IMPLEMENTING VOICE OVER IP
This book is dedicated to:

Srijesa, Inrava, and Ashmita;
My parents, sisters, and brothers;
My teachers, present and past
Colleagues, and friends; and
All of those who consciously and honestly contributed to making me what I am today.
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PREFACE

In general, voice transmission over the Internet protocol (IP), or VoIP, means transmission of real-time voice signals and associated call control information over an IP-based (public or private) network. The term IP telephony is commonly used to specify delivery of a superset of the advanced public switched telephone network (PSTN) services using IP phones and IP-based access, transport, and control networks. These networks can be either logically overlayed on the public Internet or connected to the Internet via one or more gateways or edge routers with appropriate service protection functions embedded in them. In this book, I use VoIP and IP telephony synonymously, most of the times.

This book grew out of my participation in many VoIP-related projects over the past several years. Some of the early projects were exploratory in nature; oscillators had to be used to generate certain tones or signals, and oscilloscopes were used to measure the dial-tone delay, call setup time, and voice transmission delay. However, as the technology matured, a handful of test and measurement devices became available. Consequently, we turned out to be better equipped to make more informed decisions regarding the computing and networking infrastructures that are required to implement the VoIP service. Many of the recent VoIP-related projects in the enterprise and public network industries involve specifying a VoIP service design or upgrading an existing VoIP service platform to satisfy the growth and/or additional feature requirements of the customers. These are living proof of the facts that all-distance voice transmission service providers (retailers and wholesalers) and enterprise network designers are seriously deploying or considering the deployment of VoIP services in their networks.
This book discusses various VoIP-related call control, signaling, and transmission technologies including architectures, devices, protocols, and service requirements. A testbed and the necessary test scripts to evaluate the VoIP service and the devices are also included. These provide the essential knowledge and tools required for successful implementation of the VoIP service in both service providers’ networks and enterprise networks. I have organized this information into nine chapters and three appendixes.

Chapter 1 provides some background and preliminary information on introducing the VoIP service for both residential and enterprise customers. I also discuss the evolution of the monolithic PSTN switching and networking infrastructures to more modular, distributed, and open-interface-based architectures. These help rapid rollout of value-added services very quickly and cost-effectively.

Chapter 2 reviews the emerging protocols, hardware, and related standards that can be used to implement the VoIP service. These include the bandwidth efficient voice coding algorithms, advanced packet queueing, routing, and quality of service delivery mechanisms, intelligent network design and dimensioning techniques, and others.

No service can be maintained and managed without proper signaling and control information, and VoIP is no exception. The problems become more challenging when one attempts to deliver real-time services over a routed packet-based network. Chapter 3 discusses the VoIP signaling and call control protocols designed to provide PSTN-like call setup, performance, and availability of services.

Next, I discuss the criteria for evaluating the VoIP service. In traditional PSTN networks, the greater the end-to-end delay, the more significant or audible becomes the return path and talker echo, resulting in unintelligible speech quality. Therefore, hardware-based echo cancellers have been developed and are commonly used in PSTN switches to improve voice quality. In packet networks, in addition to delay, packet loss and variation of delay (or delay jitter) are common impairments. These impairments cause degradation of voice quality and must be taken into consideration when designing an IP-based network for delivering the VoIP service. I discuss these and related issues in Chapter 4.

Various computing and networking elements of a recently developed VoIP testbed are considered in Chapter 5. This testbed has been used both to prototype and develop operational engineering rules to deliver high-quality VoIP service over an IP network.

Chapters 6, 7, and 8 focus on various possible VoIP deployment scenarios in enterprise networks, public networks, and global enterprises. Enterprise networks can utilize VoIP technology to offer voice communications services both within and between corporate sites, irrespective of whether these sites are within the national boundary or anywhere in the world. In the public networking arena, the VoIP service can be introduced in PSTN, cable TV, and wireless local loop–based networks for local, long-distance, and international calls.
Chapter 9 is the final chapter. In addition to presenting some concluding remarks and future research topics, I provide some guidelines for implementing the VoIP service in any operational IP network. These include the reference architectures, implementation agreements, and recommendations for network design and operations from a handful of telecom, datacom, and cable TV network/system standardization organizations.

Implementation of a few techniques that can be utilized to measure the call set performance and bulk call-handling performance of the VoIP network elements (e.g., IP-PSTN gateways, the VoIP call server) are presented in Appendices A and B. Appendix C illustrates experimental evaluation of the quality of transmission of voice signal and DTMF digits in both PSTN-like and IP networks with added packet delay, delay jitter, and packet loss scenarios.

In the Glossary of Acronyms and Terms, definitions and explanations of widely used VoIP terms and abbreviations are presented.

Finally, I hope that you will enjoy reading this book, and find its contents useful for your VoIP implementation projects. As the technologies mature or change, much of the information presented in this book will need to be updated. I look forward to your comments and suggestions so that I can incorporate them in the next edition of this book. In addition, I welcome your constructive criticisms and remarks. My e-mail addresses are b.khasnabish@ieee.org and bhumip@acm.org (www1.acm.org/~bumiph).

BHUMIP HASNABISH

Battle Green
Lexington, Massachusetts, USA
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My hat goes off to my children who inspired me to write this book. They naively interpreted the VoIP network elements as the legos during their visits with me to many of the VoIP Labs. This elucidation is more realistic when one considers the flexibility of the VoIP network elements to help rapid rollout of new and advanced services.

By posing the issues from many different viewpoints, my friends and colleagues from GTE Laboratories (now a part of Verizon) and Verizon Laboratories helped me understand many of the emerging VoIP related matters. Accordingly, my special thanks are due to—among others—Esi Arshadnia, Nabeel Cocker, Gary Crosbie, John DeLawder, Elliot Eichen, Ron Ferrazzani, Bill Goodman, Kathie Jarosinski, Naseem Khan, Alex Laparidis, Steve Leiden, Harry Mussman, Winston Pao, Edd Rauba, Gary Trotter, and George Yum. I have touched on several topics in this book, and many of them may need further investigations for network evolution.

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Bhumip Khasnabish

Barnstable Harbor
Cape Cod, Massachusetts, USA
Implementation of real-time telephone-quality voice transmission using the Internet protocol (IP, the Internet Engineering Task Force’s [IETF’s] request for comment [RFC] 2460 and RFC 791) is no longer as challenging a task as it was a few years ago [1,2]. In this introductory chapter, I define the instances and interfaces of both public switched telephone networks (PSTN) and corporate or enterprise communication networks where voice over IP (VoIP) can be implemented. The goals of VoIP implementation are to achieve (a) significant savings in network maintenance and operations costs and (b) rapid rollout of new services. The objective is to utilize open, flexible, and distributed implementation of PSTN-type services using IP-based signaling, routing, protocol, and interface technologies. To achieve this, it is necessary to change the mindset of those responsible for the design and operations of traditional voice services networks. Furthermore, one has to be ready to face the challenging problems of achieving reliability, availability, quality of service (QoS), and security up to the levels that are equivalent to those of the PSTN networks.

I discuss two paradigms for implementing the VoIP service in the next section, and then present a few scenarios in which VoIP-based telephone service can be achieved for both residential and enterprise customers. A functionally layered architecture is then presented that can be utilized to facilitate the separation of call control, media adaptation, and applications and feature hosts. Finally, I describe the organization of the rest of the book.

1 The ideas and viewpoints presented here belong solely to Bhumip Khasnabish, Massachusetts, USA.

2 300 to 3400 Hz (or 3.4 KHz) of analog speech signal.
THE PARADIGMS

The following two paradigms are most prevalent for implementation of the VoIP service:

- Server, router, and personal computer (PC)/plain old telephone service (POTS) phone-based (mostly) flat network and
- PSTN switch and mainframe computers, VoIP gateway (GW)\(^3\) and gatekeeper (GK),\(^4\) SS7 signaling gateway (SG),\(^5\) and the POTS-phone/PC-based (mostly) hierarchical network.

In order to provide VoIP and IP telephony services, PCs need to be equipped with a full-duplex audio or sound card, a modem or network interface card (NIC) such as an Ethernet\(^6\) card, a stereo speaker, a microphone, and a software package for telephone (keypad, display, feature buttons, etc.) emulation. Hardware-based IP phones can be used with a traditional PSTN network using special adapter cards—to convert the IP packets into appropriate TDM-formatted voice signals and call control messages—as well.

In the server-router-based networking paradigm, the servers are used for hosting telephony applications and services, and call routing is provided by traditional packet routing mechanisms. In the other case, the telephone features and services can still reside in the PSTN switch and/or the adjacent mainframe computer, and the packet-based network elements—for example, the VoIP GW, GK, and SG—can offer a sufficient amount of signaling, control, and transport mediation services. Call routing in this case follows mainly the traditional hierarchical call routing architecture commonly utilized in the PSTN networks.

The details of network evolution and service, network, control, and management architectures depend on the existing infrastructures and on technical, strategic, and budgetary constraints.

VoIP FOR RESIDENTIAL CUSTOMERS

In the traditional PSTN networks, the network elements and their interconnections are usually organized into five hierarchical layers [3] or tiers, as shown

\(^3\)VoIP GW translates time division multiplex (TDM) formatted voice signals into a real-time transport protocol (RTP) over a user datagram protocol (UDP) over IP packets.

\(^4\)The GK controls one or more GWs and can interwork with the billing and management system of the PSTN network.

\(^5\)The SG offers a mechanism for carrying SS7 signaling (mainly integrated services digital network [ISDN] user part [ISUP] and transaction capabilities application part [TCAP] messages over an IP network. IETF’s RFC 2960 defines the stream control transmission protocol (SCTP) to facilitate this.

\(^6\)Ethernet is the protocol of choice for local area networking (LAN). It has been standardized by the IEEE as its 802.3 protocol for media access control (MAC).
in Figure 1-1. The fifth layer contains end-office switches called CLASS-5 switches; examples are Lucent’s 5ESSS, Nortel’s DMS-100, and Siemens’ EWSD. These switches provide connectivity to the end users via POTS or a black phone over the local copper plant or loop. In the United States, the regional Bell operating companies (RBOCs) such Verizon, Bell-South, SBC, and Qwest provide traditional POTS service to the residential and business customers (or users) in different local access and transport areas (LATAs).

Implementation of VoIP for CLASS-5 switch replacement for intra-LATA communication would require a breakdown of the PSTN switching system in a fashion similar to breaking down the mainframe computing model into a PC-based computing model. Therefore, one needs to think in terms of distributed implementation of control of call, service, and information transmission. Services that are hosted in the mainframe computer or in the CLASS-5 switches could be gradually migrated to server-based platforms and could be made available to end users inexpensively over IP-based networks.

VoIP can be implemented for inter-LATA (CLASS-4) and long-distance (both national and international, CLASS-3, -2, and -1) transmission of the voice signal as well. Figure 1-2 shows an implementation of long-distance voice transmission using the IP network for domestic long-distance services, assuming that the same company is allowed to offer both local and long-distance services in the LATAs that are being interconnected by an IP network. Here the network access from the terminal device (e.g., a black phone) can still be provided by a traditional CLASS-5 switch, but the inter-LATA transmission of a voice signal is offered over an IP network. The resulting architecture demands VoIP GWs to convert the TDM-formatted voice signal into IP packets at the ingress and vice versa at the egress. The VoIP GK controls call authentication, billing, and routing on the basis of the called phone number (E.164 address) and the IP address of the terminating VoIP GK. This is a classical implementation of VoIP service using the International Telecommunications Union’s (ITU-T’s) H.323 [4] umbrella protocols. The same architecture can be utilized or extended for international VoIP services, except that now the call-originating and call-terminating VoIP GWs would be located in two countries. Different countries usually deploy different voice signal companding schemes, use different formatting of voice signal compression mechanisms, and prefer different kinds of coding of signaling messages [5]. Therefore, the details of this type of design need to be carefully considered on a case-by-case basis.

VoIP FOR ENTERPRISE CUSTOMERS

Some form of data communication network usually exists within any enterprise or corporation. These networks commonly utilize X.25, IP, frame relay (FR), and asynchronous transfer mode (ATM) technologies. However, recently, most of these networks have migrated to or are planning to use IP-based networks. Figure 1-3 shows such a network.
The five-layer hierarchy of a traditional PSTN consisting of CLASS-1 to 5 of central office (CO) switches. CLASS-5 COs are commonly referred to as end office (EO) switches and CLASS-4 COs as tandem switches. The private automatic branch exchanges (PBX) are known as CLASS-6 COs as well. PBXs are used to provide traditional and enhanced PSTN/telephony services to business customers.
For voice communications within the logical boundaries of an enterprise or corporation, VoIP can be implemented in buildings and on campuses both nationally and internationally. For small office home office (SOHO)-type services, multiple (e.g., two to four) derived phone lines with a moderately high (e.g., sub-T1 rate) speed would probably be sufficient. VoIP over the digital subscriber line (DSL; see, e.g., www.dsllife.com, 2001) channels or over coaxial cable can easily satisfy the technical and service requirements of the SOHOs. These open up new revenue opportunities for both telecom and cable TV service providers.

Most medium-sized and large enterprises have their own private branch exchanges (PBXs) for POTS/voice communication service, and hence they use sub-T1 or T1 rate physical connections to the telephone service providers’ networks. They also have T1 rate and/or digital subscriber line (DSL)-type connections to facilitate data communications over the Internet. This current mode...