Internet Communications Using SIP

Delivering VoIP and Multimedia Services with Session Initiation Protocol

Second Edition

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Alan B. Johnston

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We could not have written this book without the support of our forgiving spouses, Fabienne and Lisa, who held the fort while we were working on SIP. And to both our family members shouting, "Your SIP phone is ringing."

About the Authors

Dr. Henry Sinnreich (Richardson, TX) is Chief Technology Officer at Pulver.com, a leading media company for VoIP and Internet communication services. Dr. Sinnreich has held engineering and executive positions at MCI where he was an MCI fellow and has been involved in Internet and multimedia services for more than 12 years, including the development of the flagship MCI Advantage service based on SIP. Henry Sinnreich is also a contributor to IETF standards for Internet communications in such areas as SIP telephony devices and using RTP extensions for voice quality monitoring. He was awarded the title Pioneer for VoIP in 2000 at the VON Europe conference. Henry Sinnreich has been a cofounder and board member of the International SIP Forum based in Stockholm. He is a frequent speaker and is known as the leading evangelist, worldwide, for SIP based VoIP, presence, IM, multimedia, and integration of applications with communications. Dr. Sinnreich is also a guest lecturer at the Engineering School of the Southern Methodist University in Dallas, TX.

Alan B. Johnston (St. Louis, MO) is a Consulting Member of Technical Staff at Avaya, Inc. He has coauthored the core Internet SIP standard RFC 3261 and four other SIP related RFCs. He is the co-chair of the IETF Centralized Conferencing Working Group and is on the board of directors of the International SIP Forum. His current areas of interest include peer-to-peer SIP and security. Dr. Johnston is a frequent speaker and lecturer on SIP and contributor to various publications, and is an adjunct professor at Washington University in St. Louis, MO.

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Foreword

About 10 years ago, the first drafts describing the Session Initiation Protocol (1996) were published, with the rather modest ambition of setting up multicast groups for multimedia conferences. In the intervening decade, a draft of about 20 pages has turned into an ecosystem of dozens of RFCs, hundreds of Internet drafts—and several books, conferences, and a magazine. It has become difficult to get a feel for the overall landscape, to distinguish the important core concepts from the niche applications. This book offers a detailed, technically informed, yet accessible, introduction to the overall SIP ecosystem, suitable both for someone who needs to understand the technology to make strategic decisions and implementers who need to build new components.

SIP is part of the second wave of Internet application protocol. While the first wave largely focused on asynchronous communications (such as e-mail, and data transfer), this second wave introduces the notion of interactive, human-to-human communication that allows integration with any media, not just voice. As SIP and interactive communications have matured, the goal for human-to-human communication has shifted. Initially, cell phones promised voice communication at any time, at any place. Multimedia communications, on PCs and maybe emerging cellular networks, allow us to add "any media." However, the "any time, any place, any media" can also turn us into slaves of our communications devices, interrupting our ability to think, to eat in peace, and to meet in person. Thus, our goal has to be to design communications technology that offers the right media, at the right place, and at the right time. With some of the advanced functionality of SIP, such as presence, location-based services, user-created services, and caller preferences, we can get closer to creating communication systems that support our work and enhance our personal life.

With new communications technologies, there is always the temptation to mimic the old. E-mail inherited aspects of the interoffice memo and fax; web pages attempted to look like newsprint and brochures. However, in VoIP, there is the particular temptation to recreate old technology features, as interoperability with the old PSTN will remain important for at least another decade. Fax-to-email gateways were never quite as important as VoIP-to-PSTN gateways. This emphasis on interoperability with 100-year-old technology has provided a financial motivation—provide the same service more cheaply. However, this may also hold back the promise offered by Internet-based multimedia communications, such as the integration of presence, the ability not just to communicate by voice and maybe video but also to share any application, or the ability to customize the user experience and integrate interactive communications with existing Internet tools and applications. Just as most microprocessors are embedded in household appliances and cars, not desktop PCs and laptops, we might find that Internet-based voice and multimedia communications will be integrated into games, appliances, and cameras, or be hidden behind a link on a web page, rather than dialed by name or number. As for many of the most innovative applications, users will likely not even consider them phone services at all, but extensions that make some other application more productive or more fun.

This book is like a good tour guide to a foreign country. It doesn't just describe the major sites and tourist attractions; it lets the reader share in the history, spirit, language, and culture of the place. Natives write the best tour guides, and the authors have been living and working in SIP land since it was a small outpost in one large country called the IETF. The authors have served as ambassadors in lands near and far, but have also made major contributions to the development of this part of the Internet landscape, always reminding others of the original goals of the first inhabitants. After taking the tour, the reader will be ready not just to show off a stamp on a passport or certificate but also to contribute to new modes of communications. SIP land is still young and needs lots of pioneers who can push the frontiers of Internet-enabled communications. There might not always be gold in those hills, but enriching human communications will always be its own reward.

Henning Schulzrinne
Professor, Columbia University

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Most ideas and inspirations driving SIP are due to Prof. Henning Schulzrinne from Columbia University and to Jonathan Rosenberg from DynamicSoft and are reflected in this book. Among the many industry contributors, we gratefully acknowledge discussions and guidance from Rohan Mahy from Cisco Corporation, Gonzalo Camarillo and Adam Roach from L.M. Ericsson. Jiri Kuthan from GMD Focus, Berlin, was helpful with SIP tutorial charts and with discussions in transatlantic calls using SIP phones—again, calls of crystal clear clarity to our surprise. The authors are grateful to Richard Shockey from NeuStar, Inc. and Douglas Ranalli from NetNumber, Inc. for numerous discussions regarding ENUM. Theodore Havinis has contributed to the SIP-QoS-AAA aspect for mobile users.

We acknowledge countless helpful discussions and insight from many participants in the IETF and especially to Scott Bradner for holding the authors and others in the IETF SIP community in line to the true conceptual, technical, and procedural spirit of the Internet.

Jeff Pulver has played a special role in providing a platform and leading exhibition of products for what was initially an obscure and unknown protocol in the Voice ON the Net (VON) and other conferences held in America, Europe, and Asia.

Carrol Long, Kevin Shafer, and Adoabi Obi Tulton from John Wiley & Sons have been instrumental in editing this book.

Introduction

The second edition of *Internet Communications Using SIP* had to be rewritten almost from the ground up, because of the dramatic changes in the industry in the five years that have passed since the first edition. Some of the developments had been envisaged in the first edition, but naturally, some have not.

The Internet Has Replaced the Telephone System and the Telecommunication Networks

Since the publication in 2001 of the first edition of this book, *Internet Communications Using SIP*, Voice over IP (VoIP) has developed from an emerging technology to the recognized replacement of existing global telephone systems based on Time Division Multiplex (TDM) circuit switching. The Internet has also replaced the proposed connection-oriented offsprings of TDM, such as the Integrated Services Digital Network (ISDN) and the Asynchronous Transfer Multiplex (ATM) based broadband version BISDN, envisaged for the telecommunications industry by the International Telecommunications Union ITU-T standards body. TDM, ATM, ISDN, and BISDN are now history.

All wired and wireless communications are instead migrating to the Internet standards developed by the Internet Engineering Task Force (IETF). The legacy telecommunication networks, while still dominant, are recognized as a present-day cash cow only and are scheduled for replacement by IP networks.

The end-to-end nature of the Internet that places intelligence in the applications running in the endpoints and gives control to the user at the endpoints has indeed replaced TDM-based telephony with central control. The Internet has also proven to be the home network for other types of communications, information, entertainment, and data applications. To quote Jon Peterson, area director of the IETF:

"The Internet is the service."

The Session Initiation Protocol Is the Standard for VoIP and Multimedia Communications

Another change from the first edition of this book is the Session Initiation Protocol (SIP), which has been adopted by practically all public VoIP service providers for wired and wireless communications. The discussions about SIP versus H.323 standardized by the ITU-T are over as well. The installed base of H.323 is considered a liability and planned for replacement by SIP sooner or later.

A global industry has emerged to take advantage of SIP and its associated IETF standards for real-time communications. More than 560 VoIP service providers have been reported [1] in early 2006, most of them using SIP-based networks. The list of SIP-based equipment (such as SIP phones, software for PCs, and mobile devices, servers, gateways, and so on) is now large and still growing. Actually, all equipment and system vendors are now supporting SIP.

Presence and Instant Messaging Are Mainstream Communications

Presence and instant messaging (IM) are now mainstream with consumers and, in the enterprise, complementing or sometimes replacing voice communications in specific situations (such as in circumstances where silence is required). Even for VoIP, presence has emerged not only as a valuable enhancement, but presence may be the dial tone of the twenty-first century.

Presence and event-based communications have enabled the integration of communications with applications. Presence and IM are discussed in Chapter 13, "Presence and Instant Messaging."

The so-called IM services provided by large Internet companies, such as AOL, Apple, Google, IBM, Microsoft, Skype (not SIP-based), and Yahoo!, actually carry at present most of the public VoIP traffic between end users around the globe.

It is not far-fetched to see the IM Internet companies replacing the former telephone companies in the voice communication business. Many legacy telecommunication companies are also using VoIP to replace the internal TDM voice networks, but their VoIP services may not survive the advanced technologies deployed by the IM Internet companies and the challenge posed by peer-to-peer (P2P) communications.

Redefining Communications: Mobility, Emergency and Equal Access for the Disabled

Internet communications have been known not to be dependent on the location on the Internet. Application-level mobility based on SIP is a key component to seamless mobile communications, as discussed in Chapter 15, "SIP Application Level Mobility."

Emergency calling services by users in distress using the Internet (such as 911 in the United States or 112 in Europe) are far more powerful and cost less than the Public Switched Telephone Network (PSTN) based emergency services. Internet-based emergency calling is indeed in the design stage in a number of countries. Chapter 16, "Emergency and Preemption Communication Services," discusses Internet-based emergency services.

The multimedia nature of Internet communications gives hearing- and speech-impaired people the opportunity to fully participate in rich communications for work and in personal life. Chapter 17, "Accessibility for the Disabled," discusses access to communications for disabled people.

The Rise of Peer-to-Peer Communications

P2P traffic has risen in the Internet since around 2000 and became the dominant part of Internet traffic by 2004. Since 2004, Skype (which is based on P2P VoIP, IM, and presence) has also become by far the dominant VoIP provider worldwide. Since P2P SIP standards work is just emerging as of this writing, Skype can be considered a prestandard P2P Internet communication service.

The reasons for the emergence of overlay networks and P2P applications and their nature are discussed in Chapter 20, "Peer-to-Peer SIP," and also in Chapter 6, "SIP Overview." Though the present VoIP industry is built on client-server (CS) SIP, this may significantly change. To quote David Bryan from `p2p.org`:

> *"P2P SIP may change VoIP to the same extent that VoIP has changed telecommunications."*

VoIP and Multimedia Communications Services Are Still Fragmented

In spite of all the technological progress, VoIP, IM, presence, and multimedia services are still a highly fragmented industry:

- Telephone services based on VoIP operate as islands and can interconnect (as of this writing) using mostly the legacy Public Switched Telephone Network (PSTN). The service model is giving broadband users access to the legacy telephone system, actually a voice gateway service between the Internet and TDM. The business model of most VoIP service providers is just lower cost for legacy-style telephone service, also called *PSTN over IP*. The PSTN gateway services are using IP inside their networks, but users are not exposed to the rich IP services, except when all parties are on the same network.

- The most successful public voice, IM, and presence service is Skype, which is not standards-based.
- Walled gardens: The fragmentation of communications is still actively pursued by most mobile service providers by deploying systems where their users can get rich IP multimedia services only on their own networks. The fees to communicate between mobile service providers are a significant part of the business model, and open connectivity to the Internet ("Internet neutrality") is still a hotly debated issue. Internet neutrality is also still debated by many broadband Internet access providers (such as DSL and cable companies), although we believe that enlightened government regulators in the developed countries will weigh in favor of users and open network access in general.

The proliferation of islands for communications makes them less useful the more there are, since this proliferation is in denial of Metcalf's law that the value of a network increases with the square of the number of points attached to the network. The Internet with more than *1 billion* attached endpoints has thus the highest value for communications. By contrast, the mobile phone industry boasts *3 billion* users, but in many fragmented networks.

Past Obsessions and Present Dangers: QoS and Security

Network-based quality of service (QoS) for voice and the reliability of the legacy telephone network have long been used by telephone industry marketers to scare users away from VoIP. In the meantime, all public VoIP services have proven that Internet best-effort QoS works just fine, as long network congestion is avoided. Internet-based voice can actually be much better than the 3.1 kHz voice over the PSTN. As for reliability, all recent major man-made and natural disasters have proven the Internet and VoIP to be more resilient than the existing wireline and wireless telephone networks.

Chapter 18, "Quality of Service for Real-Time Internet Communications," is aimed at a balanced approach for QoS, and Chapter 16, "Emergency and Preemption Communication Services," discusses the Emergency Services based on SIP.