CHAPTER 1

Introduction and Motivation

1.1 Introduction

Data networks have progressed to the point that it is now possible to support voice and multimedia applications right over the corporate enterprise network and the intranet, for both on-net and off-net applications. Many companies have already deployed IP-based backbones that provide both broadband capabilities and Quality of Service (QoS)–enabled communications. Some companies have deployed Asynchronous Transfer Mode (ATM) networks. Switching technology, particularly in terms of the switched local area network (LAN), has gone a long way in the past five years, providing higher-capacity, lower-contention services across the enterprise campus network. High-speed wide area technology, such as Packet over Synchronous (POS) Optical Network and metro optical services as metro gigabit Ethernet, provide increased bandwidth across the enterprise regional, nationwide, and international networks.

The Internet Engineering Task Force (IETF) Multiprotocol Label Switching (MPLS) specification also directly or indirectly provides improved support of IP services. In addition, QoS-supporting protocols, such as IPv6, Resource, Integrated Services Architecture (*intserv*), differentiated services (*diffserv*) in IPv4, and Real-time Transport Protocol (RTP), are now entering the corporate enterprise network (Reference [1] provides a treatment of the trends listed here). A lot of industry effort has gone into supporting IP over ATM using a number of technologies, such as Classical IP over ATM (CIOA). All of this opens the door for the possibility of carrying voice over the enterprise network. Interest exists in the carrier arena as to the possibility of modernizing the existing public-switched telephone network (PSTN) with an IP-based infrastructure that would support multiple services, including Voice over IP (VOIP) (see Figure 1.1).

At the same time, commercialized Internet use has increased significantly in the past few years, as companies ventured into Web-based commerce [2]. During the

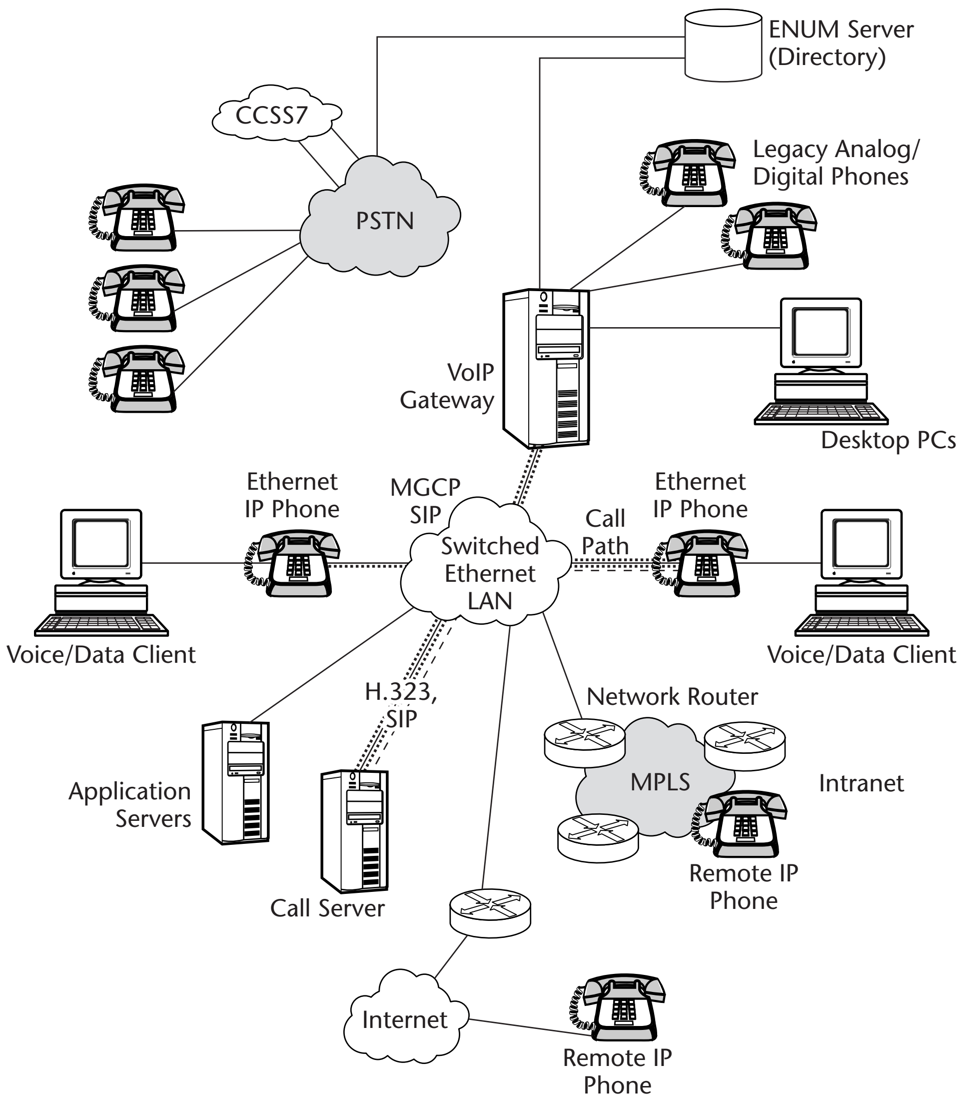


Figure 1.1
The VoIP environment.

late 1990s, the Internet was reportedly growing 100 percent per quarter. More recently, people have quoted 80 percent growth (or perhaps slightly less) per year. That large collection of backbones, access subnetworks, server farms, and hypertext information that is known as the Internet is acquiring ever-increasing importance, not only for the business community but also for the population at large. Access to information is proving increasingly valuable for education, collaborative work, scientific research, commerce, and entertainment. The advent of HTML-formatted, URL-addressable, and HTTP-obtainable information over the Internet—what is often called the World Wide Web (WWW or W3)—has generated a lot of attention in the past 10 years. Now there is a movement afoot to make the transition to fully multimedia enabled sites that allow voice, video, data, and graphics to be accessed anywhere in the world. The issue so far, however, has been that voice and video, by and large, have been of the stored kind—namely, a one-way download of sound files that are played out in non–real time at the user's PC.

Given this extensive deployment of data networking resources, the question naturally presents itself, is it possible to use the investment already made to carry real-time voice in addition to the data? The desire to build one integrated network goes back to the 1970s, if not even earlier. The Advanced Research Projects Agency, with project DACH-15-75-C0135 (and many other projects with many other researchers), funded the senior author's work in 1975 to look at the feasibility of *integrated voice and data packet networks.* And Integrated Services Digital Network (ISDN) research started in Japan in the early 1970s (before the idea started to get some real attention in the late 1970s and early 1980s) with the explicit goal of developing and deploying integrated networks. However, a lot of the mainstream work has been in supporting voice and data over *circuit-switched* time-division multiplexed (TDM) networks. Only some early packet over data work, and then some Fiber Distributed Data Interface II (FDDI II) and Integrated Voice/Data LAN (IEEE 802.9) work, looked at voice support in a non-circuit-mode network. Even for ATM, the emphasis has been, until the past few years, on data services.

The idea of carrying voice over data networks has received considerable commercial attention in the past five years. The ATM Forum, the Frame Relay Forum, and the MPLS Forum have published specifications, and a whole range of voice over data network has appeared and/or is appearing. The work of the ATM Forum and the Frame Relay Forum has focused on connection-oriented networks. However, connectionless IP-based networks are ubiquitous, and so there is a desire to carry business-quality voice over them. The major challenge in this regard is that IP networks do not yet support QoS features. Nonetheless, a plethora of *IP phones* and IP-to-public-network *gateways* has entered the market.

This book is one of two related Wiley books published by the authors. This book focuses on the IP telephony technology itself. Figure 1.2 depicts the various voice over data network technologies now evolving, including VOIP. Also note that IP can utilize a number of data link layer services, such as ATM, MPLS, and frame relay. Figure 1.3 depicts a possible scenario of VOIP, as is addressed in this book.

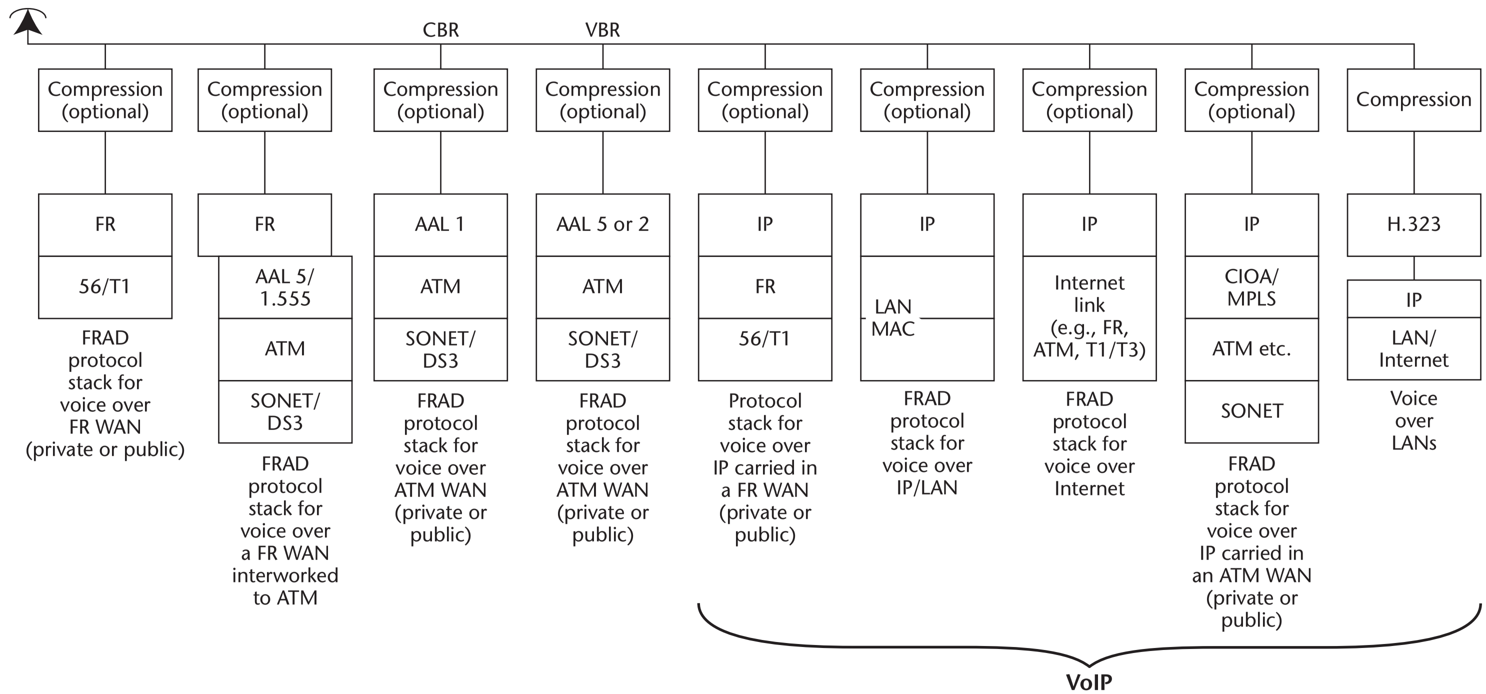


Figure 1.2
Various packet technologies usable for Voice over Packet.

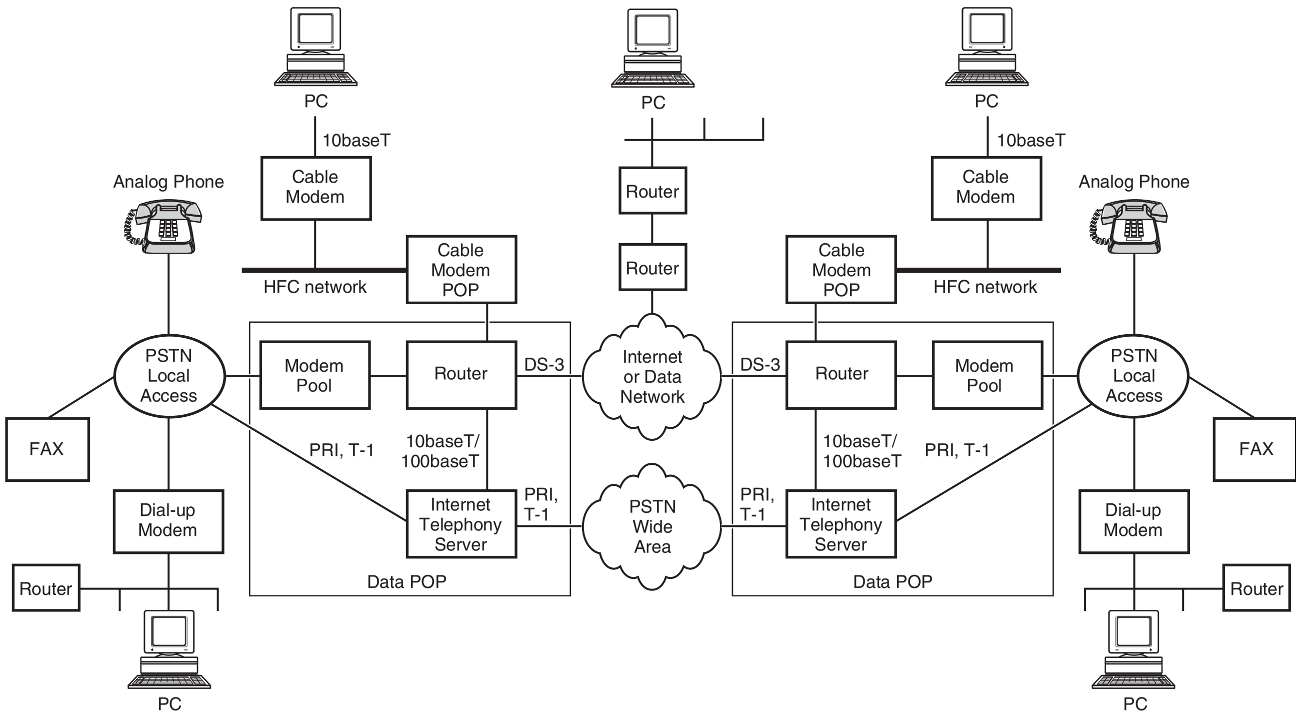


Figure 1.3
Internet telephony.

After this introduction in Chapter 1, a basic review of IP technologies is provided in Chapter 2, which covers IP, IPv6, RSVP, RTP, and MPLS. Chapter 3 discusses voice characteristics that can be utilized in packet networks. Chapter 4 discusses adaptive differential pulse code modulation (ADPCM) as applied to packet network environments. Chapter 5 provides an overview of vocoder-based compression methods used in IP. Chapter 6 covers various proposals for delivery of voice in IP environments. Chapter 7 covers the important topic of signaling. Chapter 8 provides a major review of QoS technologies. Chapter 9 covers voice over MPLS. Chapter 10 addresses directory services. Chapter 11 looks at opportunities for traditional carriers. Finally, Chapter 12 briefly looks at wireless opportunities.

1.2 Drivers for Voice over IP

This section discusses a number of drivers for voice over IP.

The Positive Drivers

Besides the potential for savings on long-distance phone charges to communicate with friends or relatives, Internet phones already have a place in the business world. For example, one can leave Internet phones turned on and ready for calls throughout the day; the technology is useful for communicating with coworkers in other parts of the building and at other locations by simply dialing them up on the Internet videophone. If they are at their desks, they can answer immediately. It can be a fine way to ask work-related questions without taking one's hands off the keyboard [3]. The technology is good for telecommuters, who can dial in to the office and see and speak to coworkers while getting a glimpse of the office from home [3, 4]. Similarly, it can be good for distance learning applications [5]. There are both market and business drivers for the introduction of voice telephony over IP at this time.

There have been four main stages of VOIP evolution in the past few years [6]:

1. PC-to-PC (since 1994)
   ⇒ Connects multimedia PC users, simultaneously online
   ⇒ Cheap, good for chat, but inconvenient and low quality
2. PC-to-phone (since 1996)
   ⇒ PC users make domestic and international calls via gateway
   ⇒ Increasingly services are "free" (e.g., Dialpad.com)
3. Phone-to-phone (since 1997)
   ⇒ Accounting rate bypass
   ⇒ Low-cost market entry (e.g., using calling cards)

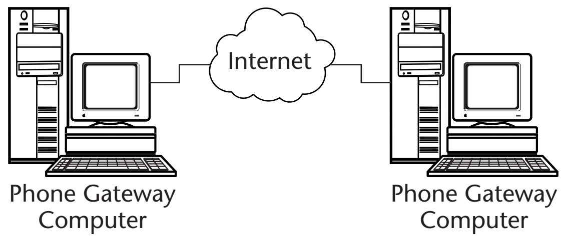
4. Voice/Web integration (since 1998)
   ⇒ Calls to Web site/call centers and freephone numbers
   ⇒ Enhanced voice services (e.g., integrated messaging)

Deregulation in the United States and elsewhere could mean that both incumbent carriers and new carriers can enter the market with new services. At various times in the past twenty years, a variety of carriers in the United States were precluded from entering certain telecommunication service sectors. One of the goals of the Telecommunications Act of 1996 was to change that. However, there has recently been a major slowdown in the competitive carrier landscape. This slowdown will be a major drag on the introduction of VOIP, since the new carriers were the principal beneficiaries of a less expensive technology that could be deployed in greenfield environments.

The technology to carry voice over data networks is evolving, as noted in the introduction. There are economic advantages to end users in utilizing an integrated network, not only in terms of direct transmission costs, but also in reducing the network management costs of running separate and technologically different networks. That is the ultimate goal. In the meantime, many companies are, and will be for some time in the future, supporting the infrastructure and cost of multiple networks, including PSTN, private enterprise networks, wireless networks, intranets, business video networks, Internet access networks, and Internet-based Virtual Private Networks. Hence, the need to optimize the usage of all media components on all networks simultaneously, and to take advantage of pricing alternatives between networks, will become even more important as these networks proliferate in the corporate environment, and as the service providers offer increasingly competitive prices [7].

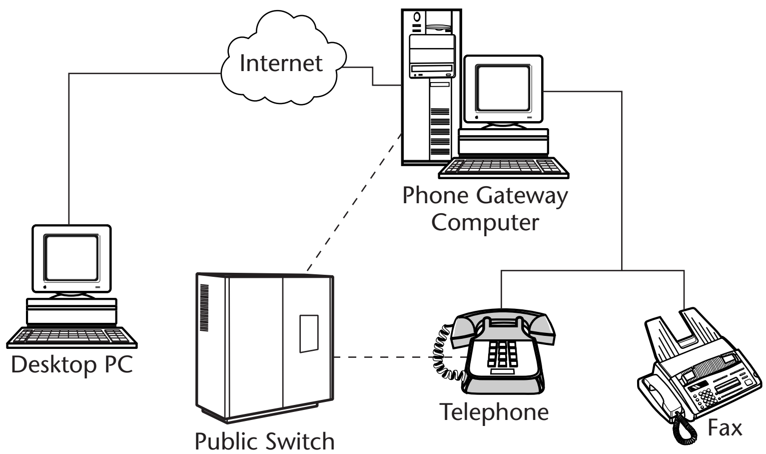
Separate from technology considerations, business drivers must come into play. Carriers need to make a positive bottom line (e.g., a 15 to 25 percent net bottom line) and be sustainable and self-sufficient. There are new revenue opportunities for Internet service providers (ISPs) in bundling voice service with Internet access. The interexchange carriers (IXCs) can avoid access charges. The local exchange carriers (LECs) can undercut the long-distance prices and offer Inter-LATA services without necessarily having to follow the traditional approach. Cable TV operators can bundle packet voice with cable services and perhaps find a better way to enter the telephony business without having to follow the classical *time-slot-interchange* method. Wireless companies can make more efficient use of the precious radio spectrum. Figures 1.4 to 1.9 depict typical carrier applications, based on reference [6]. All of these stakeholders can benefit by adding value to the network instead of just growing linearly to simply reach more physical points, and they can benefit by optimizing the economics of both packet-switched and circuit-switched networks. However, the major breakthrough has to come in the form of new services. Simply replacing a circuit-based transport mode with a packet-based transport mode will not justify the replacement of the old network with the new.

Figure 1.4
PC-to-PC, over IP.



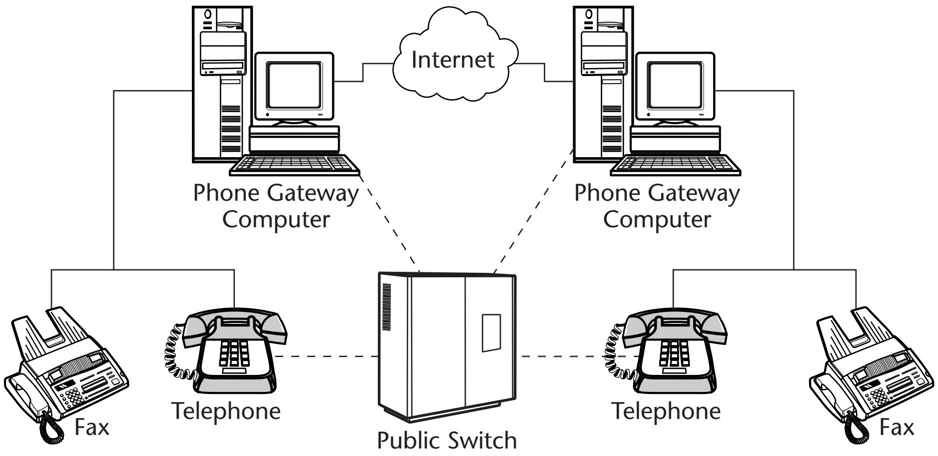
- Needs similarly equipped Internet users (e.g., IP telephony software, multimedia PC etc), both logged-on simultaneously
- Main applications: avoidance of usage-based telephone charges, chat-rooms, company LANs
- Application providers include Firetalk, Phonefree
- Potential Market: <50 million users?

In particular, the past few years have seen the emergence of reduced bit-rate voice compression algorithms that can increase the carrying capacity of a network by nearly tenfold (that is, by an order of magnitude) without the investment of additional resources in long-haul transmission facilities. The deployment, for example, of a network supporting near-toll-quality voice at 5.3 kbps rather than the twenty-five-year-old method of 64-kbps-per-call pulse code modulation (PCM) is not likely to be feasible in the context of an existing public switched telephone network because of the extensive embedded base of legacy equipment. Hence, if



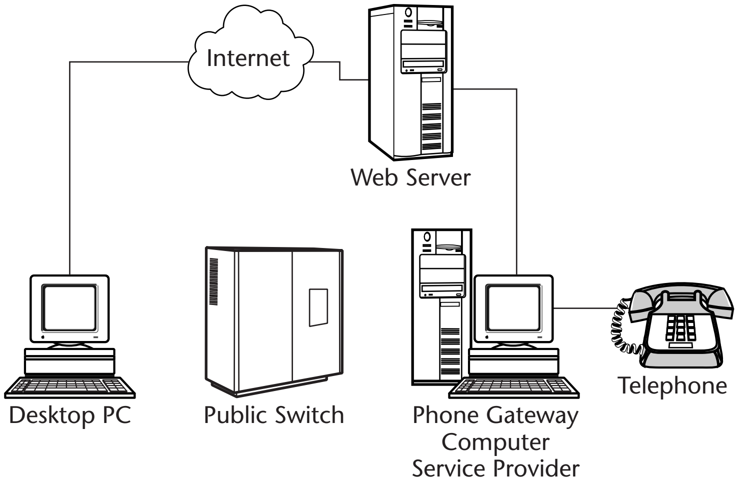
- Internet users with multimedia PC able to call any phone or fax user (not, at present, *vice versa*)
- Main motivation: Reduced telephone charges, "free" calls to U.S., Korea, Hongkong SAR etc.
- Service providers include Net2Phone, DialPad etc.
- Market potential: Sending > 250 million Web users, receiving >1.3 billion telephone/mobile users

Figure 1.5
PC-to-phone (or fax), over IP.



- Any phone/fax/mobilephone user to any other
- Main motivation: Reduced call charges, accounting rate bypass, market entry for non-facilities-based carriers (e.g., via pre-paid cards)
- Service providers include speak4free, I-link etc.
- Market potential: >1.4 billion phone/fax/mobiles

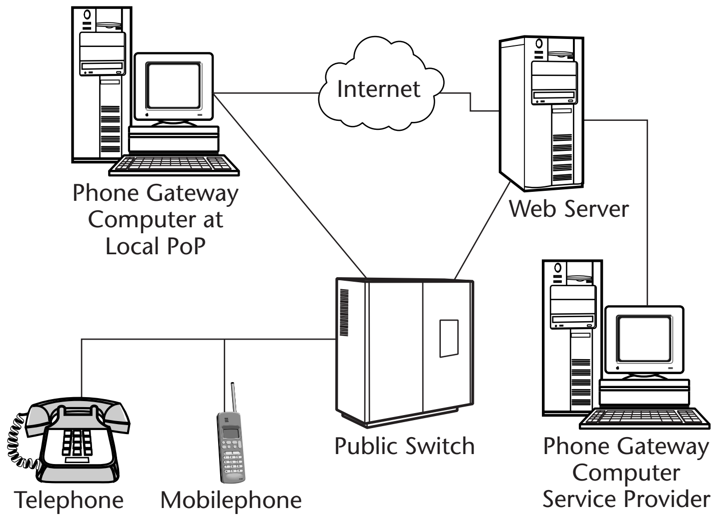
Figure 1.6
Phone/mobile-to-phone/mobile (fax-to-fax), over IP.



- Internet users with multimedia PC browse Website and choose voice/video connection option
- Main motivation: Service provider can interact directly with potential clients, via voice or video, for instance for telemarketing, freephone access
- Service providers include NetCall, ITXC etc.
- Market potential: >300 million Internet users

Figure 1.7
PC-to-Website/call center, over IP.

Figure 1.8
Phone/mobile-to-Website/e-mail, over IP.



- Phone or mobilephone users utilize enhance services (e.g., integrated messaging, voice response) available from IP service provider
- Main motivation: Integrated messaging, computer telephony integration, m-commerce
- Market potential: >1.4 billion phone/mobile users
- Service providers include Yac.com, T2mail etc.

there is a desire to use the new compression algorithms and achieve a tenfold efficiency gain, then the IP route may be the way to go.

Voice over IP can be deployed in private enterprise networks, but some technology suppliers are concentrating on providing new solutions for carriers, consistent with the approach just outlined. Applications of the evolving VOIP technology include the following:

- Internet voice telephony
- Intranet and enterprise network voice telephony
- Internet fax service
- Internet videoconferencing
- Multimedia Internet collaboration
- Internet call centers
- PBX interconnection

It is worth noting that there has been considerable progress recently in developing standards (with supporting equipment to follow) in the area of LAN/intranet-based multimedia (with compressed speech), as shown in Figure

1.9. These efforts will likely become the underpinnings of standards-based approaches to VOIP.

Recent analysis from the firm Frost & Sullivan, published in the 2001 report *U.S. Market for Enhanced IP-Based Voice Services*, reveals that the VOIP industry generated a revenue of $520 million in 2001 and that the market is projected to reach $31.8 billion by 2007 [8]. There has been some penetration in intranet environments. For example, Cisco reportedly expects to deploy VOIP service at over 41 sites, connecting the 35,000 Cisco IP phones already in place, to provide enhanced internal communications [9].

A few years ago, VOIP was the domain of just a handful of early pioneering companies, including 3Com, Cisco, Clarent, Nuera Communications, and Hypercom. Now, however, this convergence technology has finally been embraced by the more traditional networking and telecommunications vendors who had previously viewed VOIP as a serious threat to their installed bases. The early VOIP pioneering companies have been joined by the classic PBX vendors—Alcatel, Avaya, Ericsson,

Network	*N-ISDN*	*PSTN*	*Iso-Ethernet IEEE802.9*	*Packet-switched*	*B-ISDN (ATM)*	*B-ISDN (ATM)*
Multimedia standard	H.320	H.324	H.322	H.323	H.321	H.310
Audio/voice	G.711 (M) G.722 G.728	G.723.1 (M) G.729	G.711 (M) G.722 G.728	G.711 (M) G.722 G.728 G.723.1 G.729	G.711 (M) G.722 G.728	MPEG1 (M) G.711 (M) G.722 G.728
Audio rates, Mbps	64 48–64 16	5.3–6.3 8	64 48–64 16	64 48–64 16 5.3–6.3 8	64 48–64 16	$n \times 64$ 64 48–64 16
Video	H.261 (M)	H.261 (M) H.263 (M)	H.261 (M)	H.261 (M) H.263	H.261 (M)	H.262 (M) (MPEG-2) H.261 (M)
Data*	T.120	T.120	T.120	T.120	T.120	T.120
Multiplex	H.221 (M)	H.223 (M)	H.221 (M)	H.225.0 (M)	H.221 (M)	H.222.0 (M) H.222.1 (M)
Control	H.242 (M)	H.245 (M)	H.242 (M)	H.245 (M)	H.242 (M)	H.245 (M)
Signaling	Q.931	—	Q.931	H.225.0 (Q.931)	Q.931	Q.2931

(M) = Mandatory
*For example, Whiteboarding application

Figure 1.9
Evolution of voice over data networks via multimedia applications.

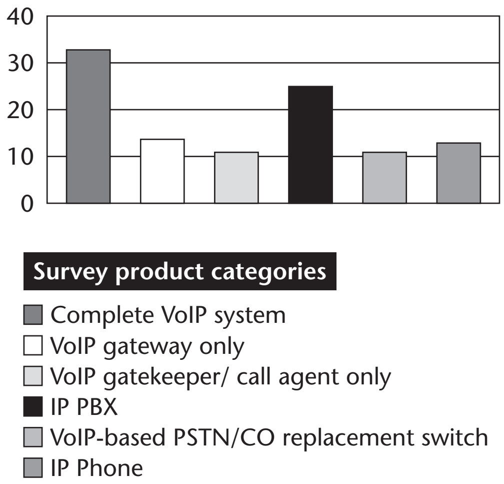


Figure 1.10 *VOIP vendor breakdown. (Courtesy of NetworkWorld, 1/29/01.)*

Mitel, NEC, Nortel Networks, and Siemens. By 2000, all of the vendors had introduced viable VOIP products—often in the form of add-ons, which “IP-enabled” the latest versions of the vendors’ TDMs and switching matrix-based PBXs [10]. There were 175 VOIP vendors in 2000; a segment-by-segment breakdown of the key players is shown in Figure 1.10.

Interoperability among VOIP products has been a major stumbling block to widespread acceptance of the technology. The International Telecommunications Union—Telecommunications (ITU-T) H.323 “umbrella” standard, the first posed for VOIP interoperability, proved complex to implement. As a result, in its place were proposed other less complicated standards, from which a number of loosely related standards—for example, the ITU-T H.323 and H.248/MEGACO, the IETF Session Initiation Protocol (SIP), the Media Gateway Control Protocol (MGCP), and the International Softswitch Consortium (ISC) specifications—emerged. The expectation is that no single standard will be predominant over the next couple of years; interoperability and coexistence will therefore be important [10].

Naturally, there are going to be challenges in deploying IP-based voice services. Table 1.1 depicts some of these challenges and some potential ways around them.

The Negative Drivers

Voice over packet in general and voice over IP in particular have seen tremendous trade press, consultant, vendor, and conference hype in the past five years. Considering the amount of air time given to the topic, the amount of actual deployment to date is underwhelming. What has held back the deployment of the technology includes the following:

Table 1.1 Challenges and Quality Issues

Problem	*Possible solution*
PC too overloaded to run vocoder; processing delays too long	Use standard terminals and PBXs, and use a high-power Internet voice telephony processor.
Congested data networks	Use compression, in conjunction with echo cancellation and packet recovery technology. Move to POS gigabit and/or Ethernet-based backbones and switched LAN segments.
Protocol limitations	Look to deploy IPv6, RSVP, and MPLS.
Poor end-to-end network service	Upgrade to broadband, use switching, and use premium Internet services.
Limited routing and directory capabilities	Directory services are becoming available, for example, IETF ENUM.

- An underestimation of the importance, complexity, and purpose of signaling and the need to interconnect with the 1.4 billion telephone sets already deployed globally.
- The confusion brought about by the multiplicity of signaling protocols that have been advanced.
- A lack of understanding of the economic value of the embedded base of equipment in carrier networks that likely will continue to operate flawlessly for what it was intended to do (make money with net bottom line of around 20 percent) for a decade or more.
- A lack of understanding of where the true costs to deliver telephone services are.
- A lack of understanding of the commoditization of both switched voice and high-speed bandwidth, obliterating the value from any savings related to the "bandwidth efficiency" of VOIP.
- A lack of understanding that any bandwidth savings in VOIP has nothing to do with IP but everything to do with vocoding algorithms that can well be supported without IP.
- The difficulty in securing QoS in public (and even private) IP networks.
- The most critical failure of all: a misunderstanding of where the value of VOIP has to be to make it successful. The bright spot for VOIP is that it brings forth a panoply of *new* services not possible with circuit-switched voice, not that it acts as a transport mechanism to replace existing trunks and/or Class 5 switches.

Why would anyone scrap what has over the years been called the best telephone system in the world just to replace it with something that does just the same and nothing more? A VOIP network cannot be just a "me too" network: "me too" carries voice.

Developers have to start focusing in earnest on bringing out new applications, not the chronic litany of "bandwidth efficiency" advantages. There is a reported glut of fiber. The Dense Wavelength Division Multiplexing (DWDM) can deliver terabits per second on fibers. The alleged need to save bandwidth is suspect and anachronistic; it is a dead end for VOIP.

Telecom professionals have to follow the lead of the PC developers: For well over a decade now, software developers have stopped trying to be highly parsimonious in RAM/disk drive efficiency so that an entire suite of new services of convenience on the desktop could be developed. Many new applications and capabilities have been developed on the desktop, including program linkage (audio and video) and easy-to-use browsers.

The story of VOIP has to be *less* on IP and the many new protocols that one can invent overnight, and *much more* on new applications, possibilities, services, interactions, voice capture, storage, manipulation, dissemination, interplay, and so on. That is why the first edition of this book had a chapter that, at face value, did not seem to fit in with the sophistication and intricacies of all those 300-page Internet protocols out there; it was a chapter on applications, or more specifically, on self-serve customer service Web-site applications. The message conveyed for VOIP to be successful: It is the applications, not the protocols. The same is true with audiophiles: It is the musical composition/content coming off the speaker that makes all the difference, not that one speaker rolls off at 44 Hz and the other at 43 Hz.

Voice over IP is not going to replace existing carrier networks in North America. Also, with PSTN-based voice becoming so cheap, it may not be worth the trouble to place voice on the intranet, except possibly for international applications. The future of VOIP must reside with new applications that it can and should bring forth.

1.3 Approaches for IP-Based Voice Systems

There have basically been two approaches to deploying VOIP in networks:

1. *Desktop approach*. Each individual in the organization purchases VOIP-enabled terminals, which can be used to support remote communications. This is in the vein of the Computer Telephony Integration (CTI) approach and the ITU-T H.323 terminal (as implied by the capabilities listed in Figure 1.10). In effect, this is similar to each user having a personal computer (PC)–based modem, a desk-resident fax machine, or a dedicated printer, rather than having a shared, network-resident server to support these functions.

2. *Shared approach*. The VOIP capabilities are developed in an industrial-strength mode, using shared, network-resident servers (as implied by Figure 1.2).

The earlier approach is usually the first approach to market, but there are advantages in migrating up to the network-resident model. Figure 1.11 depicts what a VOIP server could look like.

As noted, voice compression is going to be a key enabling technology for new IP-based voice services. Digital Signal Processing (DSP) has progressed to the point where (in supporting 10 to 20 million instructions per second [MIPS]) good-quality voice is achieved. Recently adopted ITU-T standards, such as G.723.1, G.729, and G.729A, are discussed in some depth in the chapters that follow. In the past, voice analysis and synthesis using what is called *vocoder* technology produced a robotic-sounding voice, but this changed dramatically during the 1990s. A lot of work has also gone into subjective testing of the voice to determine how good the proposed algorithms are. The most frequently used test in ITU-T SG12 is the *absolute category rating* (ACR) test. Subjects listen to about 8 to 10 s of speech material and are asked to rate the quality of what they heard. Usually a five-point scale is used to represent the quality ratings, such as 5 = excellent to 1 = bad. By assigning the corresponding numerical values to each rating, a *mean opinion score* (MOS) can be computed for each coder by averaging these scores. Typically, a test must include a selection of material (e.g., male and female utterances) [11]. Vocoder technologies are covered in Chapters 4 and 5.

Voice Servers Approach

A number of vendors have announced Internet Telephony Server (ITS) gateway technologies. These servers can be installed on PSTNs to route telephone calls over data networks such as the Internet. This kind of technology is being positioned to the carriers as a means to offer cost-effective alternatives to traditional long-distance calling or to enhance their data offerings by adding voice service, therefore creating a new revenue-generating or revenue-protecting opportunity. In addition to voice or fax services over the Internet, future applications of the ITS include phone-to-computer, computer-to-phone, and computer-to-computer connections. It will also enable audioconferencing, messaging, videoconferencing, call center operations, and media collaboration over the Internet [12].

A typical ITS server functions at a user or system level, like a PBX Tie Line. User selection of the IP network for voice or fax calling can be automated using PBX Least Cost Routing algorithms or can be dial-selected by users (e.g., by dialing 8 to access the IP/Internet). The use of the IP network could even be mandated through customer programming of PBX networking features. For example, customers could choose to have fax machines use only the Internet for intra-company

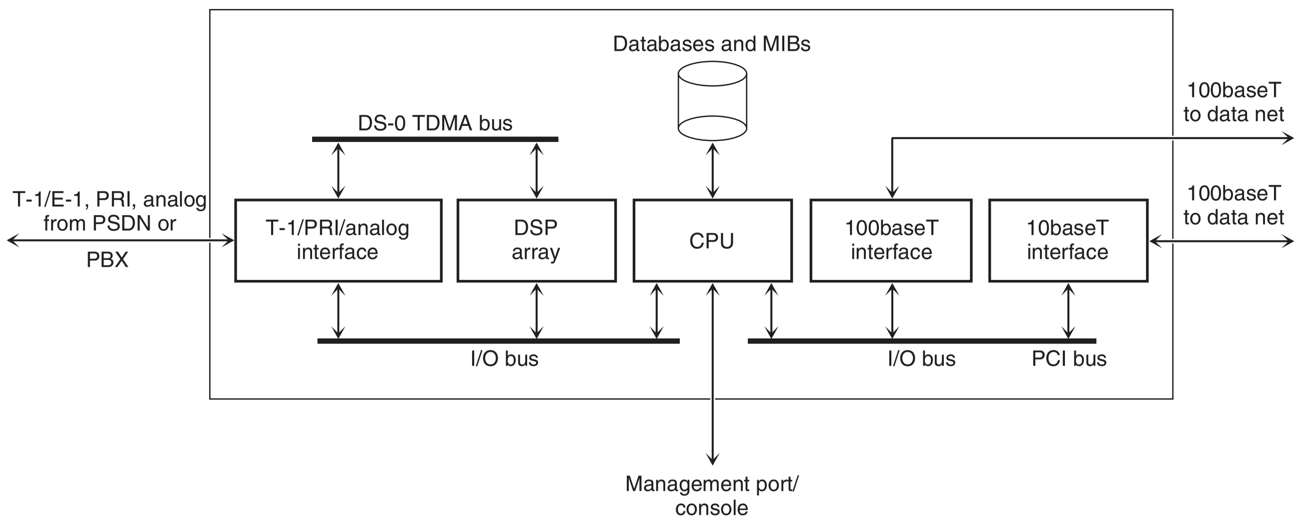


Figure 1.11
Typical voice over IP server.

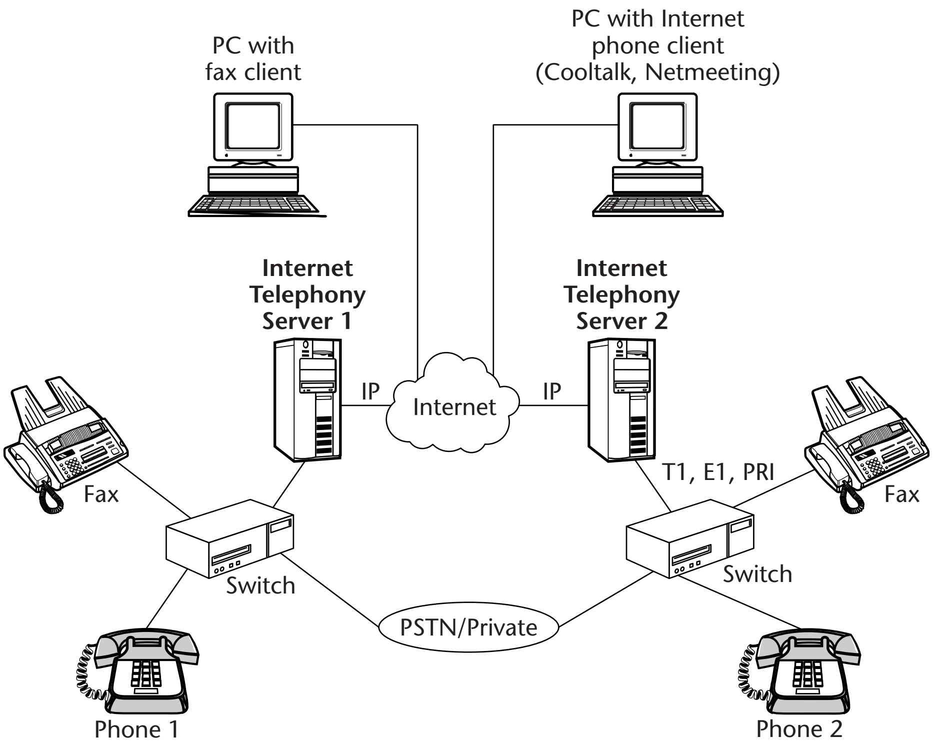


Figure 1.12
Typical Internet Technology Server (ITS).

correspondence, while allowing voice calls to use a combination of private or PSTN and intranet facilities. External fax correspondence could also use the intranet or Internet by using the tail-end hop-off networking feature of the PBX (or ISP) to escape from the data network at the closest point of presence to the terminating fax number. Also, remote users could access the ITS for voice and fax calls over the Internet from PC-net phones (IP phones), allowing a single remote connection to be used for data, e-mail, voice mail, fax, and real-time voice calls. Such remote users might also receive all of the benefits of their host PBX or ISP service through remote access over the intranet or Internet [7].

> [T]o make a call, one of the users will simply pick up the phone as usual and dial the phone number of the second user. The PBX will route the call to its Internet Telephony Server (treating it as just another PBX Tie Line) [see Figure 1.12]. The ITS will then make a "packet call" over the Internet to the distant ITS that will place the call through its associated switch. After the connection between the parties has been established, the rest of the phone

operation will continue as if the call was going over the regular PSTN. Up to 24 sessions can take place simultaneously, each including either a single voice or fax call, per each T1 card in the ITS.

An ITS may employ a dynamic and transparent routing algorithm for its operation (e.g., no routing decisions on the part of the customers are needed). In addition, if the quality of the "IP" network falls below pre-specified threshold, backup to PSTN is automatic. [7]

IP Voice and Video Phones

Internet phones and Internet videophones are entering the market, making it possible to talk to and see one or more remote parties over the Internet. Proponents make the case that PCs have always been the perfect vehicle for communication. Now, in conjunction with an ISP, one can use the PC to call anyone on the Internet (with the appropriate hardware and software) anywhere, anytime. For the time being, it does not cost any more to make these calls than the monthly ISP charges of $19.95, $14.95, or even $9.95. Some voice over Internet providers do not charge monthly access fees, but charge only on a per-minute basis.

There is a gamut of products on the market, including the following:

1. Audio-only Internet phones
2. Videoconferencing systems
3. Internet videophones with audio and video transmission capabilities
4. Server-based products or higher-end videoconferencing products that cost between $5,000 and $10,000

1.4 The Future

It is expected that IP-based telephony is going to see continued penetration in corporate intranets, extranets, and the Internet in the next few years. This book, in conjunction with the companion Wiley texts [1, 13], should give planners enough information to begin to assess and evaluate the value of this evolving technology for their own environments.

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