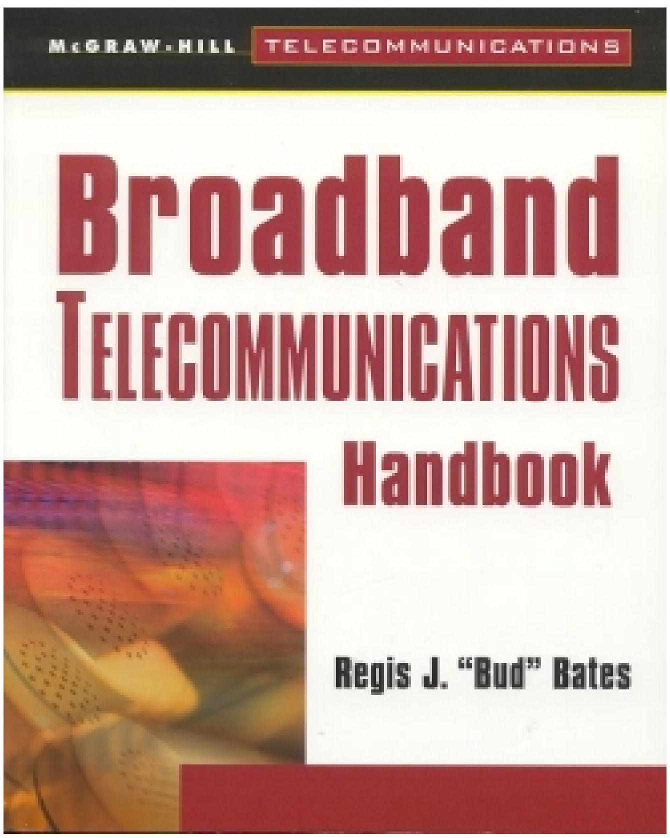
FURTHER READING:

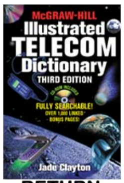
As a preview for further reading, the following reference has been provided from the pages of the book below:

Title: Broadband Telecommunications Handbook
Author: Regis J. "Bud" Bates
Publisher: McGraw-Hill

ISBN: 0071346481



Voice over the Internet Protocol



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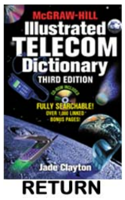
The public telephone network and the circuit switching systems are usually taken for granted. Over the past few decades, they have grown to be accepted as almost 100 percent reliable. Manufacturers built in all the necessary stopgaps to prevent downtime and increase lifeline availability of telephony. It was even assumed that when the commercial power failed, the telephony business continued to operate. This did not happen accidentally, but through a very concerted development cycle jointly by the carriers and the manufacturers alike.

Access to a low-cost, high-quality worldwide network is considered essential in today's world. Anything that would jeopardize this access is treated with suspicion. A new paradigm is beginning to occur because more of our basic communications occur in digital form, transported via packet networks such as IP, ATM, and Frame Relay. Packet data networking has matured over the same period of time that the voice technologies were maturing. The old basic voice and basic data networks have been replaced with highly reliable networks that carry voice, data, video, and multimedia transmissions. Proprietary solutions manufactured by various providers have fallen to the side, opening the industry to a more open and standards-based environment.

In 1996 there was as much data traffic running on the networks as there was voice traffic. Admittedly, industry pundits are all still saying that 90 percent of the revenue in this industry is generated by voice applications. This may be an accounting problem, because on average 57 percent of all international calls originating in North America and going to Europe and Asia are actually carrying fax, not voice. Yet, they are considered dial-up voice communications transmissions because of the methodology used. Moreover, the voice market is growing at approximately 3–4 percent per year, whereas data is growing at approximately 30 percent per year. 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 important 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. Delivering high-quality commercial products, establishing public services, and convincing users to buy into the vision are all still in their infancy. The evolution of all networks begins this way, so there is no mystique in it.

IP telephony will also have to change, somewhat. We will expect it to deliver interpersonal communications that end users are already accustomed to using. These added capabilities will include (but not be limited to) the following:



- *Calling Line ID* (CLID)
- Three-way calling
- Call transfer
- Voice mail
- Voice-to-text conversions

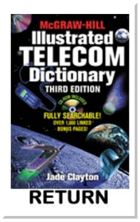
Users are very comfortable with the services and capabilities delivered by the telephone companies on the standard dial-up telephone set, using the touch-tone pad. IP telephony will have to match these services and ease-of-use functions in order to be successful.

IP telephony will not replace the circuit switched telephone networks overnight; this will be a co-existence for the near future. In 2003 analysts expect that the amount of IP telephony will amount to 3% of all voice traffic domestically and approximately 10–15% of international traffic. This amounts to 50 billion minutes of traffic, so it is consequential. One must be prepared for both alternatives to carrying voice in the next decade. Thus, the differences between the two opposing network strategies will be ironed out, and the world may shift into a packet switched voice network over the next decade.

Quality of Service

One of the arguments against IP based telephony today is the lack of *Quality of Service* (QoS). The manufacturers and developers will have to overcome the objections by producing transmission systems that will assure a quality of service for lifeline voice communications. Mission critical applications in the corporate world will also demand the ability to have specified grade of service available. Applications that will be critical include some of the technologies we discussed in Chapters 6 and 8. The CTI applications with call centers being Web-enabled, interactive voice recognition, response, and other speech activated technologies will demand a quality of service to facilitate the use of these systems. Each will demand the grade and quality of service expected in the telephone industry.

Another critical application for IP telephony will be the results of quality of voice transmission. Noisy lines, delays in voice delivery, and clicking and chipping all tend to frustrate users on a voice network. Packet data networks carrying voice services today may produce the same results. Therefore, overcoming these pitfalls is essential to the success and acceptance of Voice over IP telephony applications. Merely installing more capacity (bandwidth) is not a solution to the problem, but it is a temporary fix. Instead, developers must concentrate on delivering several solution sets to the industry such as those shown in Table 30-1.



The QoS requirements for IP Telephony can therefore be summarized as shown in Table 30-2, which considers the layered approach that vendors will be aggressively pursuing. IP Telephony datagrams entering the network will be treated with a priority to deliver the QoS expected by the end user. The routers and switches in the network will assign a high priority marking on each datagram carrying voice, and treat these datagrams specially. Queues throughout the network will be established with variable treatments to handle the voice datagrams first, followed by the data datagrams.

Table 30-1

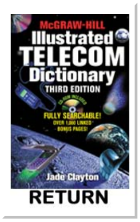
Different approaches to QoS

Strategy	**Description**
Integrated Services Architectures (Int-Serv)	Int-Serv includes the specifications to reserve network resources in support of a specific application. Using the RSVP protocol, the application or user can request and allocate sufficient bandwidth to support the short- or long-term connection. This is a partial solution because Int-Serv does not scale well, because each networking device (routers and switches) must maintain and manage the information for each flow established across their path.
Differentiated Services (Diff-Serv)	Easier to use than Int-Serv, Diff-Serv uses a different mechanism to handle the flow across the network. Instead of trying to manage individual flows and per-flow signaling needs, Diff-Serv uses DS bits in the header to recognize the flow and the need for QoS on a particular datagram-by-datagram basis. This is more scalable than Int-Serv, and does not rely solely on RSVP to control flows.
802.1p Prioritization	The IEEE standard specifies a priority scheme for the Layer 2 switching in a switched-LAN. When a packet leaves a subnetwork or a domain, the 802.1p priority can be mapped to Diff-Serv to satisfy the layer 2 switching demands across the network.

Table 30-2

QoS Requirements for IP telephony

Layer Addressed	Technique	Variable
1	Physical Port	Variations of port definitions or the prioritization of port interfaces based on application
2	IEEE 802.1p Bits	Dedicated paths or ports for high bandwidth applications, but very expensive to maintain
3	IP addressing	RSVP protocol (Int-Serv) DS bits in the IP header (Diff-Serv)

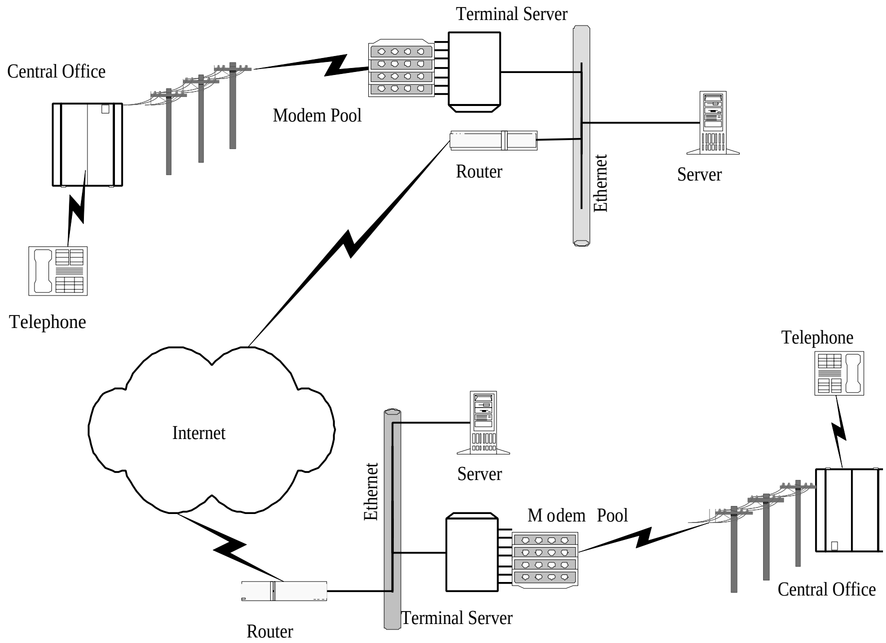


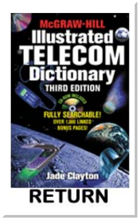
Given the differences, why should we even consider the use of IP telephony? If the IP telephony world is only going to account for 3 percent of domestic traffic by 2003, is it worth all the hassles? The answer is a mixed bag, but overall the efficiencies of VoIP will outweigh the need to develop better control mechanisms to satisfy the telephony industry.

Applications for VoIP

Voice communications will remain a basic staple for the industry. The public switched telephone networks will not be replaced, or dramatically changed for that matter, in the near term. The immediate need then for VoIP providers is to deliver equal services, similar to the PSTN, at a lower cost of operation and to offer a suitable competitive product as an alternative. This alternative will increase the competition in the industry and force stodgy providers into a new era of meeting the consumer's demands. The first and foremost yardstick that everyone is using is the pricing model. Although switched voice is now down to $.05– .10 per minute, just what is the equilibrium price that should be used? When we consider the IECs' costs per minute is less than $.00125, the $.10 per minute charge they levy on corporate or residential consumers appears excessive. These pricing models are also changing as new technology is implemented in the SONET and DWDM architecture as discussed in previous chapters. In Figure 30-1, we see one of the basic scenarios of how the new providers may challenge the existing infrastructure providers of telephony service. This assumes that a telephone to telephone access method is used through the networks. This, by the way, is the preferred method because it uses the tools and

Figure 30-1
Telephone to telephone through the IP network



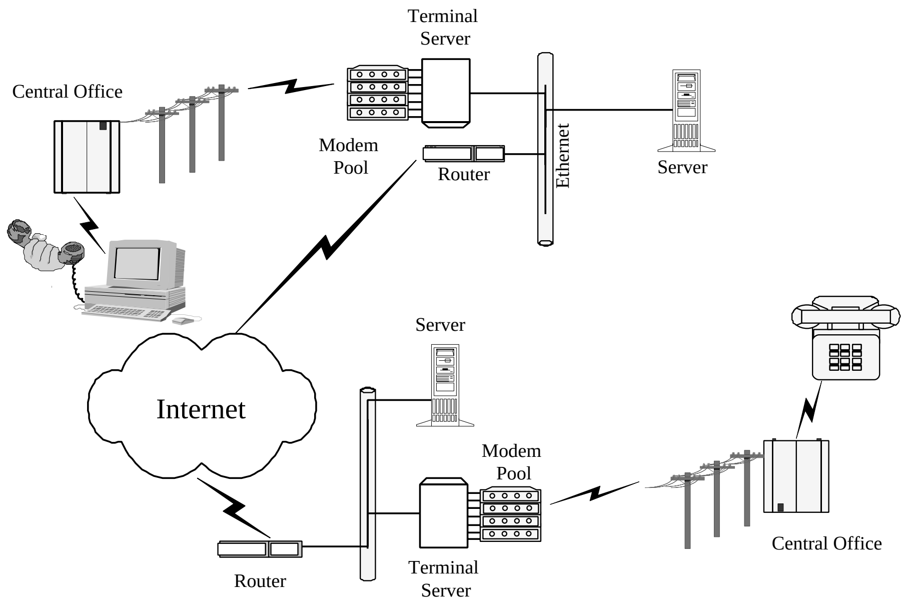


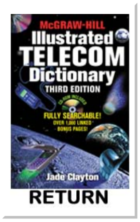
techniques to which everyone is accustomed. The basic telephone set has become a staple in the business by having an ergonomically designed device that is easy to use. This is the best way to handle the voice over the Internet or IP for a call. The telephone sets are ergonomically designed to have the mouthpiece close to the human's mouth and the ear piece that helps to screen out a lot of the ambient noise in the room. Moreover, the sets are not actually doing any of the compression or conversions; these are handled by the devices in the network such as gateways and gatekeepers, specifically designed for this task.

VoIP can be used for just about any requirement for a telephony call. It can be used for fax and for video applications as well. From a single point-to-point call to a multiparty conference call, and a Web-enabled call center application, the manufacturers have to devise tools and techniques to support the masses.

A second use for VoIP is shown in Figure 30-2, which uses a PC to telephone call. This is slightly less desirable because the PC uses an integrated microphone and speakers for the most part. Because they are not designed specifically for this application, they pick up all the noise in the background. Further, to prepare the real-time voice through the PC, the compression and conversion process occurs inside the PC, causing some clipping and

Figure 30-2
PC to telephone calls over the IP protocol



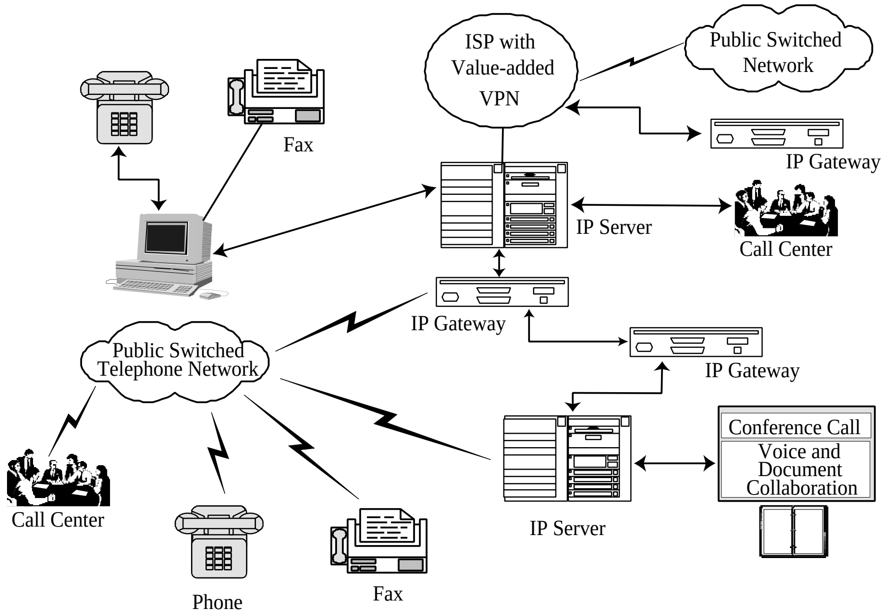


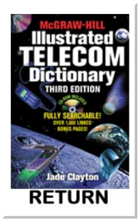
chipping, echo, and other forms or latency and distortion. It is not the best solution in the industry, but it does work.

The third way to handle this is to use a PC to PC connection, which is the worst of the three scenarios. This one introduces the use of the PC substandard technology on both ends of the connection, compounding the problems of echo, delay, clipping, and chipping. This is not the best way to experiment with the technology. Moreover, I have seen people using PC to PC communications, whereby the caller places a telephone call over the PSTN to the party desired and tells the recipient to turn on their PC so they can be called. This is ludicrous: If the IP system does not know whether the called party is active and powered up, then you have to place a telephone call to tell the person to turn on their PC. Why not just have the conversation while you have the called party on the line? This is just an aberration of the technology, but it highlights the fact that VoIP is not yet ready for prime time. There are several tools that IP telephony can bring to the table allowing the use of VoIP to handle

- Voice to voice calls
- Fax to fax
- Voice to call centers
- Voice to conference calling arrangements and PC to PC

Figure 30-3
Various ways of using VoIP





These various paths for handling VoIP are shown in Figure 30-3. The main point here is that there are alternatives to VoIP being used as more than just a voice call. The benefits are the enabling technologies to handle myriad options for now and in the future.

VoIP equipment must be equipped with the capabilities to handle the various options designed around the PSTN and cater to a wide range of configurations. Some of the applications are summarized in Table 30-3 as a means of placing some semblance of order on the VoIP equipment market.

H.323 Protocol Suites

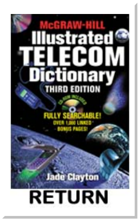
The IP telephony architecture must emulate the PSTN to accommodate the various ways of interconnecting. In Figure 30-4, the ITU has set the H.323 specification to accommodate the various ways the IP telephony industry must work. Defining gateways and gatekeepers, H.323 is a mechanism for the manufacturers to produce the necessary devices and protocol compatibility for interoperability in the PSTN and the Internet convergence.

Figure 30-5 is a different view of the H.323 specifications for the VoIP standards showing the various other protocol suites needed to support the

Table 30-3

Summary of areas where VoIP equipment must work

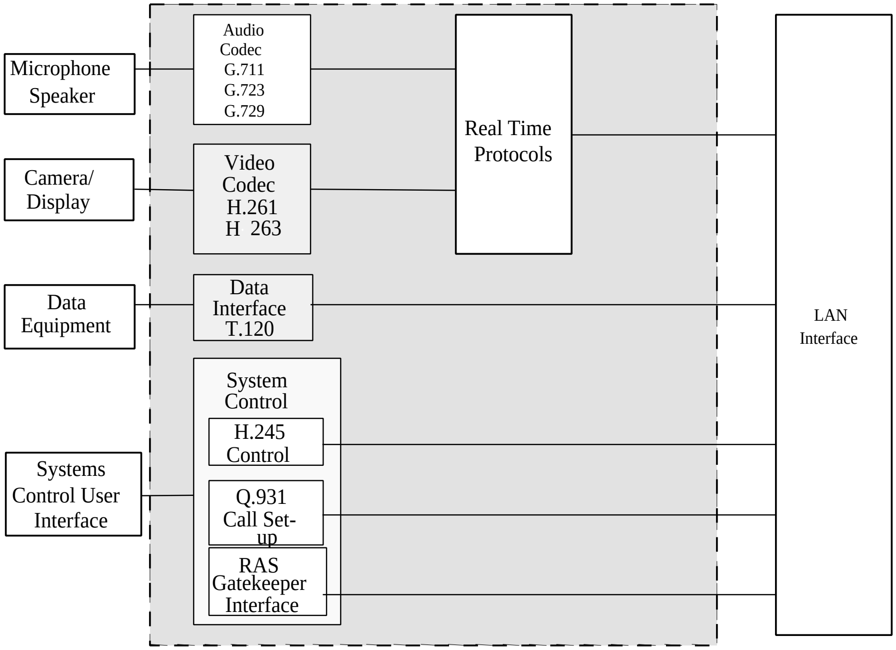
Applications	Discussion
Internet Telephones	Enhanced telephone sets, both wired and wireless, provide basic telephony services across the Internet instead of the PSTN. These may use a display that will display CLID and possibly some text messaging capabilities.
Remote Access Services	Branch office or SOHO users can gain necessary access to the corporate voice, data, video, and multimedia applications using an Intranet. The application here is the home-based worker or telemarketer.
PC-based calls from a travelling person	Using the PC with an Internet connection (dial up from a mobile PC or connected through a CATV modem in a hotel), a travelling user can access all of the corporate services. Excellent for the road warriors in today's generation.
Web-enabled Call Center Services	Call centers can be equipped with access to the Internet. As a user surfs home pages and wants to place an order for a product, or if a question arises, the user clicks a button on the PC screen to reach a real-time operator. Customer service applications like this will enhance the overall e-commerce industry and the telephony applications.
PSTN gateways	As shown in Figure 30-3, the gateway function allows the interconnection from the PSTN to the Internet over a gateway. A PC-based telephone can access the PSTN. Another application is for a USB port telephone connected to a PC that will have unlimited access to customers on the PSTN.



standard. In this case, the terminal devices running on various telephony protocols such as ISDN terminals, H.320 terminals, and video conferencing devices are all accounted for in the H.323 specification. The standard also addresses the various protocols for the gateways and gatekeepers on a LAN, as well as any compression techniques to manifest the VoIP capabilities in various points along the network.

What also has to occur in the network is to use a layer two protocol to carry the IP traffic. Traditionally, an HDLC frame or a PPP protocol frame is used. However, to assure the QoS, the standards are recommending the use of ATM at the layer two switching architecture. In most cases, ATM introduces added overhead by producing the cells to be mapped onto a SONET frame at the layer one. However, what ATM does assure is the guaranteed bandwidth to deliver the necessary QoS. Only recently has a guaranteed QoS in Frame Relay been introduced. Figure 30-6 is a summary of the bandwidth needs and the QoS issues using the ATM or FR networks

Figure 30-4 ITU H.323 specifications define the various roles in IP telephony



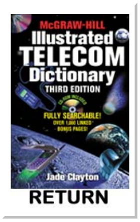
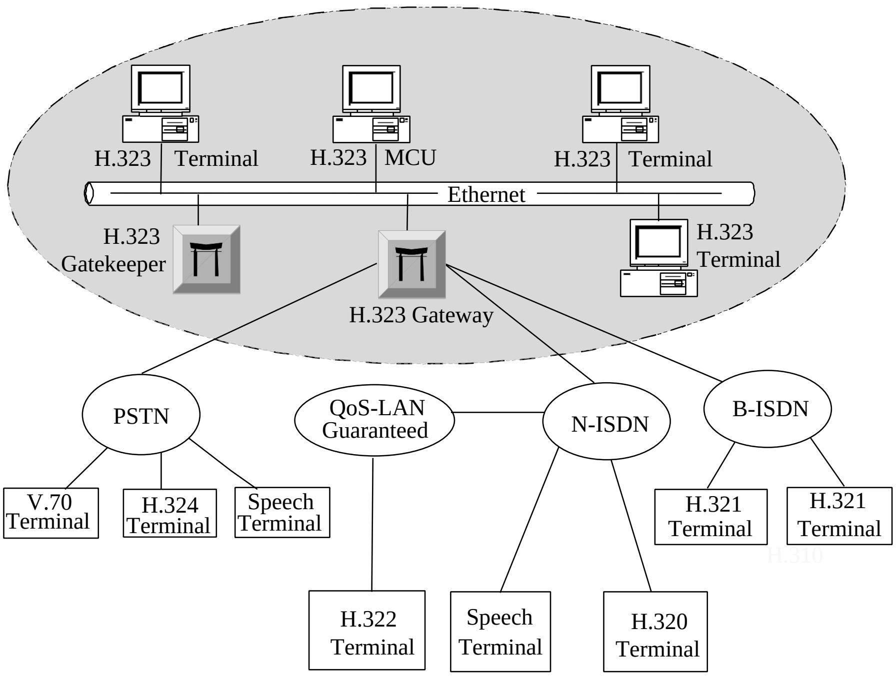


Figure 30-5 H.323 in action



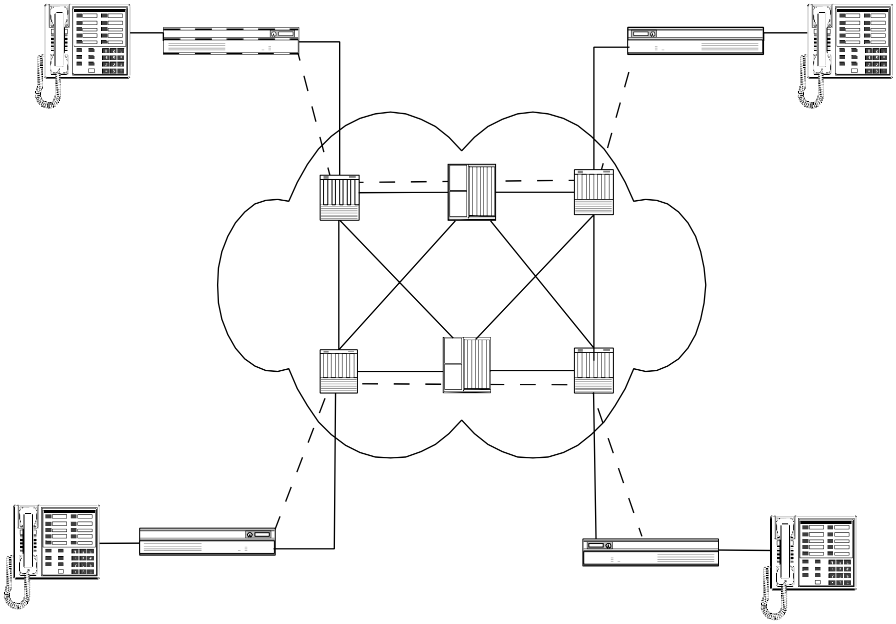
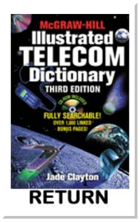


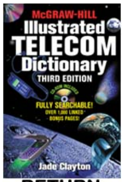
Figure 30-6 The use of ATM or Frame Relay guarantees the QoS today



to carry the VoIP traffic. It is through this combined use of Layer 2 switching and Layer 3 network addressing that the systems are able to map and multiplex the traffic on the data networks with any reliability. By the time this book is published, there will probably be three or four new approaches that will emerge, each claiming to be the better way to handle the traffic for VoIP. In most cases, they will be efficient and better than what has been available previously. That doesn't mean a thing, because each of the techniques that exist today will be implemented and upgraded or improved over time. No one technology or technique stands to be the winning or "killer application" today.

Delay and Jitter on VoIP Networks

Two of the major concerns today are the problems experienced with delay and jitter. The delay factors can be the most disconcerting. In the past, the PSTN has been delivering real-time voice across the network with an average delay of 30–50 msec. When we experimented with satellite communications in the early 1980s, the end user community was disgruntled with the potential delays for real-time voice applications. The average delay on a satellite-based network in geosynchronous orbit approximated 250 msec, which was intolerable for the end user. In the VoIP delivery today, the average response and delay factors must be dealt with. On average, if a call is placed across the Internet, the delays are from 800 msec to 2 seconds domestically. This is totally unacceptable for the user community. It certainly does not lend itself to the conducting of business conversations. When placing a call across the Internet overseas, the delay can be as much as 5 seconds! Again, this will never do.



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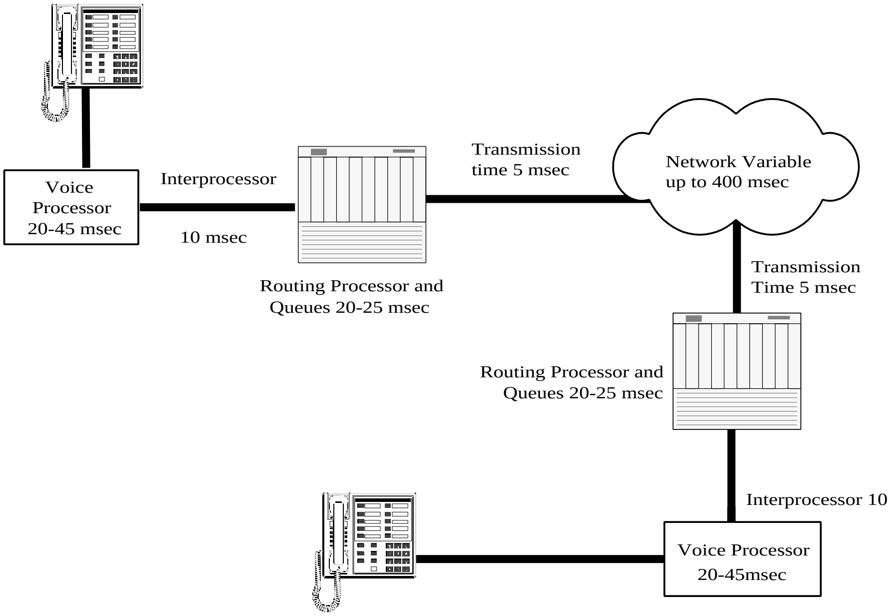
The delays have been the result of all the processing required throughout the network. One problem that results from high end-to-end communications delay is the increase in echo whenever the round trip delay exceeds 50 msec. Because echo is viewed as a quality problem, VoIP systems must accommodate echo control in order to be accepted. Talkers will overlap themselves when the delay increases if one-way transmission exceeds the 200–250 msec range. This was a major problem with satellite transmission. Too often talkers were over-talking each other, causing total chaos during the conversations. The budgeted delays for VoIP must be within the 200–250 msec range, and that is strictly the worst case. The VoIP systems have to strive for better response and delivery.

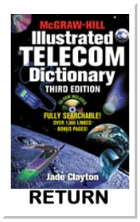
Jitter (variable delay) is a variation of the interpacket delivery introduced by the processing of each packet across the network, coupled with transmission delay across the medium. To solve the problem for real-time applications, packets must be collected into a buffer and played out as needed. By holding the packets just long enough to accommodate the longest delay in packet delivery, the added delay can be devastating. This also requires that the VoIP equipment have sufficient buffers to accommodate these interpacket delays on an ongoing basis. The bigger the buffers, the longer the potential delays.

Another major problem that one must contend with is the acceptable levels of packet loss. Several studies have been conducted indicating that acceptable packet loss is less than 10 percent. Anything greater than 10 percent packet loss will be intolerable. The network can drop packets when peak loads are being experienced, causing some degree of loss. Moreover, packets can be discarded in case of buffer overrun at the routers and switches across the network. These combined problems may move the packet loss levels to an unacceptable level.

If we combine the various levels of processing delay and the risk of packet loss, we can see where the networks must accommodate the delay, along with the need for VoIP equipment to address the issue. Figure 30-7 is a representation of the delays that can be cumulatively added to a packet delivery across the network.

Figure 30-7 Delays across a VoIP network

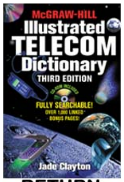




Protocol Stack

The VoIP protocol stack uses the benefits of a TCP/IP protocol suite in which to run over the network. Figure 30-8 is a representation of the protocols as they stack on top of each other. The primary areas are as follows:

- In Layer 1, we typically use SONET and DWDM for the physical architecture. Although other variations can be used, this is the more common today.
- At Layer 2, we typically will use ATM or Frame Relay, but the PPP protocol can also be used if some other form of link architecture is employed
- Layer 3 is where we will find the IP layer (Ipv4 or Ipv6), using the network layer to handle the datagram protocols
- At Layer 4, we use UDP instead of TCP for the real-time applications. Let's face it, UDP serves as a better set of protocols to use when we can ignore packet loss, or the reliability issues are addressed at a lower level.
- At the upper layers (5–7), we see the use of the *Real-Time Protocol* (RTP), the *Real-Time Control Protocol* (RTCP), and the *Resource*



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Reservation Protocols (RSVP). Others will include the *Network Time Stamp Protocol* (NTP) and the *Session Initiation Protocol* (SIP) as others.

- At the application, we see the H.323 protocols sitting on top of the heap.

Figure 30-8
The VoIP protocols stacked up

H.323	Layer 7 Application
RTP/ RTCP/ RSVP/ SIP/ NTP	Layer 5-7
UDP	Layer 4
IPv4 or IPv6	Layer 3
AAL	Layer 2
ATM or Frame Relay	
SONET	Layer 1
WDM or DWDM	