Internet Multimedia Communications Using SIP

A Modern Approach Including Java® Practice

ROGELIO MARTÍNEZ PEREA

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Rogelio Martínez Perea

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Printed in the United States of America

To my parents, for their love and support

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Preface

Why This Book

In the late 1990s, I was engineering manager at the switching department in a mobile telecom operator. The mobile switches we dealt with were based on circuit-switched technology. They were big, complex, and proprietary pieces of hardware and software involved almost exclusively in the provision of voice service. By that time, ATM (Asynchronous Transfer Mode), a packet-switched technology that followed the virtual circuit approach, started to gain maturity as a suitable way for carrying media traffic with QoS requirements. Media transport was only part of the problem. The other part, signaling, did not have, by then, a mature candidate. The industry response was to strip the existing circuit switches off their switching matrix and provide them with the interfaces to control an external packet-based switching matrix in the so-called soft-switch approach. It was kind of throwing out the old-fashioned hardware but retaining the software. That was a pragmatic approach that the market took in order to rapidly respond to the operator's needs. However, it still took several years for the telecom operators worldwide to implement these architectures. By that time, we knew there was work in the IETF about a protocol called SIP, whose first version was published in 1995, but the main focus of the industry was on H.323 for enterprise networks and in the soft-switch approach for public telecom networks. In the meantime, Internet and the web were increasing their popularity, but this fact seemed, by then, unrelated to our challenge of evolving the network. Being intrigued about the possibility of using a packet-based network for media transport, that was the first time I built an IP soft-phone. I just developed a simple Windows program over the Win32 API on a standard PC. I made up a simple signaling protocol consisting of a bunch of messages and sent them over TCP/IP using the Winsock interface. Regarding the media, I just got the raw voice samples from the Wave API and put them directly on UDP packets that I sent over the network using Winsock. Surprisingly enough, it worked, and I could test it over a medium-sized LAN. I needed no voice network equipment (neither voice switch nor soft-switch), just a dumb IP network and a Windows program that I developed in a few weeks and ran on a cheap PC. The simplicity and the flexibility of the solution convinced me that voice technology as we knew it was meant to change sooner or later, and that the new technology would be one that advocated simplicity in the network and flexibility in the endpoints as well as cheap and off-the-self hardware and software.

In the next years, I changed roles and became manager for a team doing mobile services design and development. By that time, I had already built a new version of my softphone, only that then I used a beta version of a SIP stack, an Internet protocol that was destined to revolutionize multimedia communications both in the Internet and in the telecom environment. As will be explained in this book, SIP follows the flexible Internet approach that advocates moving intelligence to the endpoints and keeping the network as simple as possible.

During that time, I became convinced that understanding, even if it is at a high level, how SIP software works helps to understand its simplicity, flexibility, and potential. And that is the reason why, when years later I decided to write a book on a state-of-the-art approach for multimedia communications, I went for an approach that combined theory with practice. And the result is this book.

Approach

This book's aim is to let readers understand what Internet multimedia communications are and how they are enabled by using the Session Initiation Protocol and other related technologies. The approach I have taken in writing this book has three main characteristics.

First, it is *Internet-orientated*. That is, it is focused on the Internet technologies, protocols, and practices for delivering these services. In the last two chapters, it also touches upon how these Internet technologies can be used in controlled network scenarios such as those present in telecom operators' multimedia networks. In fact, the bodies involved in the standardization of telecom networks, such as ETSI or 3GPP, have adopted the ideas coming from the Internet in order to design the next generation of telecommunication networks.

Second, it follows a *fully modern and up-to-date* approach where the latest Internet developments are analyzed and discussed. In addition to providing a thorough explanation of the basic concepts, it also presents the most recent proposals for utilization of SIP and related technologies in the remit of multimedia communications. The book tackles new and innovative technologies and services such as MSRP, NAT traversal, STUN, ICE, session border controllers, TCP-based media transport, XCON conferencing framework, media server control, GRUUs, RPID, latest approaches for RTP security, XCAP, Text over IP, remote call control, floor control, conference control, Fax over IP, enhanced identity management, IMS, TISPAN next generation networks, voice call continuity, IMS centralized services, and so forth.

Following a modern approach implies that the book contains not just references to official standard or informative documents (e.g., Request For Comments), but also many references to the latest IETF Internet Drafts that represent current work in progress.

Third, the book is unique in its kind by the fact that it not only *contains theory but also practice*. The practical nature of the book is twofold. On one hand, the book tackles multimedia service creation, both at SIP level and at media level. It contains a comprehensive description of the state-of-the-art technologies for multimedia service creation. More than that, the book explains in detail how to program multimedia services using Java. Readers will learn how to programmatically use an open-source SIP stack and a popular Java API for media development. Many examples and Java practices are included in the book. Readers are guided step-by-step to build a simple yet functional soft-phone supporting voice, video, and messaging, plus a simple SIP proxy and registrar to be used with the soft-phone. The main purpose for the inclusion of code in the book is derived from my experience when dealing with multimedia technology: being able to take a look, even if you are not a Java programmer, at code that illustrates how services are done facilitates the comprehension of the technical concepts and the simplicity and potential in the technology. Another aspect of the practicality of the book refers to the fact that it also contains explanations of the situations where the different technical solutions may be used in real deployments.

Audience

The book is targeted at several types of audiences. In any case, all readers should have a technical background, an interest in technology, or a passion for Internet-related topics.

First, this book is targeted at the professional in the telecom or IT industry who needs to gain an understanding of the newest Internet Protocol–based technologies for delivering voice, video, messaging, and data services, and to acquire the skills and tools to successfully design and implement multimedia solutions in different environments (from small enterprise deployments up to Internet-wide deployments). IT architects will use the book to understand how their existing enterprise IP networks can be leveraged for delivering voice, video, and messaging, and what technologies the products that they choose must support. Telecom architects will use the book to gain an understanding of how SIP and other Internet technologies can be used to evolve their networks and offer innovative services (or offer existing services but with a reduced CAPEX and OPEX!). SIP related technologies play a key role in the movement into Fixed Mobile Convergence and Total Communication propositions that most telecom operators are embracing nowadays. IT and telecom engineers will find the necessary information in the book to understand how technology works, and will be referred to the appropriate technical documents for further detail. The book is also very useful for IT and telecom managers that want to understand how their business needs to be evolved toward an all-IP infrastructure and what are the benefits and challenges in doing so.

Second, this book is targeted at the academic community, where it can be used as base material for a one-semester theoretical course on Internet multimedia communications or as support material for practices in a laboratory course.

Third, software developers will find in the book the necessary theoretical and practical information that allows them to learn how to build basic SIP applications and sets the grounds for more-complex application design and development.

And last but not least, any person who has a technical background and has a passion for being informed about the hottest stuff around the Internet is also a potential candidate for enjoying the book.

Organization

The book is organized in three parts and 24 chapters.

The first part, "Fundamentals," comprises the first five chapters in the book. These give the necessary background information on Internet multimedia architecture, protocols, and service creation tools for understanding the rest of the book.

Above all, this first part explains the rationale behind the design of the multimedia protocols and the remit in which they are used. Setting the scope of the technology is crucial for using it successfully.

The second part, "Core Protocols," is the central part of the book, and is dedicated to explaining how the main protocols work in concert to deliver multimedia services. In order to enhance the comprehension of the theory, the reader is also guided into the elaboration of simple Java-based programming practices that allow him or her to better comprehend the theoretical concepts. As part of these practices, readers will learn to build, step-by-step, a simple yet functional soft-phone supporting voice, video, and messaging. Those readers who are not interested in the programming practice can simply skip the related chapters and just focus on the theory. However, I would recommend that even these readers take a quick look at some of the code snippets so that they can get a high-level understanding of how applications can be developed.

The third part, "Advanced Topics," deals with the latest and most innovative usages of the technology. Readers who already have professional experience with the technology, either designing or developing solutions, might want to skip the first two parts and dive directly into this part. In addition to tackling the most recent advances in the technology, Part III also shows how hot issues that every multimedia deployment faces are resolved. An example of that is the hot NAT traversal topic, of which a very thorough analysis is done and several possible solutions are detailed.

Additionally, the last two chapters in the book explain how Internet multimedia technology can be used in network scenarios where a tighter relationship with the service provider exists. A paradigmatic example of this concept is the 3GPP IMS, to which a long chapter is exclusively dedicated. The approach used in this book to present the IMS architecture and concepts is very different from the traditional one used by other books on the subject. Instead of first introducing an overwhelming architecture diagram full of unintelligible names and then explaining what the role is of the various components, a different approach is followed. It is based on leveraging the Internet concepts learned throughout the book, and explaining how they naturally evolve to support additional requirements that telecom operators may have, and that are not strictly relevant in a pure Internet environment.

Code Examples

This book does not intend to teach programming. The code examples are included just for the shake of illustrating how the protocols work. Readers can build simple examples where they can test the concepts learned. I have purposely omitted the bulk of error checking and recovery so as not to deviate the reader's attention from the functional concepts. I am convinced this has resulted in more comprehensive programs that show clearly how protocols operate. On the other hand, it means that programs are not fit for commercial use, and that they need to be fed with consistent data; otherwise, they will fail. Additionally, when I have thought that good OO practice made the functional concepts more difficult to understand, I have preferred to sacrifice perfect OO programming techniques.

Acknowledgments

I wish to thank Rogier Noldus, from Ericsson, and David Page, chief scientist at OpenCloud, for their impressive work in reviewing the manuscript. Their comments and suggestions have greatly contributed to the possible quality of the final product. Also, Nick Hudson, a former colleague at Vodafone, reviewed the manuscript and provided useful feedback. Santiago Borrero, a colleague of mine at Vodafone, reviewed the first introductory chapters and contributed to making the technical concepts in them easier to understand for readers with different backgrounds.

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My biggest thanks go for my wife, whose encouragement, support, patience, and understanding have been crucial for the successful accomplishment of this project.

About the Author

Rogelio Martínez has an M.S. in Telecommunications Engineering from Universidad Politécnica de Madrid, Spain. He has worked for the Vodafone Group for more than 12 years and has held various responsibilities there. Martínez was Switching Department Engineering Manager for 4 years. For the past 5 years, as Design Manager, he has led a team of technical specialists devoted to mobile applications design and implementation. More specifically, for the past 2 years, as a Senior Manager, he has led the design and evolution of the Vodafone Group multimedia service layer. At Vodafone, Martínez has been deeply involved in the deployment of SIP-based technology in operators all around the world.

Rogelio Martínez lives in Madrid with his wife and two children, and is very fond of playing tennis and skiing.

Foreword

Jorge Gató, Vodafone España

At the end of last century—to quote Thomas L. Friedman's excellent book, *The World Is Flat*—I was part of the unflat old world, specifically the old telephony world. I was reading (and listening to) the new flat world boys coming at the speed of light to re-do and improve things in months, weeks or even days that had previously taken us years to develop.

I was able to witness the initial days of the Voice over Packets (although, to be precise, voice was over packets when it became digital, years before), the initial trials and the early deployments of Voice over ATM and over IP. It was the time of the "Internet bubble" and a lot of fast innovation was happening, with many new small and smart start-up companies created, mainly in the USA and Scandinavia. It was a beautiful, creative time.

However, things were not so simple. The initial efforts to quickly replace the old telephony (SS7) world failed, and only the strongest companies survived. Once again, the technique of copying and using the best of both worlds (SS7 and IP), was used. SIP protocol was born (congratulations SIP!). It was, and still is, difficult to find people really skilled in both (SS7 and IP telephony) areas, and interdisciplinary teams were formed, with people bringing what they had, in many cases with high personal effort. I was lucky to be part of one of these teams in IETF (with a very modest contribution) and learned a lot from it.

Such technologies have evolved a lot and, still, there are not many people with complete knowledge of the SIP (and Internet Multimedia) technology, including all aspects: from theory, prototyping, and development, to implementation. Rogelio is one of the few people I know with such broad (covering theoretical and practical aspects) and deep knowledge, based on years of work in different managerial positions in the communications area (steering and inspiring key projects in different technology units).

I strongly believe the Multimedia Internet (mainly mobile and ubiquitous) is here to stay. It is starting to happen, and I honestly do not know where it will take us within the next five years, but I dream of a richer instant multimedia communication, making our lives more comfortable, allowing us more time to enjoy the company of our family and friends.

For such dream, I am sure that protocols like SIP are the way forward. But they are nothing without innovative, high quality applications adapted to our (customer) needs (and with a sustainable business model).

This book covers both areas needed to move into my dreams. It covers in depth SIP (and many IP related) protocols and networks and how to develop applications using its full potential. This is the reason I like Rogelio's book and I think it is an excellent guide to any engineer willing to plan, deploy or operate a SIP network and to any developer wishing to build efficient applications making use of the potential of a SIP network.

I am sure you will enjoy reading the book and I hope it helps you to contribute to enrich the Multimedia Mobile Internet world.

Rogier Noldus, Ericsson, Netherlands

When the Internet was developed in the 1970s of the previous century (long before my personal involvement with this technology!), it was targeting data services. Remote users could—in a convenient way according to the standards of that time—share electronic data files, establish simple message exchange sessions or establish machine-to-machine data communication sessions. Even so, the Internet had limited usage and was applied mainly in the academic world and by research institutes. The ARPANET, as the data connection network was known in those days, was gradually replaced by the NSFNET. The main transmission protocol used by NSFNET remained TCP/IP, inherited from the ARPANET. TCP and IP have undergone a number of iterations up to the current TCP v4, IP v4 and IP v6.

Along with the rapid growth of the number of Internet based applications, initially mainly person-to-content applications, emerged the concept of Internet based *communications*. Obviously, all Internet based applications constitute some form of communication. However, this new trend relates rather to person-to-person communication. One prominent example of this is Voice over IP (VOIP) between two Internet users. There are currently a large number of VOIP applications in operation on the public Internet. A current trend is to extend VOIP to include also multimedia, i.e. *Internet multimedia communications*, encompassing voice, video, text etc.

The Session initiation protocol (SIP) was developed by the Internet engineering task force (IETF) as the artery of Internet voice and multimedia communications. SIP is considered the successor of the H.323 protocol which was developed by the ITU-T for similar application.

The third generation partnership project (3GPP) has adopted SIP as the protocol for the IP multimedia system (IMS). This underscores the faith that the industry has in the long-term usability of SIP for multimedia communications. It also gives substance to the expectation that there will be widespread deployment of SIP-based communication for the foreseeable future. Thorough understanding of SIP is therefore quintessential for anyone involved in Internet based multimedia communication such as IMS. It must be emphasized here that *Internet based* *communication* encompasses the public Internet (e.g. peer-to-peer VOIP), enterprise networks (e.g. IP based office communication) and telecommunications networks (e.g. IMS). SIP and the accompanying media transport protocol RTP, have even found their way in the more traditional architectures like Wireline networks and mobile networks.

The book from Rogelio Martínez, *Internet Multimedia Communications Using SIP*, is an excellent source of information for anybody working in this field. During the period that Rogelio and I were closely involved in the development of architecture of an Internet based communication system, I came to appreciate Rogelio's wealth of knowledge in this field of technology. This book leaves no doubt about that! The book takes the reader through essentials of VOIP and IMS. It has an easy-to-follow step-by-step approach, starting with a brief history of the Internet. When reading chapter 1 of the book, one will almost feel part of the Internet development scene. The reader is then taken gradually from 'plain SIP' to advanced techniques. Brand new topics like Presence, IMS messaging and multimedia conferencing are covered. NAT Traversal, being an important issue when running SIP through border gateways, is extensively described in a separate chapter. Quality of service is traditionally a cornerstone of the telecommunications industry. Developers of Internet based communication systems will therefore gain ample advantage of the dedicated chapter on that topic. User identification and data security are essential to any communication system and are therefore covered in-depth as well. The book shows that there is continuing development in these areas. The reader is further enticed to put theory into practice. This is accomplished through the JAVA based SIP terminal that the reader is invited to build, using the example software code contained in the book. This combination of theory and practice makes the book unique in its class.

This book is an excellent contribution to the Internet communications industry. It not only provides a good explanation of the fundamentals of VOIP and IMS, but it also includes ample references to relevant standards for further reading. This book is therefore strongly recommended to anyone who needs to build up knowledge in this area of technology.

The book further strikes a bridge between the 'old technology' (GSM, Intelligent networks) and the 'new technology' (IMS, SIP). Having worked in the area of GSM and Intelligent networks for a substantial number of years, I appreciate the links that one can draw between well-known techniques and principles from GSM on the one hand and methods applied in the Internet communications on the other hand. Quite appropriately, the book closes with a dedicated chapter on IMS, placing SIP and related techniques in a mobile context and showing the additional challenges that the mobile environment brings. The book is therefore also an ideal guide for professionals who come from a telecommunications background.

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PART

I

Fundamentals

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CHAPTER

Introduction

1

There is a growing interest, both in the Internet and in the telecommunication industries, in multimedia communication services. An increasing number of Internet users who used to just surf the web or send emails are now becoming addicted to services such as Instant Messaging (IM), online gaming, and voice and video on the Net. These are examples of multimedia communication services delivered over the Internet that are enabled by the Session Initiation Protocol (SIP) in conjunction with other protocols.

In this first introductory chapter, we will explain what we mean by multimedia communication services. We will position these services in the context of the rest of the applications provided over the Internet.

We also want to give the reader a first hint of why SIP plays so crucial a role in the Internet communications space. That will lead us to dive into the importance of the signaling concept. We will underline the relevance of the signaling concept by looking at a very simple example of voice communication.

SIP not only enables voice on the Internet, but also a completely new universe of Total Communication services. To let the reader grasp the possibilities of SIP, we will show some examples of services and commercial products that currently use SIP.

SIP is, like any other Internet protocol, defined and developed by the Internet Engineering Task Force (IETF). More specifically, the core SIP specification is documented in [RFC 3261], and we will be referring throughout this book to this and other Internet specifications. So, in this chapter, we will also try to understand a bit better the SIP-related working groups in the IETF and the specifications they produce.

1.1 IP Multimedia Communication Services

A lot of very different services can be offered on top of the Internet and, in general, on top of an Internet Protocol (IP) network—a network based on the Internet Protocol.[1] It is not at all easy to find a categorization of those services

[1] Internet Protocol and IP networks are reviewed in Chapter 3.

from the user's perspective, but we will try to offer a simple one here, with the purpose of allowing us to understand what the remit of IP multimedia communication services is.

A very high-level approach might split the services offered on the Internet into three different categories or domains from the end-user perspective. A first category might include the infotainment services—that is, those services that give the user access to information and entertainment applications typically stored and executed in remote servers. The web would represent the paradigm for this kind of services. A second category would include the streaming services. These allow the user to access, in real-time, either live or stored time-based media content. Video-on-Demand (VOD) or the hot Internet Protocol Television (IPTV) service would fall into this category. The third service type includes the communication services—that is, those that allow people to communicate with each other using different types of media. A voice call or an email exchange would be examples of communication services (Figure 1.1).

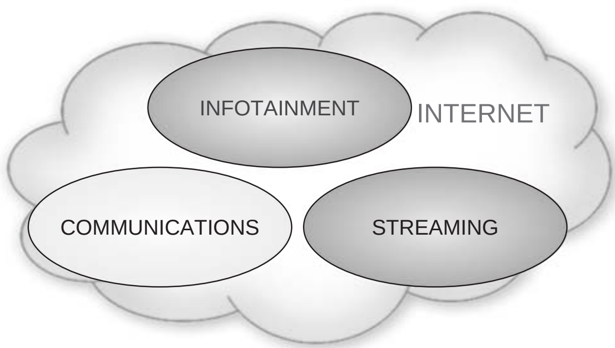

FIGURE 1.1

Communication services can be further classified into offline and online. In online communications, both originator and recipient need to be "connected" simultaneously for communication to happen, and the exchange of information occurs immediately between them. Examples of this include a voice call, an IM exchange, or a chess game.

In offline communication services, the involved parties do not necessarily need to be "connected" simultaneously for communication to happen. The popular email service is a good example of this. In the email service, the submission of information is decoupled from its reception by a store and forward mechanism, so the parties can communicate with each other even if they are not connected at the same time. Let us imagine that John wants to send an email to Alice. He switches on his computer, starts the email program, and sends the message. At that point, John closes the program and switches off his computer. Sometime later, Alice starts her email application and checks if new mail has arrived. She sees John's email and reads it. As we can see, John and Alice do not need to be connected simultaneously for the communication to happen.

The type of information (i.e., media) that can be exchanged in online communication services can be quite diverse. For instance, we might want to exchange real-time media such as voice or video, which have very stringent timing requirements. Packets containing voice samples should be received at regular intervals of some milliseconds so as to allow the receiver to play them back at the appropriate rate.

We might also want to exchange quasi-real-time information—that is, information that has requirements for timely delivery, but not as strong as in the case of voice. An example is an IM session or a chess game. In order to keep the interactivity in the session, data needs to arrive quickly enough—though, in this case, one or two seconds' delay would not impact the end user's experience.

Another type of information that we might want to exchange in online communication services is a prestored image or file. This scenario typically occurs in combination with an exchange of other types of media. Take, for instance, the case of John and Alice, who are engaged in a Voice over Internet Protocol (VoIP) conversation. John is at his 3G (third-generation) IP multimedia-enabled phone. Meanwhile, Alice is sitting at home in front of her PC. While talking to Alice, John takes a picture of a beautiful landscape with the camera integrated into his phone. He decides to show the picture to Alice. The image file would, in this case, be sent online and conveyed immediately to the recipient while both parties are talking so that they can comment on it (Figure 1.2).

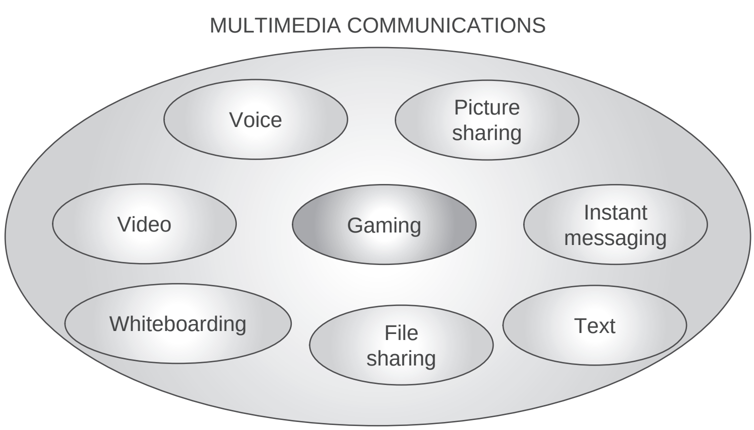

FIGURE 1.2

Online IP communication services are typically referred to as IP multimedia communication services, and that is the term that we will use throughout this book.

Unlike what occurs in other type of services, signaling plays a key role in IP multimedia communication services. SIP is typically used as the application-level signaling protocol in that remit, and therefore its role is crucial.

It is important to understand that SIP has not been designed to replace existing Internet application-level protocols such as those used in web (Hypertext Transfer Protocol, or HTTP) or email (Simple Mail Transfer Protocol, or SMTP; Post

Office Protocol version 3, or POP3; Internet Message Access Protocol version 4, or IMAP4). On the contrary, SIP covers a piece that was originally missing in the Internet architecture—that is, the signaling mechanism for multimedia communication services. SIP was designed in such a way as to fit smoothly with the existing Internet services and protocols such as web or email, so that, when combined with them, the promise of an all-IP total communications system encompassing all type of services can be made a reality.

It is also important to understand that SIP, all by itself, is not capable of delivering multimedia communication services. It needs to work alongside other protocols to accomplish that function. Most importantly, because SIP is a signaling protocol, it needs to work together with other protocols at the media layer.

1.2 The Role of Signaling and Media

In order to get a first understanding of the role of signaling and media protocols in IP multimedia communications, let us start by looking at a very simple example of voice communication on the Internet.

Let us assume that John and Alice, who are both in front of their PCs connected to the Internet, want to have a voice conversation. Each of them has a microphone and a loudspeaker connected to the soundcard in his or her computer. John is running a program such that, when he speaks on the microphone, the soundcard samples and encodes the voice signal into a bitstream. The computer program takes this stream of bits representing voice samples, and puts them into IP packets. These packets are then sent to Alice through the Internet. In order to make the packets reach Alice, the program in John's computer has to fill in the IP packets with the IP address of Alice's PC.

At the other end, Alice's PC receives the IP packets, decodes the voice samples in the payload, and feeds them into the soundcard so that they can be played.

In order for this real-time communication to work, this entire process has to be done with minimal latency, and it has to be done very regularly. Fortunately, PC programs can easily achieve this thanks to advances in computer technology.

In our example, we have considered that the voice samples are carried over IP protocol. Instead of conveying the samples directly over IP, an upper-level protocol is generally used. The information exchanged between the communicating parties (in this case, the voice) is typically referred to as media; thus, these protocols are referred to as media transport protocols. Different media transport protocols are specially suited to the type of media that needs to be conveyed. For instance, if the media is voice, a protocol called RTP (Real-time Transport Protocol) is typically used, which runs on top of User Datagram Protocol (UDP)/IP. RTP contains features that facilitate the transport of pure real-time traffic, such as voice, over IP networks. RTP and other media transport protocols are further described in Chapter 10.