CHAPTER

1

Introduction

The telecommunications, television, and data network industries are driven by growth based on new services, more complete global coverage, and consolidation. In this chapter, we will explore some of the problems and solutions for end users and service providers alike.

Problem: Too Many Networks

Before the emergence of the Internet, users and service providers were generally accustomed to thinking in terms of three distinct network types: networks for data, networks for voice, and networks for television. Each of these dedicated network types could, in turn, be divided into many incompatible regional and even country-specific flavors with different protocol variants.

Thus, we find many types of telephony numbering plans, signaling, and audio encoding, several TV standards, and several types and flavors of what the telecom industry calls *data networks*—all of them incompatible and impossible to integrate into one single global network.

Data networks that originated in the telecom industry come in many forms, such as digital private lines, X.25, ISDN, SMDS, frame relay, and ATM networks. These so called data networks are mostly inspired by circuit-switched telephony concepts. Their name is derived from the fact that they were not designed to carry voice.

Voice networks are also used for data and fax due to their general availability. However, these networks have come to the end of their evolution, since they are fundamentally optimized for voice only. Finally, TV networks were designed and optimized for the distribution of entertainment video streams. The proliferation of various types of wireless mobile networks and pagers has increased network diversity even more.

Needless to say, all three network types have specific end-user devices that cannot be ported to other service providers or network types, and most often cannot be globally deployed.

Network Consolidation

The Internet has benefited from a number of different fundamentals compared to legacy networks, such as the tremendous progress of computing technology and globalization. This progress can be attributed to the expertise of the research, academic, and engineering communities whose dedication to excellence and open collaboration on a global basis have surpassed the usual commercial pressure for time to market and competitive secrecy.

The result is an Internet that uses consistent protocols on a global basis and is equally well suited to carry data transactions, voice, and video.

Voice on the Net

Although the Internet quickly established itself as the preeminent network for data, commercial transactions, and audio-video distribution, the use of voice over the Internet has been slower to develop. This has less to do with the capability of the Internet to carry voice with equal or higher quality than the telephone network, but rather the critical nature of signaling in voice services, as we will see in Chapter 5, *SIP Overview*.

There are various approaches for voice services over the Internet, based on different signaling and control design. Some examples include the following:

- Use *signaling* concepts from the telephone industry: H.323.
- Use *control* concepts from the telephone industry: Softswitches.
- Use the Internet-centric *protocol*: Session Initiation Protocol (SIP), the topic of this book.

The movement from such concepts as telephony call models to sessions between any processes on any platforms anywhere on the Internet is opening up completely new types of communication services.

The use of SIP for establishing voice, video, and data sessions places telephony as just another service on the Internet using similar addressing, data types, software, protocols, and security. A separate network for voice is no longer necessary.

Complete integration of voice with all other Internet services is probably the greatest opportunity for innovation. The open and distributed nature of this service and network model will empower many innovators, similar to what has happened on the Internet and the resulting new economy.

SIP Is Not a Miracle Protocol

As discussed in Chapter 2, *IP Communications Enabled by SIP*, SIP is not a miracle protocol and is not designed to do more than discover remote users and establish interactive communication sessions. SIP is not meant to assure quality of service (QoS) all by itself or to transfer large amounts of data. It is not applicable for conference floor control. Neither is it meant to replace all known telephony features, many of which are due to the limitations of circuit switched voice or to the regulation of voice services. And such a list can go on.

In summary, various other Internet protocols are better suited for various features. As for telephony, not all telephone network features lend themselves to replication on the Internet.

The Short History of SIP

By 1996, the Internet Engineering Task Force (IETF) had already developed the basics for multimedia on the Internet (see Chapter 4, *Internet Multimedia and Conferencing*) in the Multi-Party, Multimedia Working Group. Two proposals, the Session Initiation Protocol (SIP) by Mark Handley and the Simple Conference Invitation Protocol (SCIP) by Henning Schulzrinne, were announced and later merged to form Session Initiation Protocol. The new protocol also preserved the HTTP orientation from the initial SCIP proposal that later proved to be crucial to the merging of IP communications on the Internet.[1]

Henning Schulzrinne focused on the continuing development of SIP with the objective of "reengineering the telephone system from ground up," an "opportunity that appears only once after 100 years" as we heard him argue at a time when few believed this was practical.

[1] The authors would like to thank Professor Dr. Jörg Ott, co-chair of the SIP WG and early contributor to the MMUSIC WG for helping with data on SIP history.

SIP was approved as RFC 2543 in the IETF in March 1999. Due to the tremendous interest and the increasing number of contributions to SIP, a separate SIP Working Group (WG) was formed in September 1999. The SIP for Instant Messaging and Presence Leveraging (SIMPLE) was formed in March 2001.

As of this writing, due to the overload of the SIP WG, another working group is being chartered, known as the Session Initiation Protocol Project INvestiGation (SIPPING) working group. The core task of this group is on moving SIP from Proposed Standard to Draft Standard. All protocol extensions will remain in the SIP WG, while SIPPING will concentrate on the frameworks, requirements, and practices related to SIP and its extensions.

References in This Book

Due to the many developments on the Internet, SIP is continuing as a work in progress, as evidenced by its rapid growth. The IETF SIP WG has twice as many drafts from contributors as any other working group. This book reflects SIP developments up to and including the fiftieth IETF in 2001.

We have included, by necessity, many Internet drafts that are designated *work in progress*, since they are the only reference source for this particular information. Some of these drafts may become standards by the time the reader is ready to use them, some may be work in progress and have a higher version number, and still others may be found only in an archive for *expired drafts*.

The SIP WG drafts that are work in progress can be found online at the IETF Web site:

http://ietf.org/html.charters/sip-charter.html

Additional individual submissions and Internet drafts from other working groups can be found at:

http:// ietf.org/ID.html.

SIP-related drafts that have *expired* (older than 6 months) can be found at several archives. At the time of this writing, some of the sites are:

http://www.cs.columbia.edu/sip/drafts/

http://softarmor.com/sipwg/

http://iptel.org/info

Few books have been published on Internet multimedia, Voice over IP, and SIP as yet, and some are listed here. They focus mainly on how SIP works. This book is less about explaining how SIP works, but rather what it does and the new communications and services it enables.

We have reproduced some of the exciting services and features discussed in the IETF SIP WG and its offspring the SIPPING and SIMPLE Working Groups. Also included in our discussion are some drafts from Bird Of Feather (BOF) sessions that have not even made it to an accepted WG charter, such as the IPAC BOF (at IETF 50) on IP appliances.

Many of these expired proposals may not make it as an IETF standard for various reasons, but represent good work, often backed up by running code. The references to such expired Internet drafts are intended to make readers aware of these ideas that may otherwise remain buried in an archive. Such references are clearly marked as expired, so as to distinguish them from accepted work in progress items of IETF WGs that are on the path toward acceptance as standards.

References for Telephony

We assume throughout this book some understanding of telephone services and of telecommunication protocols. There is a vast literature pool available on telephony and telecommunications. We refer the reader to *Newton's Telecommunications Dictionary* [4], to brush up on various terms used in the following chapters.

References

[1] Jon Crowcroft, Mark Handley, and Ian Wakeman. *Internetworking Multimedia*, Morgan Kaufmann Publishers, London, New York, October 1999.

[2] Olivier Hersent, David Gurle, and Jean Pierre-Petit. *IP Telephony: Packet-Based Multimedia Communications Systems*, Addison-Wesley, Reading, MA, December 1999.

[3] Alan B. Johnston. *SIP: Understanding the Session Initiation Protocol*, Artech House, Boston, London, January 2001.

[4] Harry Newton. *Newton's Telecommunications Dictionary*, 17th ed., CMP Books, Manhasset, NY, March 2001.

[5] Gonzalo Camarillo. *SIP Demystified*, McGraw-Hill Professional Publishing, New York, August 2001.