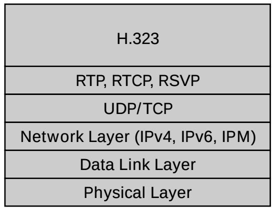

Figure 4: VoIP Protocol Structure

VoIP Network Protocols

Other Standards	Description
RTP (Real-time Transport Protocol)	IETF RFC1889, a real-time end-to-end protocol utilizing existing transport layers for data that has real-time properties
RTCP (RTP Control Protocol)	IETF RFC1889, a protocol to monitor the QoS and to convey information about the participants in an ongoing session; provides feedback on total performance and quality so that modifications can be made
RSVP (Resource Reservation Protocol)	IETF RFC2205-2209, a general purpose signaling protocol allowing network resources to be reserved for a connectionless data stream, based on receiver-controlled requests
IA 1.0	VoIP Forum Implementation Agreement 1.0 selecting protocol options for interoperable VoIP
TCP, UDP	Internet standard Transport Layer protocols
IPv4, IPv6, IP multicast and various routing protocols	Internet standard Network Layer protocols (currently IPv4 is in widespread use) both for data transfer and routing
Various subnetworks including ATM and Frame Relay	A variety of subnetworks can be used to carry IP datagrams including LANs and WANs using a variety of transmission techniques
SNMP (Simple Network Management Protocol)	Internet standard for communications between a manager and a managed object
LDAP (Lightweight Directory Access Protocol	Internet standard for accessing Internet directory services
Other Internet application protocols	Several other application protocols are used in conjunction with network nodes including FTP, Telnet, http/WWW, etc.

The most important consideration at the network level is to minimize unnecessary data transfer delays. Providing sufficient node and link capacity and using congestion avoidance mechanisms (such as prioritization, congestion control, and access controls) can help to reduce overall delay. The ability to manage network loading (as is feasible with Intranets but not available in the Internet) and optimize route choices will reduce the effects of jitter. Equipment producers should, wherever possible, avoid proprietary mechanisms (or combinations of mechanisms) that simply re-create solutions that are available “off-the-shelf.”

Software Support for VoIP

Voice and telephone calling can be viewed as one of many applications for an IP network, with software being used to support the application and interface to the network. The emergence of VoIP is a direct result of the advances that have been made in hardware and software technologies in the early 1990s.

The software functionality required for voice-to-packet conversion in a VoIP terminal or gateway are:

- The *Voice Processing* module, which prepares voice samples for transmission over the packet network (see below). This software is typically run on a DSP.
- The *Call Processing (Signaling)* module, which serves as a signaling gateway allowing calls to be established across the packet network. This software supports E&M (wink, delay and immediate), loop, or ground start FXS and FXO.
- The *Packet Processing* module, which processes voice and signaling packets, adding the appropriate transport headers prior to submitting

the packets to the IP network (or other packet networks). Signaling information is converted from telephony protocols to the packet signaling protocol.

- The *Network Management* module, which provides management agent functionality, allowing remote fault, accounting, and configuration management to be performed from standard management systems (see the next section). The Network Management module could include ancillary services such as support for security features, access to dialing directories, and remote access support.

The Voice Processing module must include software to perform the following functions:

a) The *PCM Interface*, which receives samples from the telephony (PCM) interface and forwards them to the appropriate VoIP software module for processing (and vice versa). The PCM interface performs continuous phase re-sampling of output samples to the analog interface.

b) The *Echo Cancellation Unit*, which performs echo cancellation on sampled, full-duplex voice port signals in accordance with the ITU G.165 or G.168 standard. Since round-trip delay for VoIP is always greater than 50 milliseconds (the point at which echo becomes intolerable), echo cancellation is a requirement. Operational parameters may be programmable.

c) The *Voice Activity/Idle Noise Detector*, which suppresses packet transmission when voice signals are not present (and hence saves additional bandwidth). If no activity is detected for a period of time, the voice encoder output will not be transported across the network. Idle noise levels are also measured and reported to the destination so that "comfort noise" can be inserted into the call (so that the listener does not get dead air on their telephone).

d) The *Tone Detector*, which detects the reception of DTMF tones and discriminates between voice and facsimile signals. These can be used to invoke the appropriate voice processing functions (i.e., the decoding and packetizing of facsimile information or the compression of voice).

e) The *Tone Generator*, which generates DTMF tones and call progress tones under command of the operating system.

f) The *Facsimile Processing* module, which provides a facsimile relay function by demodulating the PCM data, extracting the relevant information, and packing the scan data into packets.

g) The *Packet Voice Protocol* module, which encapsulates the compressed voice and fax data for transmission over the data network. Each packet includes a sequence number that allows the received packets to be delivered in the correct order. This also allows silence intervals to be reproduced properly and lost packets to be detected.

h) The *Voice Playout* module at the destination, which buffers the packets that are received and forwards them to the voice codec for playout. This module provides an adaptive jitter buffer and a measurement mechanism that allows buffer sizes to be adapted to the performance of the network.

The Call Processing (signaling) subsystem detects the presence of a new call and collects addressing information. Various telephony signaling standards must be supported. A number of functions must be performed if full telephone calling is to be supported.

- The interface to the telephone network must be monitored to collect incoming commands and responses.
- The signaling protocols (e.g., E&M) must be terminated and the information must be extracted.
- The signaling information must be mapped into a format that can be used to establish a session across the packet network.
- Telephone numbers (E.164 dial addresses) must be converted into IP addresses (with the potential need for an external reference to a directory service). Two approaches to dialing are being used: single stage (dial the destination number and use automatic route selection functions), and two stage (dial the VoIP gateway number, then dial the real destination).

Needless to say, the software used in VoIP devices must also be supported by a real-time operating environment and provided with the ability to communicate among the modules and with the external world. Implementation of protocols is another area where development time, testing, and risk can be minimized through the use of embedded software. The objective should always be to develop new ways to optimize the use of standard protocol software, not to re-invent basic functions that require extensive testing for standards compliance and product interoperability.

The ability to digitize and process voice streams using self-contained software building blocks is the key to success with VoIP implementation. VoIP equipment should comply with the H.323 standard which has been defined by the ITU to describe terminals, equipment, and services for multimedia communication over networks (such as LANs or the Internet) that do not provide a guaranteed QoS. H.323 is a family of software-based standards that define various options for compression and call control. Figure 5 illustrates the functional components of terminals that use the H.323 standards.

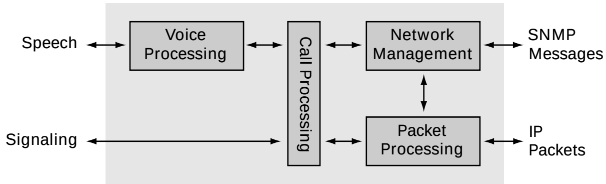

Figure 5: Voice Gateway/Terminal Functions

The following table lists the various standards that have been adopted as part of the H.323 family. These are implemented as part of the software described above and need to be "open," that is, implementations from multiple vendors must be compatible.

H.323 and Related Recommendations

Recommendation	Brief Description
H.323	Document called “Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service” (November, 1996)
H.225	Call control messages including signaling, registration, and admissions, and for the packetization and synchronization of media streams including both point-to-point and multipoint calls
H.245	Messages for opening and closing channels for media streams, and other commands, requests, and indications
H.261	Video codec for audio visual services at multiples of 64 Kbps
H.263	Specifies a codec for video over the PSTN
G.711	Audio codec for 3.1 Kbps bandwidth over 48,56, and 64 K bps channels (normal telephony)
G.722	Audio codec for 7 Kbps bandwidth over 48,56, and 64 Kbps channels
G.728	Audio codec for 3.1 Kbps bandwidth over 16 Kbps channels
G.723, G.723.1	Audio codec for 3.1 Kbps bandwidth over 5.3 and 6.3 Kbps channels (G.723.1 has been selected by the VoIP Forum for use with VoIP)
G.729, G.729a	Audio codec for 3.1 Kbps bandwidth over 8 Kbps channels (adopted by the Frame Relay Forum for voice over Frame Relay)
T.120	Data and conference control

Although H.323 is the recognized standard for VoIP terminals, there are additional standards that are more appropriately suited for client applications, such as IP phones. As H.323 was originally designed for the desktop, a higher priority was given to rich functionality, rather than resource allocation. This has given rise to alternative protocols that can interoperate with H.323, and whose orientation are to be more “lightweight” in nature. These are listed in the table below.

Other VoIP Protocols

Protocol	Brief Description
SGCP (Simple Gateway Control Protocol)	Simple UDP-based protocol for managing endpoints and connections between endpoints.
SAP (Session Announcement Protocol)	Protocol used by multicast session managers to distribute a multicast session description to a large group of recipients
SIP (Session Initiation Protocol)	Protocol used to invite an individual user to take part in a point-to-point or unicast session
RTSP (Real-Time Streaming Protocol	Protocol used to interface to a server that will provide real-time data
SDP (Session Description Protocol)	Describes the session for SAP, SIP and RTSP

An embedded VoIP software solution should be designed with well-defined interfaces between the modules (for example, the interface between the Voice Processing performed on a DSP and the rest of the system must be clearly defined). This also allows the same device to be configured to work with IP, Frame Relay, or ATM without a complete re-design.

Implementing VoIP in Systems

The deployment of a VoIP infrastructure for public use involves much more than simply adding compression functions to an IP network. Anyone must be able to call anyone else, regardless of location and form of network attachment (telephone, wireless phone, PC, or other device). Everyone must believe the service is as good as the traditional telephone network. Long-term costs (as opposed to simply avoiding regulatory costs) must make the investments in the infrastructure worthwhile. Any new approach to telephony will naturally be compared to the incumbent and must be seen as being no worse (i.e., the telephone still has to work if the power goes off), implying that all necessary management, security, and reliability functions are included.

Figure 6 is a refinement of Figure 1 that includes the placement of the VoIP gateway and the system level support functions that are integral to a high-quality VoIP system. The VoIP Gateway is shown here as a separate component, but it could also be integrated into the voice switch (a PBX or CO Switch) or into an IP Switch.

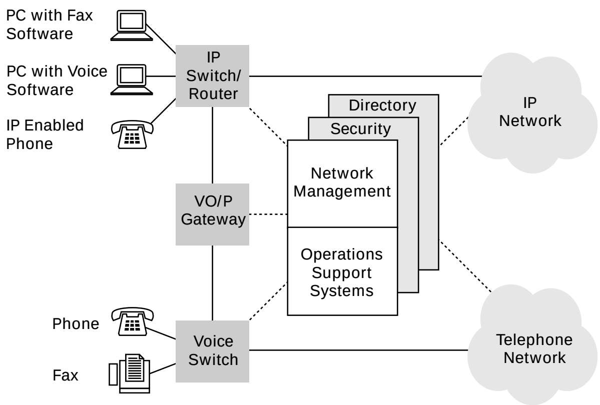

Figure 6: A Combined PSTN/VoIP System

Some of the functions that are required for a VoIP system include:

a) Fault Management: One of the most critical tasks of any telecommunications management system is to assist with the identification and resolution of problems and failures. Full SNMP management capabilities using MIBs should be provided for enterprise-level equipment. Integrating the management facilities of the telephone and data systems using TMN-based standards is essential for carrier-class systems.

b) Accounting/Billing: VoIP gateways must keep track of successful and unsuccessful calls. Call detail records that include such information as call start/stop times, dialed number, source/destination IP address, packets sent and received, etc. should be produced. This information would preferably be processed by the external accounting packages that are also used for the PSTN calls. The end user should not need to receive multiple bills.

c) Configuration: An easy-to-use management interface is needed to configure the equipment (even while the service is running). A variety of parameters and options are involved. Examples include: telephony protocols, compression algorithm selection, dialing plans, access controls, PSTN fallback features, port arrangements, Internet timers, etc.

d) Addressing/Directories: Telephone numbers and IP addresses need to be managed in a way that is transparent to the user. PCs that are used for voice calls may need telephone numbers, IP-enabled telephones will need IP addresses (or at least access to one via DHCP protocols) and Internet Directory services will need to be extended to include mappings between the two types of address.

e) *Authentication/Encryption*: VoIP offers the potential for secure telephony by making use of the security services available in TCP/IP environments. Access controls can be implemented using authentication and calls can be made private using encryption of the links.

Implementations of full-scale VoIP systems must provide all the "-abilities" that are usually taken for granted in open systems (including the PSTN). These include:

- *Interoperability*: In a public networking environment different products will need to interwork if any-to-any communications is to be possible. Using common software that has been tested for conformance to all applicable standards (such as for compression) can significantly reduce the cost of product development. Interconnection of VoIP to the PSTN also involves meeting the specific standards for telephone network access.
- *Reliability:* The VoIP network, whether by design or through management, should be fault tolerant with only a very small likelihood of complete failure. In particular, the gateway between the Telephone and VoIP systems needs to be highly reliable.
- *Availability:* Sufficient capacity must be available in the VoIP system and its gateways to minimize the likelihood of call blocking and mid-call disconnects. This will be especially important when the network is shared with data traffic that may cause congestion. Mechanisms for admission control should be available for both the voice and data traffic, with prioritization policies set.
- *Scalability*: There is potential for extremely high growth rates in VoIP systems, especially if they prove the equal of the PSTN at much lower cost. VoIP systems must be flexible enough to grow to very large user populations, to allow a mix of public and private services and to adapt to local regulations. The need for large numbers of addressable points may force the use of improved Internet protocols such as IPv6.
- *Accessibility*: Telephone systems assume that any telephone to call any other telephone and to allow conferencing of multiple telephones across wide areas. This will be driven by functions that map between telephone numbers and other types of packet network address, specifically IP addresses. There must, of course, exist gateways that allow every device to be reachable.
- *Viability*: Many are claiming significant economic advantages to the implementation of VoIP. These are often based on flat rate prices for Internet service, the fact that services such as the "Internet 911" are not required and that there is no regulatory prohibition against interconnection of telephone systems with IP systems. Also assumed is that higher performance compression will not be used in the telephone network to reduce costs. If circumstances change, the motivation for VoIP purely for cost avoidance reasons may change also.

Summary

Data traffic has traditionally been forced to fit onto the voice network (using modems, for example). The Internet has created an opportunity to reverse this integration strategy - voice and facsimile can now be carried over IP networks, with the integration of video and other multimedia applications close behind. The Internet and its underlying TCP/IP protocol suite have become the driving force for new technologies, with the unique challenges of real-time voice being the latest in a series of developments.

Telephony over the Internet cannot make compromises in voice quality, reliability, scalability, and manageability. It must also interwork seamlessly with telephone systems all over the world. Just about all of today's network devices will need to be voice-enabled (and eventually multimedia-enabled). Future extensions will include innovative new solutions including conference bridging, voice/data synchronization, combined real-time and message-based services, text-to-speech conversion and voice response systems.

The market for VoIP products is established and is beginning its rapid growth phase. Producers in this market must look for ways to improve their time-to-market if they wish to be market leaders. Buying and integrating pre-defined and pre-tested software (instead of custom building everything) is one of the options. Significant benefits of the "buy vs. build" approach include reduced development time, simplified product integration, lower costs, off-loading of standards compliance issues, and fewer risks. Software that is known to conform to standards, has built-in accommodation for differences in national telephone systems, has already been optimized for performance and reliability, and has "plug and play" capabilities can eliminate many very time-consuming development tasks.

CASE STUDY: How Entrata's Partnership with Telogy Networks Filled a Niche in the Voice over Packet Market

Entrata Communications Corporation was founded in 1996 to develop products facilitating the integration of voice, video, and data networking for business services. Today, Entrata develops advanced multi-service access products for use on copper, fiber-optic or other facilities employed in local telecommunications networks. Its primary customers are telecommunication companies such as local exchange carriers, competitive access providers, and Internet service providers, as well as large corporate enterprises.

Entrata's early view of the emerging Voice over Packet market was that an explosion of new telecom companies, coupled with a shift and expansion in direction for many of the regional bell operating companies (RBOCs), would lead to the creation of a single network structure supporting voice, video, and data.

"We recognized an exciting opportunity in the emerging Voice over Packet market," remarked Angelo Compagnoni, Vice President of Marketing for Entrata. "The major challenge for us was how to integrate a number of technologies at an acceptable price in a leading edge market."

The company founders felt that there was a lack of reasonably priced access devices supporting an integration of voice, video, and data. In their view, an opportunity existed for Entrata to develop integrated access devices for businesses to connect to the corporate network.

"Our primary philosophy was that the issue shouldn't be whether information travels across IP, Frame Relay, or ATM networks," recalled Compagoni. "We really didn't want the focus of our product to be on the network transport technology."

Compagnoni related Entrata's opportunity to the role of a service provider. Where a service provider can offer a variety of services to users and enable them to connect to their network of choice, Entrata purported to deliver those same kinds of options to the market with its proposed integrated access device.

Analysis of the highly competitive market revealed that most of Entrata's competitors were focusing on individual aspects of product development. For example, some companies were developing Voice over IP applications while others were producing ATM-only products. By pledging to deliver megabit access to anyone and everyone, Entrata's mission was to develop a single access product designed to allow users to connect to *any* network.

"Partnering with a software company like Telogy Networks was key to our success," Compagnoni explained. "It enabled us to go off and focus on our own core competencies, and most importantly, it saved us critical time to market in developing our product."

One of Entrata's major challenges in developing a Voice over Packet-enabled networking product was finding the key resources to work on the project. By working with an outside vendor like Telogy and not having to assemble an engineering team with the right experience and qualifications, Compagnoni estimated that it saved the company at least twelve months.

"The key things we were looking for were a high quality partner, good support capabilities, and a software solution equal to or better than what we would have developed in-house," said Compagnoni. "Market data has shown us that in our industry, delivering high-performance hardware with top quality software is what customers are seeking. We view Telogy as an extension of our development arm. Moving forward, we feel we can do some unique things together."

Entrata has incorporated Telogy's Golden Gateway™ Voice over IP (VoIP) embedded communications software into the company's debut multiservice access line. The product is designed for use on copper or fiber-optic lines for circuit-switched, packet based, or ATM networks, to facilitate high bandwidth connectivity between major user sites and backbone digital networks utilized worldwide. Entrata's products provide a single access point to a diverse variety of services and applications over a single integrated circuit.

This partnership gives Entrata's customers the benefits of Telogy's market leading Golden Gateway VoIP embedded software, which boasts the highest quality sound and reliability available in the market today.

Golden Gateway software, embedded on a digital signal processor (DSP) and microprocessor, provides communications equipment developers scalable Voice and Fax over IP, ATM, or Frame Relay solutions. Telogy's multi-function, multi-channel software capabilities support up to four simultaneous channels and allow multiple voice and fax calls to be dynamically allocated and processed in a single DSP.

Telogy's software combined with Entrata's high performance architecture and multiple transport protocol support provides a "top notch solution," remarked Campganonni. "Since we're representing Telogy's software in our system, it has to work and it has to be good. That's why the decision to incorporate Golden Gateway into our product was so critical."

Glossary

AAL—ATM Adaptation Layer (AAL) The standards layer that allows multiple applications to have data converted to and from the ATM cell. A protocol used that translates higher layer services into the size and format of an ATM cell.

AAL 2—Is used with time-sensitive, variable bit rate traffic such as packetized voice.

AAL 5—Accommodates bursty LAN data traffic with less overhead than AAL 3/4.

Available Bit Rate (ABR)—QoS class defined by the ATM Forum for ATM networks. ABR is used for connections that do not require timing relationships between source and destination. ABR provides no guarantees in terms of cell loss or delay, providing only best-effort service. Traffic sources adjust their transmission rate in response to information they receive describing the status of the network and its capability to successfully deliver data.

Adaptive Differential Pulse Code Modulation (ADPCM)—Process by which analog voice samples are encoded into high-quality digital signals.

Address Resolution Protocol (ARP)—Internet protocol used to map an IP address to a MAC address. Defined in RFC 826.

Asynchronous Transfer Mode (ATM)—(1) The CCITT standard for cell relay wherein information for multiple types of services (voice, video, data) is conveyed in small, fixed-size cells. ATM is a connection-oriented technology used in both LAN and WAN environments. (2) A fast-packet switching technology allowing free allocation of capacity to each channel. The SONET- synchronous payload envelope is a variation of ATM. (3) ATM is an international ISDN high-speed, high-volume, packet switching transmission protocol standard. ATM currently accommodates transmission speeds from 64 Kbps to 622 Mbps.

Central Office (CO)—(1) A local telephone company office which connects to all local loops in a given area and where circuit switching of customer lines occurs. (2) A local Telephone Company switching system where a Telephone Exchange Service customer station loops are terminated for purposes of interconnection to each other and to trunks. In the case of a Remote Switching Module (RSM), the term Central Office designates the combination of the Remote Switching Unit and its Host.

Channel Associated Signaling (CAS)—Signaling system in which signaling information is carried within the bearer channel.

Circuit-Switched Network—Network that establishes a temporary physical circuit until it receives a disconnect signal.

Circuit Emulation Services (CES)— ATM support mode emulating TDM services. Circuit emulation reduces apparent delay, but is limited to a point-to-point environment.

Code-Excited Linear Predictive Coding (CELP)—A voice compression algorithm used at 8 kbps.

Coder/Decoder (Codec)—Equipment to convert between analog and digital information format. Also may provide digital compression and switching functions. Primarily used to describe video equipment performing this function.

Committed Information Rate (CIR)—The transport speed the frame relay network will maintain between service locations.

Common Channel Signaling—A method of signaling in which signaling information relating to a multiplicity of circuits, or relating to a function for network management, is conveyed over a single channel by addressed messages.

Competitive Local Exchange Carrier (CLEC)—A company that builds and operates communication networks in metropolitan areas and provides its customers with an alternative to the local telephone company.

Compression—Reducing the size of a data set to lower the bandwidth or space required for transmission or storage.

Computer Telephony Integration (CTI)—The name given to the merger of traditional telecommunications (PBX) equipment with computers and computer applications. The use of Caller ID to automatically retrieve customer information from a database is an example of a CTI application.

Connectivity — The ability of a device to connect to another: This includes not only the physical issues associated with the busses, connector topologies, and other such matters, but also the support of the protocols required to pass data successfully over the physical connection.

Constant Bit Rate (CBR)—QoS class defined by the ATM Forum for ATM networks. CBR is used for connections that depend on precise clocking to ensure undistorted delivery.

Data-link Connection Identifier (DLCI)—Value that specifies a PVC or SVC in a Frame Relay network. In the basic Frame Relay specification, DLCIs are locally significant (connected devices might use different values to specify the same connection). In the LMI extended specification, DLCIs are globally significant (DLCIs specify individual end devices).

Dedicated Circuit—A transmission circuit leased by one customer for exclusive use around the clock. Also called a private line, or leased line.

Dedicated Line—(1) A communications circuit or channel provided for the exclusive use of a particular subscriber. Dedicated lines are used for computers when large amounts of data need to be moved between points. Also known as a “private line.” (2) A transmission circuit installed between two sites of a private network and “open,” or available, at all times.

Delay—(1) Amount of time a call spends waiting to be processed. (2) Basically, the time the information takes to transit a network or network segment. Differential delay is the difference in transit time between data taking separate transmission paths - for example, inverse-multiplexed T1s employing different routes through T1 networks.

Dial Tone Multi-Frequency (DTMF)—The set of standardized, superimposed tones used in telephony signaling - as generated by a touch tone pad.

Digital Signal Processor (DSP)—A high-speed coprocessor designed to do real-time signal manipulation.

Dynamic Host Configuration Protocol (DHCP)—Provides a mechanism for allocating IP addresses dynamically so that addresses can be reused when hosts no longer need them.

Ear and Mouth (E and M) Signaling—Trunk signaling between a PBX and a CO used to seize a line, forward digits, release the line, etc.

Echo Control—The control of reflected signals in a telephone transmission path.

File Transfer Protocol (FTP)—(1) An IP application protocol for transferring files between network nodes. (2) An Internet protocol that allows a user on one host to transfer files to and from another host over a network.

Foreign Exchange Office (FXO)—A remote Telephone Company Central Office used to provide local telephone service over dedicated circuits from that office to the user's local central office and premises.

Foreign Exchange Station (FXS)—That user premises to which a foreign exchange circuit is connected.

Frame Relay—High-performance interface for packet-switching networks. Considered more efficient than X.25 which it is expected to replace. Frame relay technology can handle "bursty" communications that have rapidly changing bandwidth requirements.

H.323—A standard approved by the International Telecommunication Union (ITU) that defines how audiovisual conferencing data is transmitted across networks. In theory, H.323 should enable users to participate in the same conference even though they are using different videoconferencing applications. Although most videoconferencing vendors have announced that their products will conform to H.323, it's too early to say whether such adherence will actually result in interoperability.

Implementation Agreement—The formal vendor agreement specifying the details of a system deployment.

Interexchange Carrier (IXC) or Interexchange Common Carrier—(1) Any individual, partnership, association, joint-stock company, trust, governmental entity, or corporation engaged for hire in interstate or foreign communication by wire or radio, between two or more exchanges. (2) A long-distance telephone company offering circuit-switched, leased-line or packet-switched service or some combination.

International Telecommunications Union-Telecommunications Standards Sector (ITU-TSS)—The new name for CCITT. An international standards body which is a committee of the ITU, a UN treaty organization.

Internet—(note the capital "I") The largest internet in the world consisting of large national backbone nets (such as MILNET, NSFNET, and CREN) and a myriad of regional and local campus networks all over the world. The Internet uses the Internet protocol suite. To be on the Internet you must have IP connectivity, i.e., be able to Telnet to or ping other systems. Networks with only e-mail connectivity are not actually classified as being on the Internet.

Internet Protocol (IP)—A Layer 3 (network layer) protocol that contains addressing information and some control information that allows packets to be routed. Documented in RFC 791.

Internet Service Provider (ISP)—(1) Any of a number of companies that sell Internet access to individuals or organizations at speeds ranging from 300 bps to OC-3. (2) A business that enables individuals and companies to connect to the Internet by providing the interface to the Internet backbone.

Internet Telephony—Generic term used to describe various approaches to running voice telephony over IP.

Internetwork—A collection of networks interconnected by routers that function (generally) as a single network. Sometimes called an internet, which is not to be confused with the Internet.

Intranet—A private network inside a company or organization that uses the same kinds of software that you would find on the public Internet, but that is only for internal use. As the Internet has become more popular, many of the tools used on the Internet are being used in private networks; for example, many companies have Web servers that are available only to employees.

ISDN BRI—A digital access line that is divided into three channels. Two of the channels, called B channels, operate at 64 Kbps and are always used for data or voice. The third D channel is used for signaling at 16 Kbps.

ISDN PRI—Based physically and electrically on an E1 circuit, but channelized so that two channels are used for signaling and 30 channels are allocated for user traffic. ISDN PRI is available in E1 and T1 frame formats, depending on country.

Latency—The delay between the time a device receives a frame and the frame is forwarded out of the destination port.

Local Area Network (LAN)—A network covering a relatively small geographic area (usually not larger than a floor or small building). Compared to WANs, LANs are usually characterized by relatively high data rates. (2) Network permitting transmission and communication between hardware devices, usually in one building or complex.

Management Information Base (MIB)—A database of information on managed objects that can be accessed via network management protocols such as SNMP and CMIP.

Mean Opinion Scores (MOS)—A system of grading the voice quality of telephone connections. The MOS is a statistical measurement of voice quality, derived from a large number of subscribers judging the quality of the connection.

Million Instructions Per Second (MIPS)—A measure of a computer's speed or power.

MUX— A multiplexing device. A mux combines multiple signals for transmission over a single line. The signals are demultiplexed, or separated, at the receiving end.

Off-Hook—The active condition of Switched Access or a Telephone Exchange Service line.

On-Hook—The idle condition of Switched Access or a Telephone Exchange Service line.

Operations Support System (OSS)—The computerized platform and related software used to support the operations of a network.

Overhead (OH)—Bits in frame or cell required for framing, CRC, routing, etc.

Packet—(1) A logical grouping of information that includes a header and (usually) user data. (2) Continuous sequence of binary digits of information is switched through the network and an integral unit. Consists of up to 1024 bits (128 octets) of customer data plus additional transmission and error control information.

Packet Loss Rate—The measure loss, over time, of data packets as a percentage of the total traffic transmitted.

Permanent Virtual Circuit (PVC)—Virtual circuit that is permanently established. PVCs save bandwidth associated with circuit establishment and tear down in situations where certain virtual circuits must exist all the time.

Plain Old Telephone System (POTS)—What we consider to be the "normal" phone system, used with modems. Does not include leased lines or digital lines.

Private Branch Exchange (PBX)—A small telephone network for customer premises. Provides local connectivity and switching and connections to the wide area voice network.

Protocol—(1) A formal description of a set of rules and conventions that govern how devices on a network exchange information. (2) Set of rules conducting interactions between two or more parties. These rules consist of syntax (header structure) semantics (actions and reactions that are supposed to occur) and timing (relative ordering and direction of states and events). (3) A formal set of rules.

Protocol Stack—Related layers of protocol software that function together to implement a particular communications architecture. Examples include AppleTalk and DECnet.

Public Switched Telephone Network (PSTN)—General term referring to the variety of telephone networks and services in place worldwide.

Pulse Code Modulation (PCM)—Transmission of analog information in digital form through sampling and encoding the samples with a fixed number of bits.

QSIG—Signaling system between a PBX and CO, or between PBXs uses to support enhanced features such as forwarding and follow me.

Quality of Service (QoS)—Measure of performance for a transmission system that reflects its transmission quality and service availability.

Real-Time Transport Protocol (RTP)—The standard protocol for streaming applications developed within the IETF.

Resource Reservation Protocol (RSVP)—A protocol that supports the reservation of resources across an IP network. Applications running on IP end systems can use RSVP to indicate to other nodes the nature (bandwidth, jitter, maximum burst, and so on) of the packet streams they wish to receive.

RTP Control Proctocol (RTCP)—A protocol providing support for applications with real-time properties, including timing reconstruction, loss detection, security, and content identification. RTCP provides support for real-time conferencing for large groups within an Internet, including source identification and support for gateways (like audio and video bridges) and multicast-to-unicast translators.

Switched Virtual Circuit (SVC)—Virtual circuit that is dynamically established on demand and is torn down when transmission is complete. SVCs are used in situations where data transmission is sporadic.

Time-Division Multiplexing (TDM)—Technique in which information from multiple channels can be allocated bandwidth on a single wire-based on preassigned time slots. Bandwidth is allocated to each channel regardless of whether the station has data to transmit.

Transmission Control Protocol/Internet Protocol (TCP/IP)—(1) The common name for the suite of protocols developed by the U.S. Department of Defense in the 1970s to support the construction of world-wide internetworks. TCP and IP are the two best-known protocols in the suite. TCP corresponds to Layer 4 (the transport layer) of the OSI reference model. It provides reliable transmission of data. IP corresponds to Layer 3 (the network layer) of the OSI reference model and provides connectionless datagram service. (2) The collection of transport and application protocols used to communicate on the Internet and other networks

Unspecified Bit Rate (UBR)—QoS class defined by the ATM Forum for ATM networks. UBR allows any amount of data up to a specified maximum to be sent across the network, but there are no guarantees in terms of cell loss rate and delay.

Variable Bit Rate (VBR)—Applications which produce traffic of varying bit rates, like common LAN applications, which produce varying throughput rates.

Virtual Circuit (VC)—Logical circuit created to ensure reliable communication between two network devices. A virtual circuit is defined by a VPI/VCI pair, and can be either permanent (a PVC) or switched (an SVC). Virtual circuits are used in Frame Relay and X.25. In ATM, a virtual circuit is called a virtual channel.

Voice Activity Detection (VAD)—Saves bandwidth by transmitting voice cells only when voice activity is detected.

Voice Over the Internet Protocol (VoIP)—The developing standard for transmitting voice signals over the IP based Internet.

VPN—Virtual Private Network.

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Albert Einstein

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