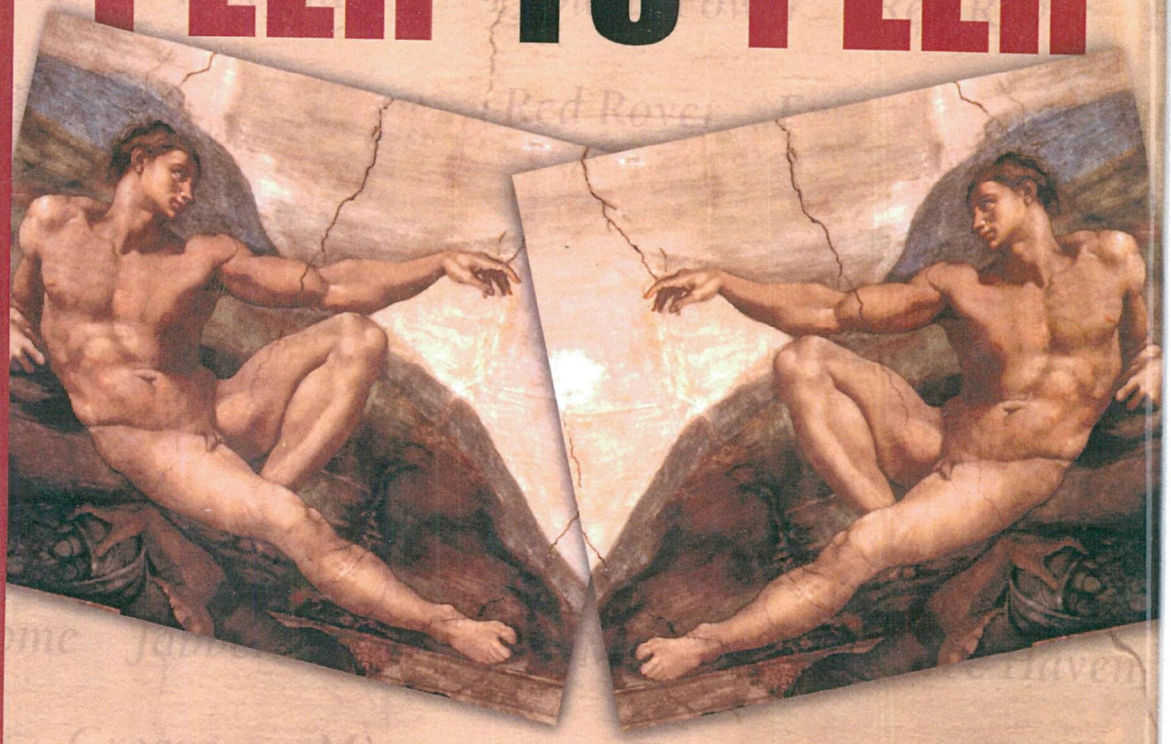
"Peer-to-peer is the next great thing for the Internet."

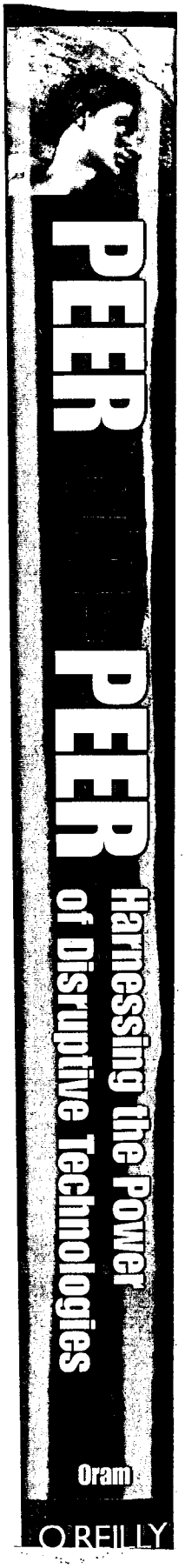
—Lawrence Lessig, Stanford Law School, author *Code and Other Laws of Cyberspace*

PEER-TO-PEER



Harnessing the Power of Disruptive Technologies

Edited by Andy Oram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Harnessing the Power
of Disruptive Technologies
Oram
O'REILLY

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Upstart software projects Napster, Gnutella, and Freenet have dominated newspaper headlines, challenging traditional approaches to content distribution with their revolutionary use of peer-to-peer file-sharing technologies. Reporters try to sort out the ramifications of seemingly ungoverned peer-to-peer networks. Lawyers, business leaders, and social commentators debate the virtues and evils of these bold new distributed systems. But what's really behind such disruptive technologies—the breakthrough innovations that have rocked the music and media worlds? And what lies ahead?

In this book, key peer-to-peer pioneers take us beyond the headlines and hype and show how the technology is changing the way we communicate and exchange information. Those working to advance peer-to-peer as a technology, a business opportunity, and an investment offer their insights into how the technology has evolved and where it's going. They explore the problems they've faced, the solutions they've discovered, the lessons they've learned, and their goals for the future of computer networking.

Until now, Internet communities have been limited by the flat interactive qualities of email and network newsgroups, where people can exchange recommendations and ideas but have great difficulty commenting on one another's postings, structuring information, performing searches, and creating

(continued on back flap)

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Harnessing the Benefits of a Disruptive Technology

Andy — This is only one of many things you set in motion by your deep + thoughtful approach to the future of computing. All the best!

Edited by Andy Oram

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Peer-to-Peer: Harnessing the Benefits of a Disruptive Technology
Edited by Andy Oram

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Contents

Preface vii
Andy Oram

Part I: Context and Overview

1 A Network of Peers: Peer-to-Peer Models Through the History of the Internet 3
Nelson Minar and Marc Hedlund

2 Listening to Napster 21
Clay Shirky

3 Remaking the Peer-to-Peer Meme 38
Tim O'Reilly

4 The Cornucopia of the Commons 59
Dan Bricklin

Part II: Projects

5 SETI@home 67
David Anderson

6 Jabber: Conversational Technologies 77
Jeremie Miller

7 Mixmaster Remailers 89
Adam Langley

8 Gnutella 94
Gene Kan

Microsoft Exhibit 1019

9 Freenet 123
Adam Langley

10 Red Rover 133
Alan Brown

11 Publius 145
Marc Waldman, Lorrie Faith Cranor, and Avi Rubin

12 Free Haven 159
Roger Dingledine, Michael J. Freedman, and David Molnar

Part III: Technical Topics

13 Metadata 191
Rael Dornfest and Dan Brickley

14 Performance 203
Theodore Hong

15 Trust 242
Marc Waldman, Lorrie Faith Cranor, and Avi Rubin

16 Accountability 271
Roger Dingledine, Michael J. Freedman, and David Molnar

17 Reputation 341
Richard Lethin

18 Security 354
Jon Udell, Nimisha Asthagiri, and Walter Tuvell

19 Interoperability Through Gateways 381
Brandon Wiley

Afterword 393
Andy Oram

Appendix: Directory of Peer-to-Peer Projects 399

Contributors 403

Index 409

Microsoft Exhibit 1019

Preface

Andy Oram, O'Reilly & Associates, Inc.

The term *peer-to-peer* rudely shoved its way to front and center stage of the computing field around the middle of the year 2000. Just as the early 20th-century advocates of psychoanalysis saw sex everywhere, industry analysts and marketing managers are starting to call everything they like in computers and telecommunications "peer-to-peer." At the same time, technologists report that fear and mistrust still hang around this concept, sometimes making it hard for them to get a fair hearing from venture capitalists and policy makers.

Yes, a new energy is erupting in the computing field, and a new cuisine is brewing. Leaving sexiness aside, this preface tries to show that the term peer-to-peer is a useful way to understand a number of current trends that are exemplified by projects and research in this book. Seemingly small technological innovations in peer-to-peer can radically alter the day-to-day use of computer systems, as well as the way ordinary people interact using computer systems.

But to really understand what makes peer-to-peer tick, where it is viable, and what it can do for you, you have to proceed to the later chapters of the book. Each is written by technology leaders who are working 'round the clock to create the new technologies that form the subject of this book. By following their thoughts and research, you can learn the state of the field today and where it might go in the future.

Some context and a definition

I mentioned at the beginning of this preface that the idea of peer-to-peer was the new eyebrow-raiser for the summer of 2000. At that point in history, it looked like the Internet had fallen into predictable patterns. Retail outlets had turned the Web into the newest mail order channel, while entertainment firms used it to rally fans of pop culture. Portals and search engines presented a small slice of

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Internet offerings in the desperate struggle to win eyes for banner ads. The average user, stuck behind a firewall at work or burdened with usage restrictions on a home connection, settled down to sending email and passive viewing.

In a word, boredom. Nothing much for creative souls to look forward to. An Olympic sports ceremony that would go on forever.

At that moment the computer field was awakened by a number of shocks. The technologies were not precisely new, but people realized for the first time that they were having a wide social impact:

Napster
: This famous and immensely popular music exchange system caused quite a ruckus, first over its demands on campus bandwidth, and later for its famous legal problems. The technology is similar to earlier systems that got less attention, and even today is rather limited (since it was designed for pop songs, though similar systems have been developed for other types of data). But Napster had a revolutionary impact because of a basic design choice: after the initial search for material, clients connect to each other and exchange data directly from one system's disk to the other.

SETI@home
: This project attracted the fascination of millions of people long before the Napster phenomenon, and it brought to public attention the promising technique of distributing a computation across numerous personal computers. This technique, which exploited the enormous amounts of idle time going to waste on PCs, had been used before in projects to crack encryption challenges, but after SETI@home began, a number of companies started up with the goal of making the technique commercially viable.

Freenet
: Several years before the peer-to-peer mania, University of Edinburgh researcher Ian Clarke started to create an elegantly simple and symmetric file exchange system that has proven to be among the purest of current models for peer-to-peer systems. Client and server are the same thing in this system; there is absolutely no centralization.

Gnutella
: This experimental system almost disappeared before being discovered and championed by open source developers. It is another file exchange system that, like Freenet, stresses decentralization. Its potential for enhanced searches is currently being explored.

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Jabber

This open source project combines instant messaging (supporting many popular systems) with XML. The emergence of Jabber proclaimed that XML was more than a tool for business-to-business (B2B) transaction processing, and in fact could be used to create spontaneous communities of ordinary users by structuring the information of interest to them.

.NET

This is the most far-reaching initiative Microsoft has released for many years, and they've announced that they're betting the house on it. .NET makes Microsoft's earlier component technology easier to use and brings it to more places, so that web servers and even web browsers can divide jobs among themselves. XML and SOAP (a protocol for doing object-oriented programming over the Web) are a part of .NET.

Analysts trying to find the source of inspiration for these developments have also noted a new world of sporadically connected Internet nodes emerging in laptops, handhelds, and cell phones, with more such nodes promised for the future in the form of household devices.

What thread winds itself around all these developments? In various ways they return content, choice, and control to ordinary users. Tiny endpoints on the Internet, sometimes without even knowing each other, exchange information and form communities. There are no more clients and servers—or at least, the servers retract themselves discreetly. Instead, the significant communication takes place between cooperating peers. That is why, diverse as these developments are, it is appropriate to lump them together under the rubric peer-to-peer.

While the technologies just listed are so new we cannot yet tell where their impact will be, peer-to-peer is also the oldest architecture in the world of communications. Telephones are peer-to-peer, as is the original UUCP implementation of Usenet. IP routing, the basis of the Internet, is peer-to-peer, even now when the largest access points raise themselves above the rest. Endpoints have also historically been peers, because until the past decade every Internet-connected system hosted both servers and clients. Aside from dial-up users, the second-class status of today's PC browser crowd didn't exist. Thus, as some of the authors in this book point out, peer-to-peer technologies return the Internet to its original vision, in which everyone creates as well as consumes.

Many early peer-to-peer projects have an overtly political mission: routing around censorship. Peer-to-peer techniques developed in deliberate evasion of mainstream networking turned out to be very useful within mainstream networking. There is nothing surprising about this move from a specialized and

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somewhat ostracized group of experimenters to the center of commercial activity; similar trends can be found in the history of many technologies. After all, organizations that are used to working within the dominant paradigm don't normally try to change that paradigm; change is more likely to come from those pushing a new cause. Many of the anti-censorship projects and their leaders are featured in this book, because they have worked for a long time on the relevant peer-to-peer issues and have a lot of experience to offer.

Peer-to-peer can be seen as the continuation of a theme that has always characterized Internet evolution: loosening the virtual from the physical. DNS decoupled names from physical systems, while URNs were meant to let users retrieve documents without knowing the domain names of their hosts. Virtual hosting and replicated servers changed the one-to-one relationship of names to systems. Perhaps it is time for another major conceptual leap, where we let go of the notion of location. Welcome to the Heisenberg Principle as applied to the Internet.

The two-way Internet also has a social impact, and while this book is mostly about the technical promise of peer-to-peer, authors also talk about its exciting social promise. Communities have been forming on the Internet for a long time, but they have been limited by the flat interactive qualities of email and network newsgroups. People can exchange recommendations and ideas over these media, but they have great difficulty commenting on each other's postings, structuring information, performing searches, or creating summaries. If tools provided ways to organize information intelligently, and if each person could serve up his or her own data and retrieve others' data, the possibilities for collaboration would take off. Peer-to-peer technologies could enhance almost any group of people who share an interest—technical, cultural, political, medical, you name it.

How this book came into being

The feat of compiling original material from the wide range of experts who contributed to this book is a story all in itself.

Long before the buzz about peer-to-peer erupted in the summer of 2000, several people at O'Reilly & Associates had been talking to leaders of interesting technologies who later found themselves identified as part of the peer-to-peer movement. At that time, for instance, we were finishing a book on SETI@home (*Beyond Contact*, by Brian McConnell) and just starting a book on Jabber. Tim O'Reilly knew Ray Ozzie of Groove Networks (the creator of Lotus Notes), Marc Hedlund and Nelson Minar of Popular Power, and a number of other technologists working on technologies like those in this book.

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As for me, I became aware of the technologies through my interest in Internet and computing policy. When the first alarmist news reports were published about Freenet and Gnutella, calling them mechanisms for evading copyright controls and censorship, I figured that anything with enough power to frighten major forces must be based on interesting and useful technologies. My hunch was borne out more readily than I could have imagined; the articles I published in defense of the technologies proved to be very popular, and Tim O'Reilly asked me to edit a book on the topic.

As a result, contributors came from many sources. Some were already known to O'Reilly & Associates, some were found through a grapevine of interested technologists, and some approached us when word got out that we were writing about peer-to-peer. We solicited chapters from several people who could have made valuable contributions but had to decline for lack of time or other reasons. I am fully willing to admit we missed some valuable contributors simply because we did not know about them, but perhaps that can be rectified in a future edition.

In addition to choosing authors, I spent a lot of effort making sure their topics accurately represented the field. I asked each author to find a topic that he or she found compelling, and I weighed each topic to make sure it was general enough to be of interest to a wide range of readers.

I was partial to topics that answered the immediate questions knowledgeable computer people ask when they hear about peer-to-peer, such as "Will performance become terrible as it scales?" or "How can you trust people?" Naturally, I admonished authors to be completely honest and to cover weaknesses as well as strengths.

We did our best, in the short time we had, to cover everything of importance while avoiding overlap. Some valuable topics could not be covered. For instance, no one among the authors we found felt comfortable writing about search techniques, which are clearly important to making peer-to-peer systems useful. I believe the reason we didn't get to search techniques is that it represents a relatively high level of system design and system use—a level the field has not yet achieved. Experiments are being conducted (such as InfraSearch, a system built on Gnutella), but the requisite body of knowledge is not in place for a chapter in this book. All the topics in the following pages—trust, accountability, metadata—have to be in place before searching is viable. Sometime in the future, when the problems in these areas are ironed out, we will be ready to discuss search techniques.

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Thanks to Steve Burbeck, Ian Clarke, Scott Miller, and Terry Steichen, whose technical reviews were critical to assuring accurate information and sharpening the arguments in this book. Thanks also to the many authors who generously and gently reviewed each other's work, and to those people whose aid is listed in particular chapters.

Thanks also to the following O'Reilly staff: Darren Kelly, production editor; Leanne Soylemez, who was the copyeditor; Rachel Wheeler, who was the proofreader; Matthew Hutchinson, Jane Ellin, Sarah Jane Shangraw, and Claire Cloutier, who provided quality control; Judy Hoer, who wrote the index; Lucy Muellner and Linley Dolby, who did interior composition; Edie Freedman, who designed the cover of this book; Emma Colby, who produced the cover layout; Melanie Wang and David Futato, who designed the interior layout; Mike Sierra, who implemented the design; and Robert Romano and Jessamyn Reed, who produced the illustrations.

Contents of this book

It's fun to find a common thread in a variety of projects, but simply noting philosophical parallels is not enough to make the term peer-to-peer useful. Rather, it is valuable only if it helps us develop and deploy the various technologies. In other words, if putting two technologies under the peer-to-peer umbrella shows that they share a set of problems, and that the solution found for one technology can perhaps be applied to another, we benefit from the buzzword. This book, then, spends most of its time on general topics rather than the details of particular existing projects.

Part I, *Context and Overview*, contains the observations of several thinkers in the computer industry about the movements that have come to be called peer-to-peer. These authors discuss what can be included in the term, where it is innovative or not so innovative, and where its future may lie.

Chapter 1, *A Network of Peers: Peer-to-Peer Models Through the History of the Internet*, describes where peer-to-peer systems might offer benefits, and the problems of fitting such systems into the current Internet. It includes a history of early antecedents. The chapter is written by Nelson Minar and Marc Hedlund, the chief officers of Popular Power.

Chapter 2, *Listening to Napster*, tries to tie down what peer-to-peer means and what we can learn from the factors that made Napster so popular. The chapter is written by investment advisor and essayist Clay Shirky.

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Chapter 3, *Remaking the Peer-to-Peer Meme*, contrasts the way the public often views a buzzword such as peer-to-peer with more constructive approaches. It is written by Tim O'Reilly, founder and CEO of O'Reilly & Associates, Inc.

Chapter 4, *The Cornucopia of the Commons*, reveals the importance of maximizing the value that normal, selfish use adds to a service. It is written by Dan Bricklin, cocreator of Visicalc, the first computer spreadsheet.

Some aspects of peer-to-peer can be understood only by looking at real systems. Part II, *Projects*, contains chapters of varying length about some important systems that are currently in operation or under development.

Chapter 5, *SETI@home*, presents one of the most famous of the early crop of peer-to-peer technologies. Project Director David Anderson explains why the team chose to crunch astronomical data on millions of scattered systems and how they pulled it off.

Chapter 6, *Jabber: Conversational Technologies*, presents the wonderful possibilities inherent in using the Internet to form communities of people as well as automated agents contacting each other freely. It is written by Jeremie Miller, leader of the Jabber project.

Chapter 7, *Mixmaster Remailers*, covers a classic system for allowing anonymous email. Other systems described in this book depend on Mixmaster to protect end-user privacy, and it represents an important and long-standing example of peer-to-peer in itself. It is written by Adam Langley, a Freenet developer.

Chapter 8, *Gnutella*, offers not only an introduction to one of the most important of current projects, but also an entertaining discussion of the value of using peer-to-peer techniques. The chapter is written by Gene Kan, one of the developers most strongly associated with Gnutella.

Chapter 9, *Freenet*, describes an important project that should be examined by anyone interested in peer-to-peer. The chapter explains how the system passes around requests and how various cryptographic keys permit searches and the retrieval of documents. It is written by Adam Langley.

Chapter 10, *Red Rover*, describes a fascinating system for avoiding censorship and recrimination for the distribution of files using electronic mail. It is written by Alan Brown, the developer of Red Rover.

Chapter 11, *Publius*, describes a system that distributes material through a collection of servers in order to prevent censorship. Although Publius is not a pure peer-to-peer system, its design offers insight and unique solutions to many of the problems faced by peer-to-peer designers and users. The chapter is written

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by Marc Waldman, Lorrie Faith Cranor, and Avi Rubin, the members of the Publius team.

Chapter 12, *Free Haven*, introduces another set of distributed storage services that promotes anonymity with the addition of some new techniques in improving accountability in the face of this anonymity. It is written by Roger Dingledine, Michael Freedman, and David Molnar, leaders of the Free Haven team.

In Part III, *Technical Topics*, project leaders choose various key topics and explore the problems, purposes, and promises of the technology.

Chapter 13, *Metadata*, shows how to turn raw data into useful information and how that information can support information seekers and communities. Metadata can be created through XML, RDF, and other standard formats. The chapter is written by Rael Dornfest, an O'Reilly Network developer, and Dan Brickley, a longstanding RDF advocate and chair of the World Wide Web Consortium's RDF Interest Group.

Chapter 14, *Performance*, covers a topic that has been much in the news recently and comes to mind immediately when people consider peer-to-peer for real-life systems. This chapter examines how well a peer-to-peer project can scale, using simulation to provide projections for Freenet and Gnutella. It is written by Theodore Hong of the Freenet project.

Chapter 15, *Trust*, begins a series of chapters on the intertwined issues of privacy, authentication, anonymity, and reliability. This chapter covers the basic elements of security, some of which will be well known to most readers, but some of which are fairly novel. It is written by the members of the Publius team.

Chapter 16, *Accountability*, covers ways to avoid the "tragedy of the commons" in shared systems—in other words, the temptation for many users to freeload off the resources contributed by a few. This problem is endemic to many peer-to-peer systems, and has led to several suggestions for micropayment systems (like Mojo Nation) and reputation systems. The chapter is written by leaders of the Free Haven team.

Chapter 17, *Reputation*, discusses ways to automate the collection and processing of information from previous transactions to help users decide whether they can trust a server with a new transaction. The chapter is written by Richard Lethin, founder of Reputation Technologies, Inc.

Chapter 18, *Security*, offers the assurance that it is technically possible for people in a peer-to-peer system to authenticate each other and ensure the integrity and secrecy of their communications. The chapter accomplishes this by describing the industrial-strength security system used in Groove, a new commercial

Microsoft Exhibit 1019

groupware system for small collections of people. It is written by Jon Udell, an independent author/consultant, and Nimisha Asthagiri and Walter Tuvell, staff of Groove Networks.

Chapter 19, *Interoperability Through Gateways*, discusses how the best of all worlds could be achieved by connecting one system to another. It includes an encapsulated comparison of several peer-to-peer systems and the advantages each one offers. It is written by Brandon Wiley, a developer of the Freenet project.

The appendix lists some interesting projects, companies, and standards that could reasonably be considered examples of peer-to-peer technology.

Peer-to-peer web site

O'Reilly has created the web site *http://openp2p.com* to cover peer-to-peer (P2P) technology for developers and technical managers. The site covers these technologies from inside the communities producing them and tries to profile the leading technologists, thinkers, and programmers in the P2P space by providing a deep technical perspective.

We'd like to hear from you

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part one

Context and Overview

This part of the book offers some high-level views, defining the term "peer-to-peer" and placing current projects in a social and technological context.

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chapter one

A Network of Peers

Peer-to-Peer Models Through the History of the Internet

Nelson Minar and Marc Hedlund, Popular Power

The Internet is a shared resource, a cooperative network built out of millions of hosts all over the world. Today there are more applications than ever that want to use the network, consume bandwidth, and send packets far and wide. Since 1994, the general public has been racing to join the community of computers on the Internet, placing strain on the most basic of resources: network bandwidth. And the increasing reliance on the Internet for critical applications has brought with it new security requirements, resulting in firewalls that strongly partition the Net into pieces. Through rain and snow and congested Network Access Providers (NAPs), the email goes through, and the system has scaled vastly beyond its original design.

In the year 2000, though, something has changed—or, perhaps, reverted. The network model that survived the enormous growth of the previous five years has been turned on its head. What was down has become up; what was passive is now active. Through the music-sharing application called Napster, and the larger movement dubbed "peer-to-peer," the millions of users connecting to the Internet have started using their ever more powerful home computers for more than just browsing the Web and trading email. Instead, machines in the home and on the desktop are connecting to each other directly, forming groups and collaborating to become user-created search engines, virtual supercomputers, and filesystems.

Not everyone thinks this is such a great idea. Some objections (dealt with elsewhere in this volume) cite legal or moral concerns. Other problems are technical. Many network providers, having set up their systems with the idea that users would spend most of their time downloading data from central servers, have economic objections to peer-to-peer models. Some have begun to cut off access to peer-to-peer services on the basis that they violate user agreements and

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consume too much bandwidth (for illicit purposes, at that). As reported by the online News.com site, a third of U.S. colleges surveyed have banned Napster because students using it have sometimes saturated campus networks.

In our own company, Popular Power, we have encountered many of these problems as we create a peer-to-peer distributed computing resource out of millions of computers all over the Internet. We have identified many specific problems where the Internet architecture has been strained; we have also found workarounds for many of these problems and have come to understand what true solutions would be like. Surprisingly, we often find ourselves looking back to the Internet of 10 or 15 years ago to consider how best to solve a problem.

The original Internet was fundamentally designed as a peer-to-peer system. Over time it has become increasingly client/server, with millions of consumer clients communicating with a relatively privileged set of servers. The current crop of peer-to-peer applications is using the Internet much as it was originally designed: as a medium for communication for machines that share resources with each other as equals. Because this network model is more revolutionary for its scale and its particular implementations than for its concept, a good number of past Internet applications can provide lessons to architects of new peer-to-peer applications. In some cases, designers of current applications can learn from distributed Internet systems like Usenet and the Domain Name System (DNS); in others, the changes that the Internet has undergone during its commercialization may need to be reversed or modified to accommodate new peer-to-peer applications. In either case, the lessons these systems provide are instructive, and may help us, as application designers, avoid causing the death of the Internet.*

A revisionist history of peer-to-peer (1969–1995)

The Internet as originally conceived in the late 1960s was a peer-to-peer system. The goal of the original ARPANET was to share computing resources around the U.S. The challenge for this effort was to integrate different kinds of existing networks as well as future technologies with one common network architecture that would allow every host to be an equal player. The first few hosts on the ARPANET—UCLA, SRI, UCSB, and the University of Utah—were already independent computing sites with equal status. The ARPANET connected them together not in a master/slave or client/server relationship, but rather as equal computing peers.

* The authors wish to thank Debbie Pfeifer for invaluable help in editing this chapter.

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The early Internet was also much more open and free than today's network. Firewalls were unknown until the late 1980s. Generally, any two machines on the Internet could send packets to each other. The Net was the playground of cooperative researchers who generally did not need protection from each other. The protocols and systems were obscure and specialized enough that security break-ins were rare and generally harmless. As we shall see later, the modern Internet is much more partitioned.

The early "killer apps" of the Internet, FTP and Telnet, were themselves client/server applications. A Telnet client logged into a compute server, and an FTP client sent and received files from a file server. But while a single application was client/server, the usage patterns as a whole were symmetric. Every host on the Net could FTP or Telnet to any other host, and in the early days of minicomputers and mainframes, the servers usually acted as clients as well.

This fundamental symmetry is what made the Internet so radical. In turn, it enabled a variety of more complex systems such as Usenet and DNS that used peer-to-peer communication patterns in an interesting fashion. In subsequent years, the Internet has become more and more restricted to client/server-type applications. But as peer-to-peer applications become common again, we believe the Internet must revert to its initial design.

Let's look at two long-established fixtures of computer networking that include important peer-to-peer components: Usenet and DNS.

Usenet

Usenet news implements a decentralized model of control that in some ways is the grandfather of today's new peer-to-peer applications such as Gnutella and Freenet. Fundamentally, Usenet is a system that, using no central control, copies files between computers. Since Usenet has been around since 1979, it offers a number of lessons and is worth considering for contemporary file-sharing applications.

The Usenet system was originally based on a facility called the Unix-to-Unix-copy protocol, or UUCP. UUCP was a mechanism by which one Unix machine would automatically dial another, exchange files with it, and disconnect. This mechanism allowed Unix sites to exchange email, files, system patches, or other messages. The Usenet used UUCP to exchange messages within a set of topics, so that students at the University of North Carolina and Duke University could each "post" messages to a topic, read messages from others on the same topic, and trade messages between the two schools. The Usenet grew from these original two hosts to hundreds of thousands of sites. As the network grew, so did the

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number and structure of the topics in which a message could be posted. Usenet today uses a TCP/IP-based protocol known as the Network News Transport Protocol (NNTP), which allows two machines on the Usenet network to discover new newsgroups efficiently and exchange new messages in each group.

The basic model of Usenet provides a great deal of local control and relatively simple administration. A Usenet site joins the rest of the world by setting up a news exchange connection with at least one other news server on the Usenet network. Today, exchange is typically provided by a company's ISP. The administrator tells the company's news server to get in touch with the ISP's news server and exchange messages on a regular schedule. Company employees contact the company's local news server, and transact with it to read and post news messages. When a user in the company posts a new message in a newsgroup, the next time the company news server contacts the ISP's server it will notify the ISP's server that it has a new article and then transmit that article. At the same time, the ISP's server sends its new articles to the company's server.

Today, the volume of Usenet traffic is enormous, and not every server will want to carry the full complement of newsgroups or messages. The company administrator can control the size of the news installation by specifying which newsgroups the server will carry. In addition, the administrator can specify an expiration time by group or hierarchy, so that articles in a newsgroup will be retained for that time period but no longer. These controls allow each organization to voluntarily join the network on its own terms. Many organizations decide not to carry newsgroups that transmit sexually oriented or illegal material. This is a distinct difference from, say, Freenet, which (as a design choice) does not let a user know what material he or she has received.

Usenet has evolved some of the best examples of decentralized control structures on the Net. There is no central authority that controls the news system. The addition of new newsgroups to the main topic hierarchy is controlled by a rigorous democratic process, using the Usenet group *news.admin* to propose and discuss the creation of new groups. After a new group is proposed and discussed for a set period of time, anyone with an email address may submit an email vote for or against the proposal. If a newsgroup vote passes, a new group message is sent and propagated through the Usenet network.

There is even an institutionalized form of anarchy, the *alt.** hierarchy, that subverts the *news.admin* process in a codified way. An *alt* newsgroup can be added at any time by anybody, but sites that don't want to deal with the resulting absurdity can avoid the whole hierarchy. The beauty of Usenet is that each of the participating hosts can set their own local policies, but the network as a

Microsoft Exhibit 1019

whole functions through the cooperation and good will of the community. Many of the peer-to-peer systems currently emerging have not yet effectively addressed decentralized control as a goal. Others, such as Freenet, deliberately avoid giving local administrators control over the content of their machines because this control would weaken the political aims of the system. In each case, the interesting question is: how much control can or should the local administrator have?

NNTP as a protocol contains a number of optimizations that modern peer-to-peer systems would do well to copy. For instance, news messages maintain a "Path" header that traces their transmission from one news server to another. If news server A receives a request from server B, and A's copy of a message lists B in the Path header, A will not try to retransmit that message to B. Since the purpose of NNTP transmission is to make sure every news server on Usenet can receive an article (if it wants to), the Path header avoids a flood of repeated messages. Gnutella, as an example, does not use a similar system when transmitting search requests, so as a result a single Gnutella node can receive the same request repeatedly.

The open, decentralized nature of Usenet can be harmful as well as beneficial. Usenet has been enormously successful as a system in the sense that it has survived since 1979 and continues to be home to thriving communities of experts. It has swelled far beyond its modest beginnings. But in many ways the trusting, decentralized nature of the protocol has reduced its utility and made it an extremely noisy communication channel. Particularly, as we will discuss later, Usenet fell victim to spam early in the rise of the commercial Internet. Still, Usenet's systems for decentralized control, its methods of avoiding a network flood, and other characteristics make it an excellent object lesson for designers of peer-to-peer systems.

DNS

The Domain Name System (DNS) is an example of a system that blends peer-to-peer networking with a hierarchical model of information ownership. The remarkable thing about DNS is how well it has scaled, from the few thousand hosts it was originally designed to support in 1983 to the hundreds of millions of hosts currently on the Internet. The lessons from DNS are directly applicable to contemporary peer-to-peer data sharing applications.

DNS was established as a solution to a file-sharing problem. In the early days of the Internet, the way to map a human-friendly name like *bbn* to an IP address like *4.2.49.2* was through a single flat file, *hosts.txt*, which was copied around the Internet periodically. As the Net grew to thousands of hosts and managing that

Microsoft Exhibit 1019

file became impossible, DNS was developed as a way to distribute the data sharing across the peer-to-peer Internet.

The namespace of DNS names is naturally hierarchical. For example, O'Reilly & Associates, Inc. owns the namespace *oreilly.com*: they are the sole authority for all names in their domain, such as *www.oreilly.com*. This built-in hierarchy yields a simple, natural way to delegate responsibility for serving part of the DNS database. Each domain has an *authority*, the name server of record for hosts in that domain. When a host on the Internet wants to know the address of a given name, it queries its nearest name server to ask for the address. If that server does not know the name, it delegates the query to the authority for that namespace. That query, in turn, may be delegated to a higher authority, all the way up to the root name servers for the Internet as a whole. As the answer propagates back down to the requestor, the result is cached along the way to the name servers so the next fetch can be more efficient. Name servers operate both as clients and as servers.

DNS as a whole works amazingly well, having scaled to 10,000 times its original size. There are several key design elements in DNS that are replicated in many distributed systems today. One element is that hosts can operate both as clients and as servers, propagating requests when need be. These hosts help make the network scale well by caching replies. The second element is a natural method of propagating data requests across the network. Any DNS server can query any other, but in normal operation there is a standard path up the chain of authority. The load is naturally distributed across the DNS network, so that any individual name server needs to serve only the needs of its clients and the namespace it individually manages.

So from its earliest stages, the Internet was built out of peer-to-peer communication patterns. One advantage of this history is that we have experience to draw from in how to design new peer-to-peer systems. The problems faced today by new peer-to-peer applications systems such as file sharing are quite similar to the problems that Usenet and DNS addressed 10 or 15 years ago.

The network model of the Internet explosion (1995–1999)

The explosion of the Internet in 1994 radically changed the shape of the Internet, turning it from a quiet geek utopia into a bustling mass medium. Millions of new people flocked to the Net. This wave represented a new kind of people—ordinary folks who were interested in the Internet as a way to send email, view

Microsoft Exhibit 1019

web pages, and buy things, not computer scientists interested in the details of complex computer networks. The change of the Internet to a mass cultural phenomenon has had a far-reaching impact on the network architecture, an impact that directly affects our ability to create peer-to-peer applications in today's Internet. These changes are seen in the way we use the network, the breakdown of cooperation on the Net, the increasing deployment of firewalls on the Net, and the growth of asymmetric network links such as ADSL and cable modems.

The switch to client/server

The network model of user applications—not just their consumption of bandwidth, but also their methods of addressing and communicating with other machines—changed significantly with the rise of the commercial Internet and the advent of millions of home users in the 1990s. Modem connection protocols such as SLIP and PPP became more common, typical applications targeted slow-speed analog modems, and corporations began to manage their networks with firewalls and Network Address Translation (NAT). Many of these changes were built around the usage patterns common at the time, most of which involved downloading data, not publishing or uploading information.

The web browser, and many of the other applications that sprung up during the early commercialization of the Internet, were based around a simple client/server protocol: the client initiates a connection to a well-known server, downloads some data, and disconnects. When the user is finished with the data retrieved, the process is repeated. The model is simple and straightforward. It works for everything from browsing the Web to watching streaming video, and developers cram shopping carts, stock transactions, interactive games, and a host of other things into it. The machine running a web client doesn't need to have a permanent or well-known address. It doesn't need a continuous connection to the Internet. It doesn't need to accommodate multiple users. It just needs to know how to ask a question and listen for a response.

Not all of the applications used at home fit this model. Email, for instance, requires much more two-way communication between an email client and server. In these cases, though, the client is often talking to a server on the local network (either the ISP's mail server or a corporate one). Chat systems that achieved widespread usage, such as AOL's Instant Messenger, have similar "local" properties, and Usenet systems do as well. As a result, the typical ISP configuration instructions give detailed (and often misunderstood) instructions for email, news, and sometimes chat. These were the exceptions that were worth some manual configuration on the user's part. The "download" model is simpler

Microsoft Exhibit 1019

and works without much configuration; the "two-way" model is used less frequently but perhaps to greater effect.

While early visions of the Web always called it a great equalizer of communications—a system that allowed every user to publish their viewpoints rather than simply consume media—the commercial explosion on the Internet quickly fit the majority of traffic into the downstream paradigm already used by television and newspapers. Architects of the systems that enabled the commercial expansion of the Net often took this model into account, assuming that it was here to stay. Peer-to-peer applications may require these systems to change.

The breakdown of cooperation

The early Internet was designed on principles of cooperation and good engineering. Everyone working on Internet design had the same goal: build a reliable, efficient, powerful network. As the Internet entered its current commercial phase, the incentive structures changed, resulting in a series of stresses that have highlighted the Internet's susceptibility to the tragedy of the commons. This phenomenon has shown itself in many ways, particularly the rise of spam on the Internet and the challenges of building efficient network protocols that correctly manage the common resource.

Spam: Uncooperative people

Spam, or unsolicited commercial messages, is now an everyday occurrence on the Internet. Back in the pre-commercial network, however, unsolicited advertisements were met with surprise and outrage. The end of innocence occurred on April 12, 1994, the day the infamous Canter and Seigel "green card spam" appeared on the Usenet. Their offense was an advertisement posted individually to every Usenet newsgroup, blanketing the whole world with a message advertising their services. At the time, this kind of action was unprecedented and engendered strong disapproval. Not only were most of the audience uninterested in the service, but many people felt that Canter and Seigel had stolen the Usenet's resources. The advertisers did not pay for the transmission of the advertisement; instead the costs were born by the Usenet as a whole.

In the contemporary Internet, spam does not seem surprising; Usenet has largely been given over to it, and ISPs now provide spam filtering services for their users' email both to help their users and in self-defense. Email and Usenet relied on individuals' cooperation to not flood the commons with junk mail, and that cooperation broke down. Today the Internet generally lacks effective technology to prevent spam.

Microsoft Exhibit 1019

The problem is the lack of accountability in the Internet architecture. Because any host can connect to any other host, and because connections are nearly anonymous, people can insert spam into the network at any point. There has been an arms race of trying to hold people accountable—closing down open sendmail relays, tracking sources of spam on Usenet, retaliation against spammers—but the battle has been lost, and today we have all learned to live with spam.

The lesson for peer-to-peer designers is that without accountability in a network, it is difficult to enforce rules of social responsibility. Just like Usenet and email, today's peer-to-peer systems run the risk of being overrun by unsolicited advertisements. It is difficult to design a system where socially inappropriate use is prevented. Technologies for accountability, such as cryptographic identification or reputation systems, can be valuable tools to help manage a peer-to-peer network. There have been proposals to retrofit these capabilities into Usenet and email, but none today are widespread; it is important to build these capabilities into the system from the beginning. Chapter 16, *Accountability*, discusses some techniques for controlling spam, but these are still arcane.

The TCP rate equation: Cooperative protocols

A fundamental design principle of the Internet is *best effort* packet delivery. "Best effort" means the Internet does not guarantee that a packet will get through, simply that the Net will do its best to get the packet to the destination. Higher-level protocols such as TCP create reliable connections by detecting when a packet gets lost and resending it. A major reason packets do not get delivered on the Internet is congestion: if a router in the network is overwhelmed, it will start dropping packets at random. TCP accounts for this by throttling the speed at which it sends data. When the network is congested, each individual TCP connection independently slows down, seeking to find the optimal rate while not losing too many packets. But not only do individual TCP connections optimize their bandwidth usage, TCP is also designed to make the Internet as a whole operate efficiently. The collective behavior of many individual TCP connections backing off independently results in a lessening of the congestion at the router, in a way that is exquisitely tuned to use the router's capacity efficiently. In essence, the TCP backoff algorithm is a way for individual peers to manage a shared resource without a central coordinator.

The problem is that the efficiency of TCP on the Internet scale fundamentally requires cooperation: each network user has to play by the same rules. The performance of an individual TCP connection is inversely proportional to the

Microsoft Exhibit 1019

square root of the packet loss rate—part of the "TCP rate equation," a fundamental governing law of the Internet. Protocols that follow this law are known as "TCP-friendly protocols." It is possible to design other protocols that do not follow the TCP rate equation, ones that rudely try to consume more bandwidth than they should. Such protocols can wreak havoc on the Net, not only using more than their fair share but actually spoiling the common resource for all. This abstract networking problem is a classic example of a tragedy of the commons, and the Internet today is quite vulnerable to it.

The problem is not only theoretical, it is also quite practical. As protocols have been built in the past few years by companies with commercial demands, there has been growing concern that unfriendly protocols will begin to hurt the Internet.

An early example was a feature added by Netscape to their browser—the ability to download several files at the same time. The Netscape engineers discovered that if you downloaded embedded images in parallel, rather than one at a time, the whole page would load faster and users would be happier. But there was a question: was this usage of bandwidth fair? Not only does it tax the server to have to send out more images simultaneously, but it creates more TCP channels and sidesteps TCP's congestion algorithms. There was some controversy about this feature when Netscape first introduced it, a debate quelled only after Netscape released the client and people discovered in practice that the parallel download strategy did not unduly harm the Internet. Today this technique is standard in all browsers and goes unquestioned. The questions have reemerged at the new frontier of "download accelerator" programs that download different chunks of the same file simultaneously, again threatening to upset the delicate management of Internet congestion.

A more troubling concern about congestion management is the growth of bandwidth-hungry streaming broadband media. Typical streaming media applications do *not* use TCP, instead favoring custom UDP-based protocols with their own congestion control and failure handling strategies. Many of these protocols are proprietary; network engineers do not even have access to their implementations to examine if they are TCP-friendly. So far there has been no major problem. The streaming media vendors seem to be playing by the rules, and all is well. But fundamentally the system is brittle, and either through a mistake or through greed the Internet's current delicate cooperation could be toppled.

What do spam and the TCP rate algorithm have in common? They both demonstrate that the proper operation of the Internet is fragile and requires the cooperation of everyone involved. In the case of TCP, the system has mostly worked and the network has been preserved. In the case of spam, however, the battle has been lost and unsocial behavior is with us forever. The lesson for peer-to-

Microsoft Exhibit 1019

peer system designers is to consider the issue of polite behavior up front. Either we must design systems that do not require cooperation to function correctly, or we must create incentives for cooperation by rewarding proper behavior or auditing usage so that misbehavior can be punished.

Firewalls, dynamic IP, NAT: The end of the open network

At the same time that the cooperative nature of the Internet was being threatened, network administrators implemented a variety of management measures that resulted in the Internet being a much less open network. In the early days of the Internet, all hosts were equal participants. The network was symmetric—if a host could reach the Net, everyone on the Net could reach that host. Every computer could equally be a client and a server. This capability began to erode in the mid-1990s with the deployment of firewalls, the rise of dynamic IP addresses, and the popularity of Network Address Translation (NAT).

As the Internet matured there came a need to secure the network, to protect individual hosts from unlimited access. By default, any host that can access the Internet can also *be accessed* on the Internet. Since average users could not handle the security risks that resulted from a symmetric design, network managers turned to firewalls as a tool to control access to their machines.

Firewalls stand at the gateway between the internal network and the Internet outside. They filter packets, choosing which traffic to let through and which to deny. A firewall changes the fundamental Internet model: some parts of the network cannot fully talk to other parts. Firewalls are a very useful security tool, but they pose a serious obstacle to peer-to-peer communication models.

A typical firewall works by allowing anyone inside the internal network to initiate a connection to anyone on the Internet, but it prevents random hosts on the Internet from initiating connections to hosts in the internal network. This kind of firewall is like a one-way gate: you can go out, but you cannot come in. A host protected in this way cannot easily function as a server; it can only be a client. In addition, outgoing connections may be restricted to certain applications like FTP and the Web by blocking traffic to certain ports at the firewall.

Allowing an Internet host to be only a client, not a server, is a theme that runs through a lot of the changes in the Internet after the consumer explosion. With the rise of modem users connecting to the Internet, the old practice of giving every Internet host a fixed IP address became impractical, because there were not enough IP addresses to go around. Dynamic IP address assignment is now the norm for many hosts on the Internet, where an individual computer's

Microsoft Exhibit 1019

address may change every single day. Broadband providers are even finding dynamic IP useful for their "always on" services. The end result is that many hosts on the Internet are not easily reachable, because they keep moving around. Peer-to-peer applications such as instant messaging or file sharing have to work hard to circumvent this problem, building dynamic directories of hosts. In the early Internet, where hosts remained static, it was much simpler.

A final trend is to not even give a host a valid public Internet address at all, but instead to use NAT to hide the address of a host behind a firewall. NAT combines the problems of firewalls and dynamic IP addresses: not only is the host's true address unstable, it is not even reachable! All communication has to go through a fairly simple pattern that the NAT router can understand, resulting in a great loss of flexibility in applications communications. For example, many cooperative Internet games have trouble with NAT: every player in the game wants to be able to contact every other player, but the packets cannot get through the NAT router. The result is that a central server on the Internet has to act as an application-level message router, emulating the function that TCP/IP itself used to serve.

Firewalls, dynamic IP, and NAT grew out of a clear need in Internet architecture to make scalable, secure systems. They solved the problem of bringing millions of client computers onto the Internet quickly and manageably. But these same technologies have weakened the Internet infrastructure as a whole, relegating most computers to second-class status as clients only. New peer-to-peer applications challenge this architecture, demanding that participants serve resources as well as use them. As peer-to-peer applications become more common, there will be a need for common technical solutions to these problems.

Asymmetric bandwidth

A final Internet trend of the late 1990s that presents a challenge to peer-to-peer applications is the rise in asymmetric network connections such as ADSL and cable modems. In order to get the most efficiency out of available wiring, current broadband providers have chosen to provide asymmetric bandwidth. A typical ADSL or cable modem installation offers three to eight times more bandwidth when getting data from the Internet than when sending data to it, favoring client over server usage.

The reason this has been tolerated by most users is clear: the Web is the killer app for the Internet, and most users are only clients of the Web, not servers. Even users who publish their own web pages typically do not do so from a home broadband connection, but instead use third-party dedicated servers pro-

Microsoft Exhibit 1019

vided by companies like GeoCities or Exodus. In the early days of the Web it was not clear how this was going to work: could each user have a personal web server? But in the end most Web use is itself asymmetric—many clients, few servers—and most users are well served by asymmetric bandwidth.

The problem today is that peer-to-peer applications are changing the assumption that end users only want to download from the Internet, never upload to it. File-sharing applications such as Napster or Gnutella can reverse the bandwidth usage, making a machine serve many more files than it downloads. The upstream pipe cannot meet demand. Even worse, because of the details of TCP's rate control, if the upstream path is clogged, the downstream performance suffers as well. So if a computer is serving files on the slow side of a link, it cannot easily download simultaneously on the fast side.

ADSL and cable modems assume asymmetric bandwidth for an individual user. This assumption takes hold even more strongly inside ISP networks, which are engineered for bits to flow to the users, not from them. The end result is a network infrastructure that is optimized for computers that are only clients, not servers. But peer-to-peer technology generally makes every host act both as a client and a server; the asymmetric assumption is incorrect. There is not much an individual peer-to-peer application can do to work around asymmetric bandwidth; as peer-to-peer applications become more widespread, the network architecture is going to have to change to better handle the new traffic patterns.

Observations on the current crop of peer-to-peer applications (2000)

While the new breed of peer-to-peer applications can take lessons from earlier models, these applications also introduce new characteristics or features that are novel. Peer-to-peer allows us to separate the concepts of authoring information and publishing that same information. Peer-to-peer allows for decentralized application design, something that is both an opportunity and a challenge. And peer-to-peer applications place unique strains on firewalls, something well demonstrated by the current trend to use the HTTP port for operations other than web transactions.

Authoring is not the same as publishing

One of the promises of the Internet is that people are able to be their own publishers, for example, by using personal web sites to make their views and interests known. Self-publishing has certainly become more common with the

Microsoft Exhibit 1019

commercialization of the Internet. More often, however, users spend most of their time reading (downloading) information and less time publishing, and as discussed previously, commercial providers of Internet access have structured their offering around this asymmetry.

The example of Napster creates an interesting middle ground between the ideal of "everyone publishes" and the seeming reality of "everyone consumes." Napster particularly (and famously) makes it very easy to *publish* data you did not *author*. In effect, your machine is being used as a repeater to retransmit data once it reaches you. A network designer, assuming that there are only so many authors in the world and therefore that asymmetric broadband is the perfect optimization, is confounded by this development. This is why many networks such as college campuses have banned Napster from use.

Napster changes the flow of data. The assumptions that servers would be owned by publishers and that publishers and authors would combine into a single network location have proven untrue. The same observation also applies to Gnutella, Freenet, and others. Users don't need to create content in order to want to publish it—in fact, the benefits of publication by the "reader" have been demonstrated by the scale some of these systems have been able to reach.

Decentralization

Peer-to-peer systems seem to go hand-in-hand with decentralized systems. In a fully decentralized system, not only is every host an equal participant, but there are *no* hosts with special facilitating or administrative roles. In practice, building fully decentralized systems can be difficult, and many peer-to-peer applications take hybrid approaches to solving problems. As we have already seen, DNS is peer-to-peer in protocol design but with a built-in sense of hierarchy. There are many other examples of systems that are peer-to-peer at the core and yet have some semi-centralized organization in application, such as Usenet, instant messaging, and Napster.

Usenet is an instructive example of the evolution of a decentralized system. Usenet propagation is symmetric: hosts share traffic. But because of the high cost of keeping a full news feed, in practice there is a backbone of hosts that carry all of the traffic and serve it to a large number of "leaf nodes" whose role is mostly to receive articles. Within Usenet, there was a natural trend toward making traffic propagation hierarchical, even though the underlying protocols do not demand it. This form of "soft centralization" may prove to be economic for many peer-to-peer systems with high-cost data transmission.

Microsoft Exhibit 1019

Many other current peer-to-peer applications present a decentralized face while relying on a central facilitator to coordinate operations. To a user of an instant messaging system, the application appears peer-to-peer, sending data directly to the friend being messaged. But all major instant messaging systems have some sort of server on the back end that facilitates nodes talking to each other. The server maintains an association between the user's name and his or her current IP address, buffers messages in case the user is offline, and routes messages to users behind firewalls. Some systems (such as ICQ) allow direct client-to-client communication when possible but have a server as a fallback. A fully decentralized approach to instant messaging would not work on today's Internet, but there are scaling advantages to allowing client-to-client communication when possible.

Napster is another example of a hybrid system. Napster's file sharing is decentralized: one Napster client downloads a file directly from another Napster client's machine. But the directory of files is centralized, with the Napster servers answering search queries and brokering client connections. This hybrid approach seems to scale well: the directory can be made efficient and uses low bandwidth, and the file sharing can happen on the edges of the network.

In practice, some applications might work better with a fully centralized design, not using any peer-to-peer technology at all. One example is a search on a large, relatively static database. Current web search engines are able to serve up to one billion pages all from a single place. Search algorithms have been highly optimized for centralized operation; there appears to be little benefit to spreading the search operation out on a peer-to-peer network (database generation, however, is another matter).

Also, applications that require centralized information sharing for accountability or correctness are hard to spread out on a decentralized network. For example, an auction site needs to guarantee that the best price wins; that can be difficult if the bidding process has been spread across many locations. Decentralization engenders a whole new area of network-related failures: unreliability, incorrect data synchronization, etc. Peer-to-peer designers need to balance the power of peer-to-peer models against the complications and limitations of decentralized systems.

Abusing port 80

One of the stranger phenomena in the current Internet is the abuse of port 80, the port that HTTP traffic uses when people browse the Web. Firewalls typically filter traffic based on the direction of traffic (incoming or outgoing) and the

Microsoft Exhibit 1019

destination port of the traffic. Because the Web is a primary application of many Internet users, almost all firewalls allow outgoing connections on port 80 even if the firewall policy is otherwise very restrictive.

In the early days of the Internet, the port number usually indicated which application was using the network; the firewall could count on port 80 being only for Web traffic. But precisely because many firewalls allow connections to port 80, other application authors started routing traffic through that port. Streaming audio, instant messaging, remote method invocations, even whole mobile agents are being sent through port 80. Most current peer-to-peer applications have some way to use port 80 as well in order to circumvent network security policies. Naive firewalls are none the wiser; they are unaware that they are passing the exact sorts of traffic the network administrator intended to block.

The problem is twofold. First, there is no good way for a firewall to identify what applications are running through it. The port number has already been circumvented. Fancier firewalls can analyze the actual traffic going through the firewall and see if it is a legitimate HTTP stream, but that just encourages application designers to masquerade as HTTP, leading to an escalating arms race that benefits no one.

The second problem is that even if an application has a legitimate reason to go through the firewall, there is no simple way for the application to request permission. The firewall, as a network security measure, is outmoded. As long as a firewall allows some sort of traffic through, peer-to-peer applications will find a way to slip through that opening.

Peer-to-peer prescriptions (2001–?)

The story is clear: The Internet was designed with peer-to-peer applications in mind, but as it has grown the network has become more asymmetric. What can we do to permit new peer-to-peer applications to flourish while respecting the pressures that have shaped the Internet to date?

Technical solutions: Return to the old Internet

As we have seen, the explosion of the Internet into the consumer space brought with it changes that have made it difficult to do peer-to-peer networking. Firewalls make it hard to contact hosts; dynamic IP and NAT make it nearly impossible. Asymmetric bandwidth is holding users back from efficiently serving files on their systems. Current peer-to-peer applications generally would benefit from an Internet more like the original network, where these restrictions were not in

Microsoft Exhibit 1019

place. How can we enable peer-to-peer applications to work better with the current technological situation?

Firewalls serve an important need: they allow administrators to express and enforce policies about the use of their networks. That need will not change with peer-to-peer applications. Neither application designers nor network security administrators are benefiting from the current state of affairs. The solution lies in making firewalls smarter so that peer-to-peer applications can cooperate with the firewall to allow traffic the administrator wants. Firewalls must become more sophisticated, allowing systems behind the firewall to ask permission to run a particular peer-to-peer application. Peer-to-peer designers must contribute to this design discussion, then enable their applications to use these mechanisms. There is a good start to this solution in the SOCKS protocol, but it needs to be expanded to be more flexible and more tied toward applications rather than simple port numbers.

The problems engendered by dynamic IP and NAT already have a technical solution: IPv6. This new version of IP, the next generation Internet protocol architecture, has a 128-bit address space—enough for every host on the Internet to have a permanent address. Eliminating address scarcity means that every host has a home and, in theory, can be reached. The main thing holding up the deployment of IPv6 is the complexity of the changeover. At this stage, it remains to be seen when or even if IPv6 will be commonly deployed, but without it peer-to-peer applications will continue to need to build alternate address spaces to work around the limitations set by NAT and dynamic IP.

Peer-to-peer applications stress the bandwidth usage of the current Internet. First, they break the assumption of asymmetry upon which today's ADSL and cable modem providers rely. There is no simple way that peer-to-peer applications can work around this problem; we simply must encourage broadband connections to catch up.

However, peer-to-peer applications can do several things to use the existing bandwidth more efficiently. First, data caching is a natural optimization for any peer-to-peer application that is transmitting bulk data; it would be a significant advance to make sure that a program does not have to retransmit or resend data to another host. Caching is a well understood technology: distributed caches like Squid have worked out many of the consistency and load sharing issues that peer-to-peer applications face.

Second, a peer-to-peer application must have effective means for allowing users to control the bandwidth the application uses. If I run a Gnutella node at home, I want to specify that it can use only 50% of my bandwidth. Current operating

Microsoft Exhibit 1019

systems and programming libraries do not provide good tools for this kind of limitation, but as peer-to-peer applications start demanding more network resources from hosts, users will need tools to control that resource usage.

Social solutions: Engineer polite behavior

Technical measures can help create better peer-to-peer applications, but good system design can also yield social stability. A key challenge in creating peer-to-peer systems is to have a mechanism of accountability and the enforcement of community standards. Usenet breaks down because it is impossible to hold people accountable for their actions. If a system has a way to identify individuals (even pseudonymously, to preserve privacy), that system can be made more secure against antisocial behavior. Reputation tracking mechanisms, discussed in Chapter 16, and in Chapter 17, *Reputation*, are valuable tools here as well, to give the user community a collective memory about the behavior of individuals.

Peer-to-peer systems also present the challenge of integrating local administrative control with global system correctness. Usenet was successful at this goal. The local news administrator sets policy for his or her own site, allowing the application to be customized to each user group's needs. The shared communication channel of *news.admin* allows a community governance procedure for the entire Usenet community. These mechanisms of local and global control were built into Usenet from the beginning, setting the rules of correct behavior. New breed peer-to-peer applications should follow this lead, building in their own social expectations.

Conclusions

The Internet started out as a fully symmetric, peer-to-peer network of cooperating users. As the Net has grown to accommodate the millions of people flocking online, technologies have been put in place that have split the Net up into a system with relatively few servers and many clients. At the same time, some of the basic expectations of cooperation are showing the risk of breaking down, threatening the structure of the Net.

These phenomena pose challenges and obstacles to peer-to-peer applications: both the network and the applications have to be designed together to work in tandem. Application authors must design robust applications that can function in the complex Internet environment, and network designers must build in capabilities to handle new peer-to-peer applications. Fortunately, many of these issues are familiar from the experience of the early Internet; the lessons learned there can be brought forward to design tomorrow's systems.

Microsoft Exhibit 1019

chapter two

Listening to Napster

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Premature definition is a danger for any movement. Once a definitive label is applied to a new phenomenon, it invariably begins shaping—and possibly distorting—people's views. So it is with the present movement toward decentralized applications. After a year or so of attempting to describe the revolution in file sharing and related technologies, we have finally settled on *peer-to-peer* as a label for what's happening.*

Somehow, though, this label hasn't clarified things. Instead, it's distracted us from the phenomena that first excited us. Taken literally, servers talking to one another are peer-to-peer. The game Doom is peer-to-peer. There are even people applying the label to email and telephones. Meanwhile, Napster, which jump-started the conversation, is not peer-to-peer in the strictest sense, because it uses a centralized server to store pointers and resolve addresses.

If we treat peer-to-peer as a literal definition of what's happening, we end up with a phrase that describes Doom but not Napster and suggests that Alexander Graham Bell is a peer-to-peer engineer but Shawn Fanning is not. Eliminating Napster from the canon now that we have a definition we can apply literally is like saying, "Sure, it may work in practice, but it will never fly in theory."

This literal approach to peer-to-peer is plainly not helping us understand what makes it important. Merely having computers act as peers on the Internet is hardly novel. From the early days of PDP-11s and Vaxes to the Sun SPARCs and Windows 2000 systems of today, computers on the Internet have been peering

* Thanks to *Business 2.0*, where many of these ideas first appeared, and to Dan Gillmor of the *San Jose Mercury News*, for first pointing out the important relationship between P2P and the Domain Name System.

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with each other. So peer-to-peer architecture itself can't be the explanation for the recent changes in Internet use.

What have changed are the nodes that make up these peer-to-peer systems—Internet-connected PCs, which formerly were relegated to being nothing but clients—and where these nodes are: at the edges of the Internet, cut off from the DNS (Domain Name System) because they have no fixed IP addresses.

Resource-centric addressing for unstable environments

Peer-to-peer is a class of applications that takes advantage of resources—storage, cycles, content, human presence—available at the edges of the Internet. Because accessing these decentralized resources means operating in an environment of unstable connectivity and unpredictable IP addresses, peer-to-peer nodes must operate outside the DNS and have significant or total autonomy from central servers.

That's it. That's what makes peer-to-peer distinctive.

Note that this isn't what makes peer-to-peer important. It's not the problem designers of peer-to-peer systems set out to solve, like aggregating CPU cycles, sharing files, or chatting. But it's a problem they all had to solve to get where they wanted to go.

What makes Napster and Popular Power and Freenet and AIMster and Groove similar is that they are all leveraging previously unused resources, by tolerating and even working with variable connectivity. This lets them make new, powerful use of the hundreds of millions of devices that have been connected to the edges of the Internet in the last few years.

One could argue that the need for peer-to-peer designers to solve connectivity problems is little more than an accident of history. But improving the way computers connect to one another was the rationale behind the 1984 design of the Internet Protocol (IP), and before that DNS, and before that the Transmission Control Protocol (TCP), and before that the Net itself. The Internet is made of such frozen accidents.

So if you're looking for a litmus test for peer-to-peer, this is it:

1. Does it allow for variable connectivity and temporary network addresses?
2. Does it give the nodes at the edges of the network significant autonomy?

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If the answer to both of those questions is yes, the application is peer-to-peer. If the answer to either question is no, it's not peer-to-peer.

Another way to examine this distinction is to think about ownership. Instead of asking, "Can the nodes speak to one another?" ask, "Who owns the hardware that the service runs on?" The huge preponderance of the hardware that makes Yahoo! work is owned by Yahoo! and managed in Santa Clara. The huge preponderance of the hardware that makes Napster work is owned by Napster users and managed on tens of millions of individual desktops. Peer-to-peer is a way of decentralizing not just features, but costs and administration as well.

Peer-to-peer is as peer-to-peer does

Up until 1994, the Internet had one basic model of connectivity. Machines were assumed to be always on, always connected, and assigned permanent IP addresses. DNS was designed for this environment, in which a change in IP address was assumed to be abnormal and rare, and could take days to propagate through the system.

With the invention of Mosaic, another model began to spread. To run a web browser, a PC needed to be connected to the Internet over a modem, with its own IP address. This created a second class of connectivity, because PCs entered and left the network cloud frequently and unpredictably.

Furthermore, because there were not enough IP addresses available to handle the sudden demand caused by Mosaic, ISPs began to assign IP addresses dynamically. They gave each PC a different, possibly masked, IP address with each new session. This instability prevented PCs from having DNS entries, and therefore prevented PC users from hosting any data or applications that accepted connections from the Net.

For a few years, treating PCs as dumb but expensive clients worked well. PCs had never been designed to be part of the fabric of the Internet, and in the early days of the Web, the toy hardware and operating systems of the average PC made it an adequate life-support system for a browser but good for little else.

Over time, though, as hardware and software improved, the unused resources that existed behind this veil of second-class connectivity started to look like something worth getting at. At a conservative estimate—assuming only 100 million PCs among the Net's 300 million users, and only a 100 MHz chip and 100 MB drive on the average Net-connected PC—the world's Net-connected PCs presently host an aggregate 10 billion megahertz of processing power and 10 thousand terabytes of storage.

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The veil is pierced

The launch of ICQ, the first PC-based chat system, in 1996 marked the first time those intermittently connected PCs became directly addressable by average users. Faced with the challenge of establishing portable presence, ICQ bypassed DNS in favor of creating its own directory of protocol-specific addresses that could update IP addresses in real time, a trick followed by Groove, Napster, and NetMeeting as well. (Not all peer-to-peer systems use this trick. Gnutella and Freenet, for example, bypass DNS the old-fashioned way, by relying on numeric IP addresses. United Devices and SETI@home bypass it by giving the nodes scheduled times to contact fixed addresses, at which times they deliver their current IP addresses.)

A run of *whois* counts 23 million domain names, built up in the 16 years since the inception of IP addresses in 1984. Napster alone has created more than 23 million non-DNS addresses in 16 *months*, and when you add in all the non-DNS instant messaging addresses, the number of peer-to-peer addresses designed to reach dynamic IP addresses tops 200 million. Even if you assume that the average DNS host has 10 additional addresses of the form *foo.host.com*, the total number of peer-to-peer addresses now, after only 4 years, is of the same order of magnitude as the total number of DNS addresses, and is growing faster than the DNS universe today.

As new kinds of Net-connected devices like wireless PDAs and digital video recorders such as TiVo and Replay proliferate, they will doubtless become an important part of the Internet as well. But for now, PCs make up the enormous majority of these untapped resources. PCs are the dark matter of the Internet, and their underused resources are fueling peer-to-peer.

Real solutions to real problems

Why do we have unpredictable IP addresses in the first place? Because there weren't enough to go around when the Web happened. It's tempting to think that when enough new IP addresses are created, the old "One Device/One Address" regime will be restored, and the Net will return to its pre–peer-to-peer architecture.

This won't happen, though, because no matter how many new IP addresses there are, peer-to-peer systems often create addresses for things that aren't machines. Freenet and Mojo Nation create addresses for content intentionally spread across multiple computers. AOL Instant Messenger (AIM) and ICQ create names that refer to human beings and not machines. Peer-to-peer is designed to handle unpredictability, and nothing is more unpredictable than the humans

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who use the network. As the Net becomes more human-centered, the need for addressing schemes that tolerate and even expect temporary and unstable patterns of use will grow.

Who's in and who's out?

Napster is peer-to-peer because the addresses of Napster nodes bypass DNS, and because once the Napster server resolves the IP addresses of the PCs hosting a particular song, it shifts control of the file transfers to the nodes. Furthermore, the ability of the Napster nodes to host the songs without central intervention lets Napster users get access to several terabytes of storage and bandwidth at no additional cost.

However, Intel's "server peer-to-peer" is not peer-to-peer, because servers have always been peers. Their fixed IP addresses and permanent connections present no new problems, and calling what they already do "peer-to-peer" presents no new solutions.

ICQ and Jabber are peer-to-peer, because they not only devolve connection management to the individual nodes after resolving the addresses, but they also violate the machine-centric worldview encoded in DNS. Your address has nothing to do with the DNS hierarchy, or even with a particular machine, except temporarily; your chat address travels with you. Furthermore, by mapping "presence"—whether you are at your computer at any given moment in time—chat turns the old idea of permanent connectivity and IP addresses on its head. Transient connectivity is not an annoying hurdle in the case of chat but an important contribution of the technology.

Email, which treats variable connectivity as the norm, nevertheless fails the peer-to-peer definition test because your address is machine-dependent. If you drop AOL in favor of another ISP, your AOL email address disappears as well, because it hangs off DNS. Interestingly, in the early days of the Internet, there was a suggestion to make the part of the email address before the @ globally unique, linking email to a person rather than to a person@machine. That would have been peer-to-peer in the current sense, but it was rejected in favor of a machine-centric view of the Internet.

Popular Power is peer-to-peer, because the distributed clients that contact the server need no fixed IP address and have a high degree of autonomy in performing and reporting their calculations. They can even be offline for long stretches while still doing work for the Popular Power network.

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Dynamic DNS is not peer-to-peer, because it tries to retrofit PCs into traditional DNS.

And so on. This list of resources that current peer-to-peer systems take advantage of—storage, cycles, content, presence—is not necessarily complete. If there were some application that needed 30,000 separate video cards, or microphones, or speakers, a peer-to-peer system could be designed that used those resources as well.

Peer-to-peer is a horseless carriage

As with the "horseless" carriage or the "compact" disc, new technologies are often labeled according to some simple difference between them and what came before (horse-drawn carriages, non-compact records).

Calling this new class of applications peer-to-peer emphasizes their difference from the dominant client/server model. However, like the horselessness of the carriage or the compactness of the disc, the "peeriness" of peer-to-peer is more a label than a definition.

As we've learned from the history of the Internet, adoption is a better predictor of software longevity than elegant design. Users will not adopt peer-to-peer applications that embrace decentralization for decentralization's sake. Instead, they will adopt those applications that use just enough decentralization, in just the right way, to create novel functions or improve existing ones.

Follow the users

It seems obvious but bears repeating: Definitions are useful only as tools for sharpening one's perception of reality and improving one's ability to predict the future. Whatever one thinks of Napster's probable longevity, Napster is the killer app for this revolution.

If the Internet has taught technology watchers anything, it's that predictions of the future success of a particular software method or paradigm are of tenuous accuracy at best. Consider the history of "multimedia." If you had read almost any computer trade magazine or followed any technology analyst's predictions for the rise of multimedia in the early '90s, the future they predicted was one of top-down design, and this multimedia future was to be made up of professionally produced CD-ROMs and "walled garden" online services such as CompuServe and Delphi. And then the Web came along and let absolute amateurs build pages in HTML, a language that was laughably simple compared to the tools being developed for other multimedia services.

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Users reward simplicity

HTML's simplicity, which let amateurs create content for little cost and little invested time, turned out to be HTML's long suit. Between 1993 and 1995, HTML went from an unknown protocol to the preeminent tool for designing electronic interfaces, decisively displacing almost all challengers and upstaging CD-ROMs, as well as online services and a dozen expensive and abortive experiments with interactive TV—and it did this while having no coordinated authority, no central R&D effort, and no discernible financial incentive for the majority of its initial participants.

What caught the tech watchers in the industry by surprise was that HTML was made a success not by corporations but by users. The obvious limitations of the Web for professional designers blinded many to HTML's ability to allow average users to create multimedia content.

HTML spread because it allowed ordinary users to build their own web pages, without requiring that they be software developers or even particularly savvy software users. All the confident predictions about the CD-ROM-driven multimedia future turned out to be meaningless in the face of user preference. This in turn led to network effects on adoption: once a certain number of users had adopted it, there were more people committed to making the Web better than there were people committed to making CD-ROM authoring easier for amateurs.

The lesson of HTML's astonishing rise for anyone trying to make sense of the social aspects of technology is simple: *follow the users*. Understand the theory, study the engineering, but most importantly, follow the adoption rate. The cleanest theory and the best engineering in the world mean nothing if the users don't use them, and understanding why some solution will never work in theory means nothing if users adopt it all the same.

Listen to Napster

In the present circumstance, the message that comes from paying attention to the users is simple: *Listen to Napster*.

Listen to what the rise of Napster is saying about peer-to-peer, because as important as Groove or Freenet or OpenCOLA may become, Napster is already a mainstream phenomenon. Napster has had over 40 million client downloads at the time of this writing. Its adoption rate has outstripped NCSA Mosaic, Hotmail, and even ICQ, the pioneer of P2P addressing. Because Napster is what the users are actually spending their time using, the lessons we can take from Napster are

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still our best guide to the kind of things that are becoming possible with the rise of peer-to-peer architecture.

It's the applications, stupid

The first lesson Napster holds is that it was written to solve a problem—limitations on file copying—and the technological solutions it adopted were derived from the needs of the application, not vice versa.

The fact that the limitations on file copying are legal ones matters little to the technological lessons to be learned from Napster, because technology is often brought to bear to solve nontechnological problems. In this case, the problem Shawn Fanning, Napster's creator, set out to solve was a gap between what was possible with digital songs (endless copying at a vanishingly small cost) and what was legal. The willingness of the major labels to destroy any file copying system they could reach made the classic Web model of central storage of data impractical, meaning Napster had to find a non-Web-like solution.

Decentralization is a tool, not a goal

The primary fault of much of the current thinking about peer-to-peer lies in an "if we build it, they will come" mentality, where interesting technological challenges of decentralizing applications are assumed to be the only criterion that a peer-to-peer system needs to address in order to succeed. The enthusiasm for peer-to-peer has led to a lot of incautious statements about the superiority of peer-to-peer for many, and possibly most, classes of networked applications.

In fact, peer-to-peer is distinctly bad for many classes of networked applications. Most search engines work best when they can search a central database rather than launch a meta-search of peers. Electronic marketplaces need to aggregate supply and demand in a single place at a single time in order to arrive at a single, transparent price. Any system that requires real-time group access or rapid searches through large sets of unique data will benefit from centralization in ways that will be difficult to duplicate in peer-to-peer systems.

The genius of Napster is that it understands and works within these limitations.

Napster mixes centralization and decentralization beautifully. As a search engine, it builds and maintains a master song list, adding and removing songs as individual users connect and disconnect their PCs. And because the search space for Napster—popular music—is well understood by all its users, and because there is massive redundancy in the millions of collections it indexes, the chances that any given popular song can be found are very high, even if the chances that any given user is online are low.

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Like ants building an anthill, the contribution of any given individual to the system at any given moment is trivial, but the overlapping work of the group is remarkably powerful. By centralizing pointers and decentralizing content, Napster couples the strengths of a central database with the power of distributed storage. Napster has become the fastest-growing application in the Net's history in large part because it *isn't* pure peer-to-peer. Chapter 4, *The Cornucopia of the Commons*, explores this theme farther.

Where's the content?

Napster's success in pursuing this strategy is difficult to overstate. At any given moment, Napster servers keep track of thousands of PCs holding millions of songs comprising several terabytes of data. This is a complete violation of the Web's data model, "Content at the Center," and Napster's success in violating it could be labeled "Content at the Edges."

The content-at-the-center model has one significant flaw: most Internet content is created on the PCs at the edges, but for it to become universally accessible, it must be pushed to the center, to always-on, always-up web servers. As anyone who has ever spent time trying to upload material to a web site knows, the Web has made downloading trivially easy, but uploading is still needlessly hard. Napster dispenses with uploading and leaves the files on the PCs, merely brokering requests from one PC to another—the MP3 files do not have to travel through any central Napster server. Instead of trying to store these files in a central database, Napster took advantage of the largest pool of latent storage space in the world—the disks of the Napster users. And thus, Napster became the prime example of a new principle for Internet applications: Peer-to-peer services come into being by leveraging the untapped power of the millions of PCs that have been connected to the Internet in the last five years.

PCs are the dark matter of the Internet

Napster's popularity made it the proof-of-concept application for a new networking architecture based on the recognition that bandwidth to the desktop had become fast enough to allow PCs to serve data as well as request it, and that PCs are becoming powerful enough to fulfill this new role. Just as the application service provider (ASP) model is taking off, Napster's success represents the revenge of the PC. By removing the need to upload data (the single biggest bottleneck to the ASP model), Napster points the way to a reinvention of the desktop as the center of a user's data—only this time the user will no longer need physical access to the PC.

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The latent capabilities of PC hardware made newly accessible represent a huge, untapped resource and form the fuel powering the current revolution in Internet use. No matter how it gets labeled, the thing that a file-sharing system like Gnutella and a distributed computing network like Data Synapse have in common is an ability to harness this dark matter, the otherwise underused hardware at the edges of the Net.

Promiscuous computers

While some press reports call the current trend the "Return of the PC," it's more than that. In these new models, PCs aren't just tools for personal use—they're promiscuous computers, hosting data the rest of the world has access to, and sometimes even hosting calculations that are of no use to the PC's owner at all, like Popular Power's influenza virus simulations.

Furthermore, the PCs themselves are being disaggregated: Popular Power will take as much CPU time as it can get but needs practically no storage, while Gnutella needs vast amounts of disk space but almost no CPU time. And neither kind of business particularly needs the operating system—since the important connection is often with the network rather than the local user, Intel and Seagate matter more to the peer-to-peer companies than do Microsoft or Apple.

It's too soon to understand how all these new services relate to one another, and the danger of the peer-to-peer label is that it may actually obscure the real engineering changes afoot. With improvements in hardware, connectivity, and sheer numbers still mounting rapidly, anyone who can figure out how to light up the Internet's dark matter gains access to a large and growing pool of computing resources, even if some of the functions are centralized.

It's also too soon to see who the major players will be, but don't place any bets on people or companies that reflexively use the peer-to-peer label. Bet instead on the people figuring out how to leverage the underused PC hardware, because the actual engineering challenges in taking advantage of the underused resources at the edges of the Net matter more—and will create more value—than merely taking on the theoretical challenges of peer-to-peer architecture.

Nothing succeeds like address, or, DNS isn't the only game in town

The early peer-to-peer designers, realizing that interesting services could be run off of PCs if only they had real addresses, simply ignored DNS and replaced the machine-centric model with a protocol-centric one. Protocol-centric addressing

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creates a parallel namespace for each piece of software. AIM and Napster usernames are mapped to temporary IP addresses not by the Net's DNS servers, but by privately owned servers dedicated to each protocol: the AIM server matches AIM names to the users' current IP addresses, and so on.

In Napster's case, protocol-centric addressing turns Napster into merely a customized FTP for music files. The real action in new addressing schemes lies in software like AIM, where the address points to a person, not a machine. When you log into AIM, the address points to you, no matter what machine you're sitting at, and no matter what IP address is presently assigned to that machine. This completely decouples what humans care about—*Can I find my friends and talk with them online?*—from how the machines go about it—*Route packet A to IP address X.*

This is analogous to the change in telephony brought about by mobile phones. In the same way that a phone number is no longer tied to a particular physical location but is dynamically mapped to the location of the phone's owner, an AIM address is mapped to you, not to a machine, no matter where you are.

An explosion of protocols

This does not mean that DNS is going away, any more than landlines went away with the invention of mobile telephony. It does mean that DNS is no longer the only game in town. The rush is now on, with instant messaging protocols, single sign-on and wallet applications, and the explosion in peer-to-peer businesses, to create and manage protocol-centric addresses that can be instantly updated.

Nor is this change in the direction of easier peer-to-peer addressing entirely to the good. While it is always refreshing to see people innovate their way around a bottleneck, sometimes bottlenecks are valuable. While AIM and Napster came to their addressing schemes honestly, any number of people have noticed how valuable it is to own a namespace, and many business plans making the rounds are just me-too copies of Napster or AIM. Eventually, the already growing list of kinds of addresses—phone, fax, email, URL, AIM, *ad nauseam*—could explode into meaninglessness.

Protocol-centric namespaces will also force the browser into lesser importance, as users return to the days when they managed multiple pieces of Internet software. Or it will mean that addresses like *aim://12345678* or *napster://green_day_fan* will have to be added to the browsers' repertoire of recognized URLs. Expect also the rise of "meta-address" servers, which offer to manage a user's addresses for all of these competing protocols, and even to translate from one kind of address to

Microsoft Exhibit 1019

another. (These meta-address servers will, of course, need their own addresses as well.) Chapter 19, *Interoperability Through Gateways*, looks at some of the issues involved.

It's not clear what is going to happen to Internet addressing, but it is clear that it's going to get a lot more complicated before it gets simpler. Fortunately, both the underlying IP addressing system and the design of URLs can handle this explosion of new protocols and addresses. But that familiar DNS bit in the middle (which *really* put the dot in dot-com) will never recover the central position it has occupied for the last two decades, and that means that a critical piece of Internet infrastructure is now up for grabs.

An economic rather than legal challenge

Much has been made of the use of Napster for what the music industry would like to define as "piracy." Even though the dictionary definition of piracy is quite broad, this is something of a misnomer, because pirates are ordinarily in business to sell what they copy. Not only do Napster users not profit from making copies available, but Napster works precisely because the copies are free. (Its recent business decision to charge a monthly fee for access doesn't translate into profits for the putative "pirates" at the edges.)

What Napster does is more than just evade the law, it also upends the economics of the music industry. By extension, peer-to-peer systems are changing the economics of storing and transmitting intellectual property in general.

The resources Napster is brokering between users have one of two characteristics: they are either *replicable* or *replenishable*.

Replicable resources include the MP3 files themselves. "Taking" an MP3 from another user involves no loss (if I "take" an MP3 from you, it is not removed from your hard drive)—better yet, it actually adds resources to the Napster universe by allowing me to host an alternate copy. Even if I am a freeloader and don't let anyone else copy the MP3 from me, my act of taking an MP3 has still not caused any net loss of MP3s.

Other important resources, such as bandwidth and CPU cycles (as in the case of systems like SETI@home), are not replicable, but they are replenishable. The resources can be neither depleted nor conserved. Bandwidth and CPU cycles expire if they are not used, but they are immediately replenished. Thus they cannot be conserved in the present and saved for the future, but they can't be "used up" in any long-term sense either.

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Because of these two economic characteristics, the exploitation of otherwise unused bandwidth to copy MP3s across the network means that additional music can be created at almost zero marginal cost to the user. It employs resources—storage, cycles, bandwidth—that the users have already paid for but are not fully using.

All you can eat

Economists call these kinds of valuable side effects "positive externalities." The canonical example of a positive externality is a shade tree. If you buy a tree large enough to shade your lawn, there is a good chance that for at least part of the day it will shade your neighbor's lawn as well. This free shade for your neighbor is a positive externality, a benefit to her that costs you nothing more than what you were willing to spend to shade your own lawn anyway.

Napster's signal economic genius is to coordinate such effects. Other than the central database of songs and user addresses, every resource within the Napster network is a positive externality. Furthermore, Napster coordinates these externalities in a way that encourages altruism. As long as Napster users are able to find the songs they want, they will continue to participate in the system, even if the people who download songs from them are not the same people they download songs from. And as long as even a small portion of the users accept this bargain, the system will grow, bringing in more users, who bring in more songs.

Thus Napster not only takes advantage of low marginal costs, it couldn't work without them. Imagine how few people would use Napster if it cost them even a penny every time someone else copied a song from them. As with other digital resources that used to be priced per unit but became too cheap to meter, such as connect time or per-email charges, the economic logic of infinitely copyable resources or non-conservable and non-depletable resources eventually leads to "all you can eat" business models.

Thus the shift from analog to digital data, in the form of CDs and then MP3s, is turning the music industry into a smorgasbord. Many companies in the traditional music business are not going quietly, however, but are trying to prevent these "all you can eat" models from spreading. Because they can't keep music entirely off the Internet, they are currently opting for the next best thing, which is trying to force digital data to behave like objects.

Yesterday's technology at tomorrow's prices, two days late

The music industry's set of schemes, called Digital Rights Management (DRM), is an attempt to force music files to behave less like ones and zeros and more

Microsoft Exhibit 1019

like albums and tapes. The main DRM effort is the Secure Digital Music Initiative (SDMI), which aims to create a music file format that cannot be easily copied or transferred between devices—to bring the inconvenience of the physical world to the Internet, in other words.

This in turn has led the industry to make the argument that the music-loving public should be willing to pay the same price for a song whether delivered on CD or downloaded, because it is costing the industry so much money to make the downloaded file as inconvenient as the CD. When faced with the unsurprising hostility this argument engendered, the industry has suggested that matters will go their way once users are sufficiently "educated."

Unfortunately for the music industry, the issue here is not education. In the analog world, it costs money to make a copy of something. In the digital world, it costs money to prevent copies from being made. Napster has demonstrated that systems that work with the economic logic of the Internet rather than against it can have astonishing growth characteristics, and no amount of user education will reverse that.

30 million Britney fans does not a revolution make

Within this economic inevitability, however, lies the industry's salvation, because despite the rants of a few artists and techno-anarchists who believed that Napster users were willing to go to the ramparts for the cause, large-scale civil disobedience against things like Prohibition or the 55 MPH speed limit has usually been about relaxing restrictions, not repealing them.

Despite the fact that it is still possible to make gin in your bathtub, no one does it anymore, because after Prohibition ended high-quality gin became legally available at a price and with restrictions people could live with. Legal and commercial controls did not collapse, but were merely altered.

To take a more recent example, the civil disobedience against the 55 MPH speed limit did not mean that drivers were committed to having no speed limit whatsoever; they simply wanted a higher one.

So it will be with the music industry. The present civil disobedience is against a refusal by the music industry to adapt to Internet economics. But the refusal of users to countenance per-unit prices does not mean they will never pay for music at all, merely that the economic logic of digital data—its replicability and replenishability—must be respected. Once the industry adopts economic models that do, whether through advertising or sponsorship or subscription pricing,

Microsoft Exhibit 1019

the civil disobedience will largely subside, and we will be on the way to a new speed limit.

In other words, the music industry as we know it is not finished. On the contrary, all of their functions other than the direct production of the CDs themselves will become more important in a world where Napster economics prevail. Music labels don't just produce CDs; they find, bankroll, and publicize the musicians themselves. Once they accept that Napster has destroyed the bottleneck of distribution, there will be more music to produce and promote, not less.

Peer-to-peer architecture and second-class status

With this change in addressing schemes and the renewed importance of the PC chassis, peer-to-peer is not merely erasing the distinction between client and server. It's erasing the distinction between consumer and provider as well. You can see the threat to the established order in a recent legal action: a San Diego cable ISP, Cox@Home, ordered several hundred customers to stop running Napster not because they were violating copyright laws, but because Napster leads Cox subscribers to use too much of its cable network bandwidth.

Cox built its service on the current web architecture, where producers serve content from always-connected servers at the Internet's center and consumers consume from intermittently connected client PCs at the edges. Napster, on the other hand, inaugurated a model where PCs are always on and always connected, where content is increasingly stored and served from the edges of the network, and where the distinction between client and server is erased. *Cox v. Napster* isn't just a legal fight; it's a fight between a vision of helpless, passive consumers and a vision where people at the network's edges can both consume and produce.

Users as consumers, users as providers

The question of the day is, "Can Cox (or any media business) force its users to retain their second-class status as mere consumers of information?" To judge by Napster's growth, the answer is "No."

The split between consumers and providers of information has its roots in the Internet's addressing scheme. Cox assumed that the model ushered in by the Web—in which users never have a fixed IP address, so they can consume data stored elsewhere but never provide anything from their own PCs—was a permanent feature of the landscape. This division wasn't part of the Internet's original

Microsoft Exhibit 1019

architecture, and the proposed fix (the next generation of IP, called IPv6) has been coming Real Soon Now for a long time. In the meantime, services like Cox have been built with the expectation that this consumer/provider split would remain in effect for the foreseeable future.

How short the foreseeable future sometimes is. When Napster turned the Domain Name System inside out, it became trivially easy to host content on a home PC, which destroys the asymmetry where end users consume but can't provide. If your computer is online, it can be reached even without a permanent IP address, and any material you decide to host on your PC can become globally accessible. Napster-style architecture erases the people-based distinction between provider and consumer just as surely as it erases the computer-based distinction between server and client.

There could not be worse news for any ISP that wants to limit upstream bandwidth on the expectation that edges of the network host nothing but passive consumers. The limitations of cable ISPs (and Asymmetric Digital Subscriber Line, or ADSL) become apparent only if its users actually want to do something useful with their upstream bandwidth. The technical design of the cable network that hamstrings its upstream speed (upstream speed is less than a tenth of Cox's downstream) just makes the cable networks the canary in the coal mine.

New winners and losers

Any media business that relies on a neat division between information consumer and provider will be affected by roving, peer-to-peer applications. Sites like GeoCities, which made their money providing fixed addresses for end user content, may find that users are perfectly content to use their PCs as that fixed address. Copyright holders who have assumed up until now that only a handful of relatively identifiable and central locations were capable of large-scale serving of material are suddenly going to find that the Net has sprung another million leaks.

Meanwhile, the rise of the end user as information provider will be good news for other businesses. DSL companies (using relatively symmetric technologies) will have a huge advantage in the race to provide fast upstream bandwidth; Apple may find that the ability to stream home movies over the Net from a PC at home drives adoption of Mac hardware and software; and of course companies that provide the Napster-style service of matching dynamic IP addresses with fixed names will have just the sort of sticky relationship with their users that venture capitalists slaver over.

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Real technological revolutions are human revolutions as well. The architecture of the Internet has effected the largest transfer of power from organizations to individuals the world has ever seen, and it is only getting started. Napster's destruction of the serving limitations on end users shows how temporary such bottlenecks can be. Power is gradually shifting to the individual for things like stock brokering and buying airline tickets. Media businesses that have assumed such shifts wouldn't affect them are going to be taken by surprise when millions of passive consumers are replaced by millions of one-person media channels.

This is not to say that all content is going to the edges of the Net, or that every user is going to be an enthusiastic media outlet. But enough consumers will become providers as well to blur present distinctions between producer and consumer. This social shift will make the next generation of the Internet, currently being assembled, a place with greater space for individual contributions than people accustomed to the current split between client and server, and therefore provider and consumer, had ever imagined.

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chapter three

Remaking the Peer-to-Peer Meme

Tim O'Reilly, O'Reilly & Associates

On September 18, 2000, I organized a so-called "peer-to-peer summit" to explore the bounds of peer-to-peer networking. In my invitation to the attendees, I set out three goals:

1. To make a statement, by their very coming together, about the nature of peer-to-peer and what kinds of technologies people should think of when they hear the term.
2. To make some introductions among people whom I like and respect and who are working on different aspects of what could be seen as the same problem—peer-to-peer solutions to big problems—in order to create some additional connections between technical communities that ought to be talking to and learning from each other.
3. To do some brainstorming about the issues each of us are uncovering, so we can keep projects from reinventing the wheel and foster cooperation to accelerate mutual growth.

In organizing the summit, I was thinking of the free software (open source) summit I held a few years back. Like free software at that time, peer-to-peer currently has image problems and a difficulty developing synergy. The people I was talking to all knew that peer-to-peer is more than just swapping music files, but the wider world was still focusing largely on the threats to copyright. Even people working in the field of peer-to-peer have trouble seeing how far its innovations can extend; it would benefit them to learn how many different types of technologies share the same potential and the same problems.

This is exactly what we did with the open source summit. By bringing together people from a whole lot of projects, we were able to get the world to recognize that free software was more than GNU and Linux; we introduced a lot of

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people, many of whom, remarkably, had never met; we talked shop; and ultimately, we crafted a new "meme" that completely reshaped the way people thought about the space.

The people I invited to the peer-to-peer summit tell part of the story. Gene Kan from Gnutella (*http://gnutella.wego.com*) and Ian Clarke from Freenet (*http://freenet.sourceforge.net*) were obvious choices. They matched the current industry buzz about peer-to-peer file sharing. Similarly, Marc Hedlund and Nelson Minar from Popular Power (*http://www.popularpower.com*) made sense, because there was already a sense of some kind of connection between distributed computation and file sharing.

But why did I invite Jeremie Miller of Jabber and Ray Ozzie of Groove, Ken Arnold from Sun's Jini project and Michael Tiemann of Red Hat, Marshall Rose (author of BXXP and IMXP), Rael Dornfest of meerkat and RSS 1.0, Dave Stutz of Microsoft, Andy Hertzfeld of Eazel, Don Box (one of the authors of SOAP) and Steve Burbeck (one of the authors of UDDI)? (Note that not all of these people made it to the summit; Ian Clarke sent Scott Miller in his stead, and Ken Arnold and Don Box had to cancel at the last minute.) As I said in my invitation:

> [I've invited] a group of people who collectively bracket what I consider a new paradigm, which could perhaps best be summarized by Sun's slogan, "The Network is the Computer." They're all working on parts of what I consider the next-generation Net story.

This chapter reports on some of the ideas discussed at the summit. It continues the job of trying to reshape the way people think about that "next-generation Net story" and the role of peer-to-peer in telling that story. It also shows one of the tools I used at the meeting—something I'll call a "meme map"—and presents the results of the meeting in that form.

The concepts we bear in our minds are, at bottom, maps of reality. Bad maps lead to bad decisions. If we believe peer-to-peer is about illegal sharing of copyrighted material, we'll continue to see rhetoric about copyright and censorship at the heart of the debate, and may push for ill-advised legal restrictions on the use of the technology. If we believe it's about a wider class of decentralized networking applications, we'll focus instead on understanding what those applications are good for and on advancing the state of the art.

The meme map we developed at the peer-to-peer summit has two main benefits. First, the peer-to-peer community can use it to organize itself—to understand who is doing related work and identify areas where developers can learn from each other. Second, the meme map helps the community influence outsiders. It can create excitement where there previously was indifference and turn negative impressions into positive ones. Tangentially, the map is also useful in

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understanding the thinking behind the O'Reilly Network's P2P directory, a recent version of which is republished in this book as an appendix.

First, though, a bit of background.

From business models to meme maps

Recently, I started working with Dan and Meredith Beam of Beam, Inc., a strategy consulting firm. Dan and Meredith help companies build their "business models"—one page pictures that describe "how all the elements of a business work together to build marketplace advantage and company value." It's easy to conclude that two companies selling similar products and services are in the same business, but the Beams think otherwise.

For example, O'Reilly and IDG compete in the computer book publishing business, but we have completely different business models. Their strategic positioning is to appeal to the "dummy" who needs to learn about computers but doesn't really want to. Ours is to appeal to the people who love computers and want to go as deep as possible. Their marketing strategy is to build a widely recognized consumer brand, and then dominate retail outlets and "big box" stores in hopes of putting product in front of consumers who might happen to walk by in search of any book on a given subject. Our marketing strategy is to build awareness of our brand and products in the core developer and user communities, who then buy directly or drive traffic to retail outlets. The former strategy pushes product into distribution channels in an aggressive bid to reach unknown consumers; the latter pulls products into distribution channels as they are requested by consumers who are already looking for the product. Both companies are extremely successful, but our different business models require different competencies. I won't say more lest this chapter turn into a lesson for O'Reilly competitors, but hopefully I have said enough to get the idea across.

Boiling all the elements of your business down into a one-page picture is a really useful exercise. But what is even more useful is that Dan and Meredith have you run the exercise twice, once to describe your present business, and once to describe it as you want it to be.

At any rate, fresh from the strategic planning process at O'Reilly, it struck me that an adaptation of this idea would be useful preparation for the summit. We weren't modeling a single business but a technology space—the key projects, concepts, and messages associated with it.

I call these pictures "meme maps" rather than "business models" in honor of Richard Dawkins' wonderful contribution to cultural studies. He formulated the idea of "memes" as ideas that spread and reproduce themselves, passed on from

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mind to mind. Just as gene engineering allows us to artificially shape genes, meme engineering lets us organize and shape ideas so that they can be transmitted more effectively, and have the desired effect once they are transmitted. That's what I hoped to touch off at the summit, using a single picture that shows how a set of technologies fit together and demonstrates a few central themes.

A success story: From free software to open source

In order to illustrate the idea of a meme map to the attendees at the peer-to-peer summit, I drew some maps of free software versus open source. I presented these images at the summit as a way of kickstarting the discussion. Let's look at those here as well, since it's a lot easier to demonstrate the concept than it is to explain it in the abstract.

I built the free software map in Figure 3-1 by picking out key messages from the Free Software Foundation (FSF) web site, *http://www.fsf.org*. I also added a few things (the darker ovals in the lower right quadrant of the picture) to show common misconceptions that were typically applied to free software. This figure, and the others in this chapter are slightly edited versions of slides used at the summit.

Please note that this diagram should not be taken as a complete representation of the beliefs of the Free Software Foundation. I simply summarized my interpretation of the attitudes and positioning I found on their web site. No one from the Free Software Foundation has reviewed this figure, and they might well highlight very different points if given the chance to do so.

There are a couple of things to note about the diagram. The ovals at the top represent the outward face of the movement—the projects or activities that the movement considers *canonical* in defining itself. In the case of the Free Software Foundation, these are programs like *gcc* (the GNU C Compiler), GNU Emacs, GhostScript (a free PostScript display tool), and the GNU General Public License, or GPL.

The box in the center lists the strategic positioning, the key perceived user benefit, and the core competencies. The strategic goal I chose came right up front on the Free Software Foundation web site: to build a complete free replacement for the Unix operating system. The user benefit is sold as one of standing up for what's right, even if there would be practical benefits in compromising. The web site shows little sense of what the core competencies of the free software movement might be, other than that they have right on their side, along with the goodwill of talented programmers.

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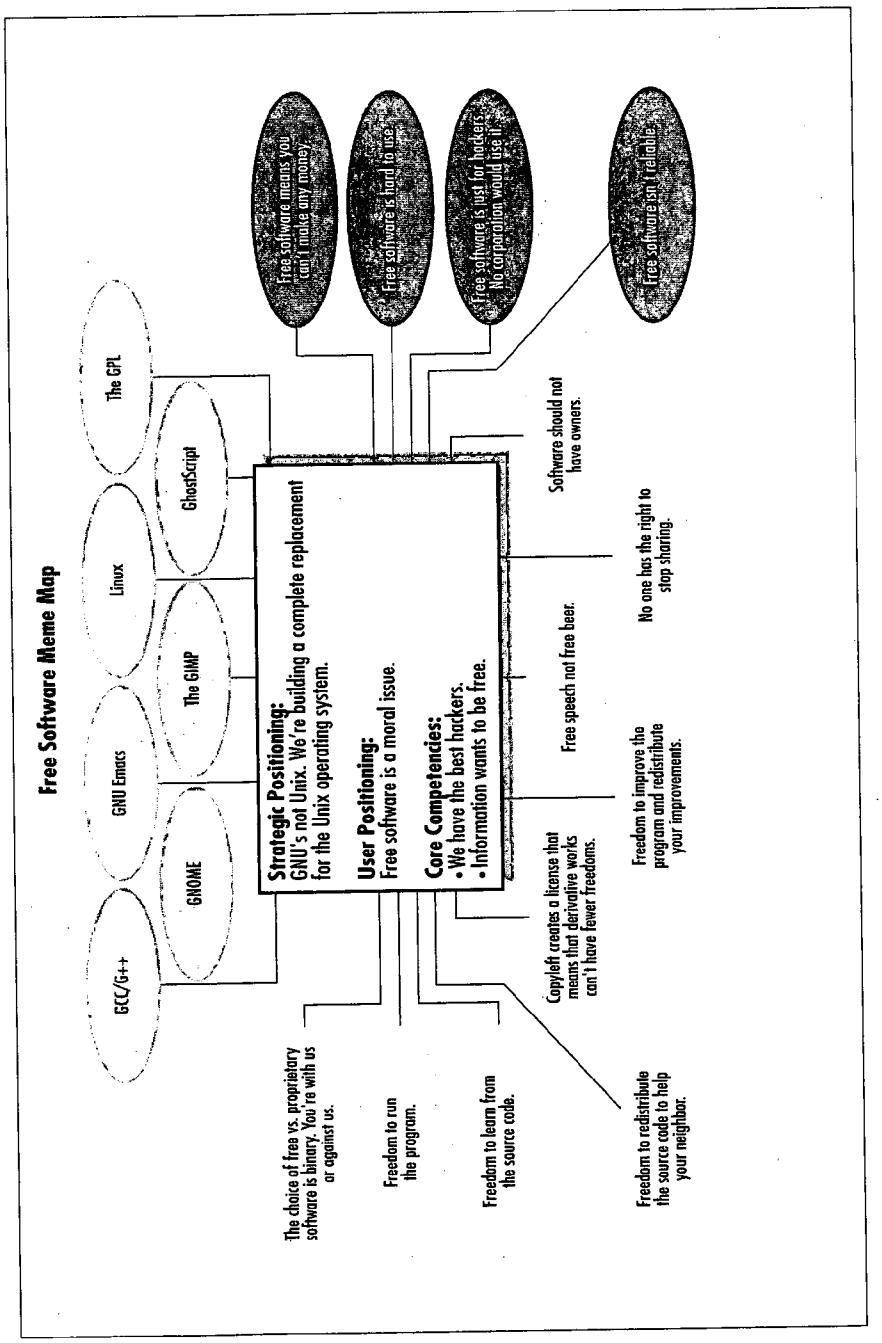

Figure 3-1. Map of the old free software meme

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In the Beam models, the ovals at the bottom of the picture represent internal activities of the business; for my purposes, I used them to represent guiding principles and key messages. I used dark ovals to represent undesirable messages that others might be creating and applying to the subject of the meme map.

As you can see, the primary messages of the free software movement, thought-provoking and well articulated as they are, don't address the negative public perceptions that are spread by opponents of the movement.

Now take a look at the diagram I drew for open source—the alternative term for free software that was invented shortly before we held our open source summit in April 1998. The content of this diagram, shown in Figure 3-2, was taken partly from the Open Source Initiative web site *http://www.opensource.org*, but also from the discussions at the summit and from my own thinking and speaking about open source in the years since. Take the time to read the diagram carefully; it should be fairly self-explanatory, but I'll offer some insights into a few subtleties. The figure demonstrates what a well-formed strategic meme map ought to look like.

As you can see by comparing the two diagrams, they put a completely different spin on what formerly might have been considered the same space. We did more than just change the name that we used to describe a collection of projects from "free software" to "open source." In addition:

- We changed the canonical list of projects that we wanted to hold up as exemplars of the movement. (Even though BIND and sendmail and Apache and Perl are "free software" by the Free Software Foundation's definition, they aren't central to its free software "meme map" in the way that we made them for open source; even today, they are not touted on the Free Software Foundation web site.) What's more, I've included a tag line that explains why each project is significant. For example, BIND isn't just another free software program; it's the heart of the Domain Name System and the single most mission-critical program on the Internet. Apache is the dominant web server on the market, sendmail routes most Internet email and Linux is more reliable than Windows. The Free Software Foundation's GNU tools are still in the picture, but they are no longer at its heart.
- The strategic positioning is much clearer. Open source is not about creating a free replacement for Unix. It's about making better software through sharing source code and using the Internet for collaboration. The user positioning (the benefit to the user) was best articulated by Bob Young of Red Hat, who insisted that what Red Hat Linux offers to its customers is control over their own destiny.

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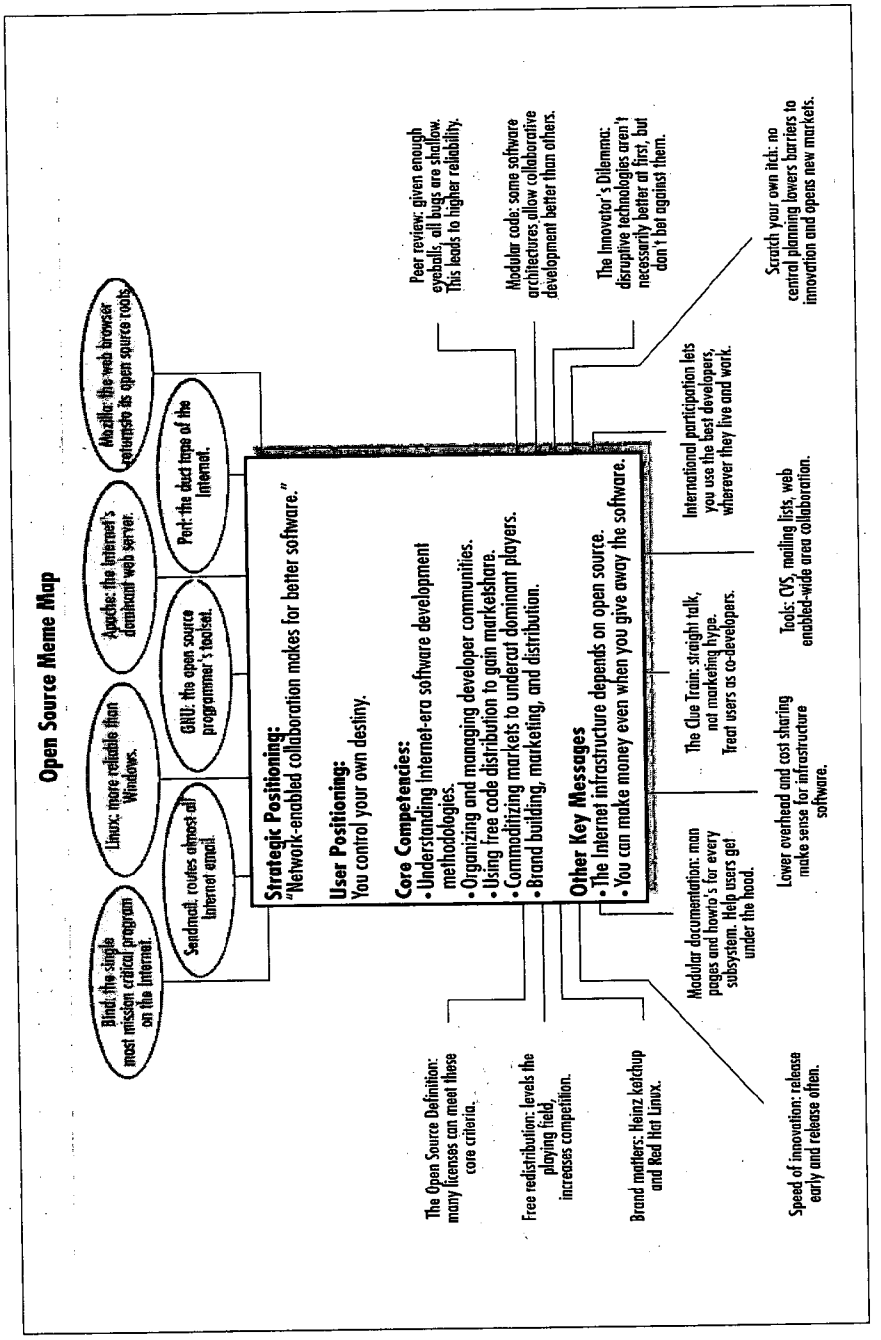

Figure 3-2. Map of the new open source meme

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- The list of core competencies is much more focused and actionable. The most successful open source communities do in fact understand something about distributed software development in the age of the Internet, organizing developer communities, using free distribution to gain market share, commoditizing markets to undercut dominant players, and creating powerful brands for their software. Any aspiring open source player needs to be good at all of these things.
- We've replaced the negative messages used against free software with directly competing messages that counter them. For instance, where free software was mischaracterized as unreliable, we set out very explicitly to demonstrate that everyone counts on open source programs, and that the peer review process actually improves reliability and support.
- We've identified a set of guiding principles that can be used by open source projects and companies to see if they're hitting all the key points, or that can be used to explain why some projects have failed to gain as much traction as expected. For example, Mozilla's initial lack of modular code, weak documentation, and long release cycles hampered its quick uptake as an open source project. (That being said, key portions of Mozilla code are finally starting to appear in a variety of other open source projects, such as ActiveState's Komodo development environment and Eazel's Nautilus file manager.)
- We made connections between open source and related concepts that help to place it in context. For example, the concept from *The ClueTrain Manifesto* of open interaction with customers, and the idea of "disruptive technologies" from Clayton Christenson's book *The Innovator's Dilemma*, link open source to trends in business management.

While some further discussion of the open source meme map might be worthwhile in another context, I present it here mainly to clarify the use of meme maps to create a single unifying vision of a set of related technologies.

The current peer-to-peer meme map

The meme map for peer-to-peer is still very unformed, and consists largely of ideas applied by the media and other outsiders.

Figure 3-3 is the slide I showed to the group at the summit. Things have evolved somewhat since that time, partly as a result of efforts such as ours to correct common misconceptions, but this picture still represents the view being bandied about by industries that feel threatened by peer-to-peer technologies.

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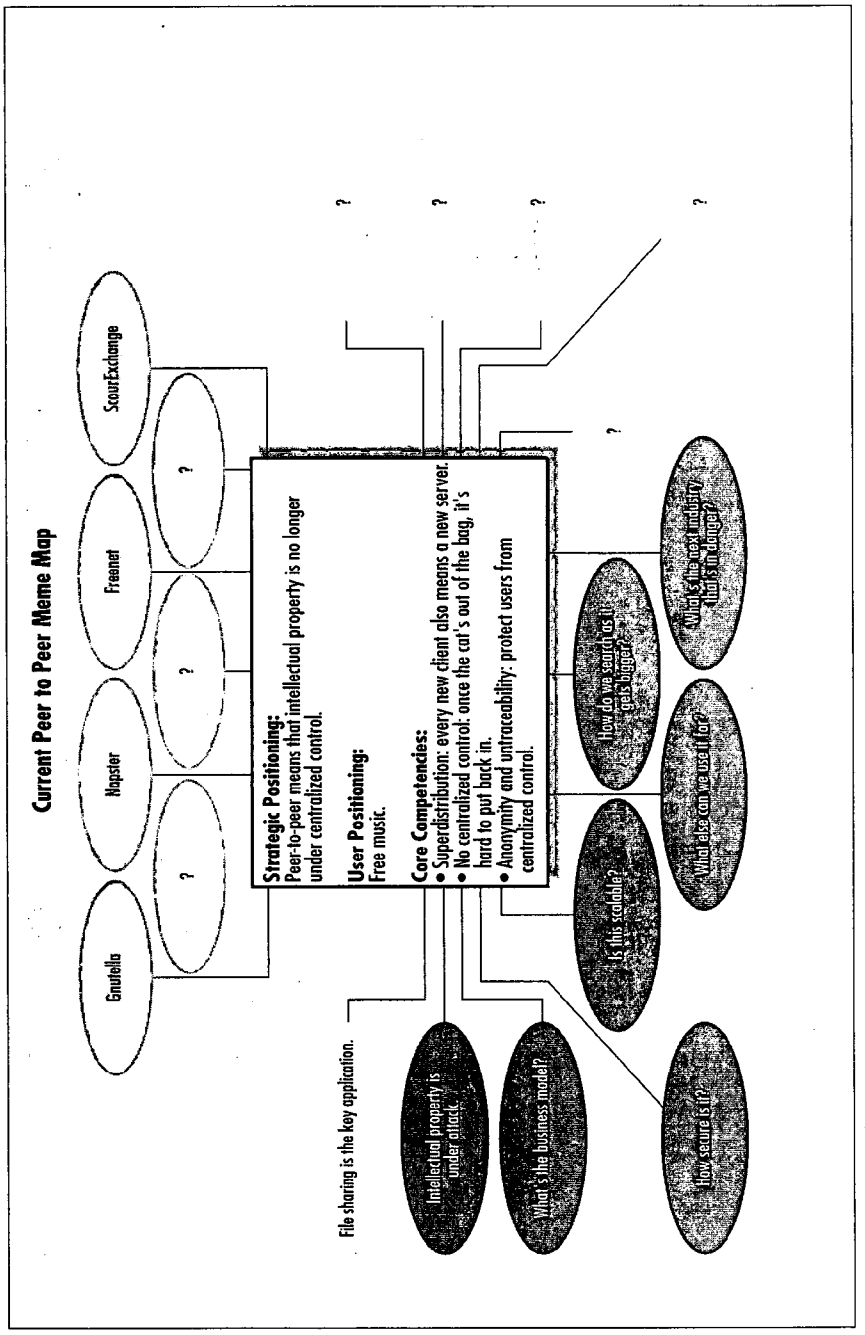

Figure 3-3. Map of currently circulating peer-to-peer meme

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Not a pretty picture. The canonical projects all feed the idea that peer-to-peer is about the subversion of intellectual property. The chief benefit presented to users is that of free music (or other copyrighted material). The core competencies of peer-to-peer projects are assumed to be superdistribution, the lack of any central control point, and anonymity as a tool to protect the system from attempts at control.

Clearly, these are characteristics of the systems that put the peer-to-peer buzzword onto everyone's radar. But are they really the key points? Will they help peer-to-peer developers work together, identify problems, develop new technologies, and win the public over to those technologies?

A map is useful only to the extent that it reflects underlying reality. A bad map gets you lost; a good one helps you find your way through unfamiliar territory. Therefore, one major goal for the summit was to develop a better map for the uncharted peer-to-peer space.

The new peer-to-peer meme map

In a space as vaguely defined as peer-to-peer, we need to consider many angles at once in order to come up with an accurate picture of what the technology is and what is possible. Our summit looked at many projects from different sources, often apparently unrelated. We spent a few hours brainstorming about important applications of peer-to-peer technology, key principles, and so on. I've tried to capture the results of that brainstorming session in the same form that I used to spark the discussion, as the meme map in Figure 3-4. Note that this is what I took away personally from the meeting. The actual map below wasn't fully developed or approved there.

A quick walkthrough of the various projects and how they fit together leads us to a new understanding of the strategic positioning and core competencies for peer-to-peer projects. In the course of this walkthrough, I'll also talk about some of the guiding principles that we can derive from studying each project, which are captured in the ovals at the top of the diagram. This discussion is necessarily quite superficial, but suggests directions for further study.

File sharing: Napster and successors

One of the most obvious things about the map I've drawn of the peer-to-peer space is that file-sharing applications such as Napster, Gnutella, and Freenet are only a small part of the picture, even though they have received the lion's share of the attention to date. Nonetheless, Napster (*http://www.napster.com*), as the application whose rapid uptake and enormous impact on the music industry sparked the furor over peer-to-peer, deserves some significant discussion.

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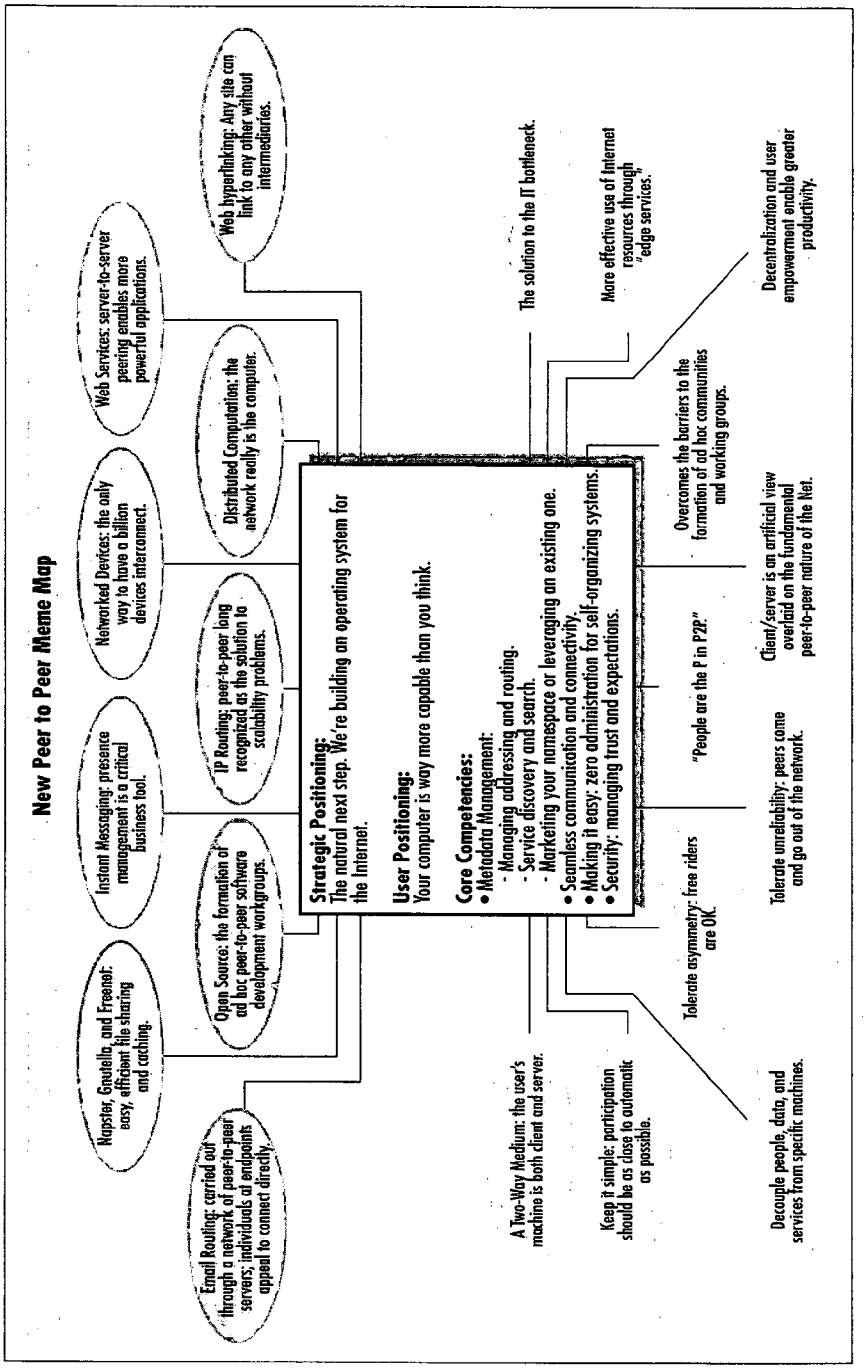

Figure 3-4. Map of peer-to-peer meme as it is starting to be understood

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One of the most interesting things about Napster is that it's not a pure peer-to-peer system in the same way that radically decentralized systems like Gnutella and Freenet are. While the Napster data is distributed across millions of hard disks, finding that data depends on a central server. In some ways, the difference between MP3.com and Napster is smaller than it appears: one centralizes the files, while the other centralizes the addresses of the files.

The real genius of Napster is the way it makes participation automatic. By default, any consumer is also a producer of files for the network. Once you download a file, your machine is available to pass along the file to other users. Automatic "pass along" participation decentralizes file storage and network bandwidth, but most importantly, it also distributes the job of building the Napster song database.

Dan Bricklin has written an excellent essay on this subject, which we've printed in this book as Chapter 4, *The Cornucopia of the Commons*. In this wonderful reversal of Hardin's tragedy of the commons, Bricklin explains why Napster demonstrates the power of collectively assembled databases in which "increasing the value of the database by adding more information is a natural by-product of using the tool for your own benefit."

This feature is also captured by an insightful comment by innovative software developer Dave Winer: "The P in P2P is People."

Dave's comment highlights why the connection to the open source movement is significant. Open source projects are self-organizing, decentralized workgroups enabled by peer-to-peer Internet technologies. If the P in P2P is people, the technologies that allow people to create self-organizing communities and the frameworks developed for managing those communities provide important lessons for those who want to work in the P2P space.

Open source isn't driven just by a set of licenses for software distribution, but more deeply by a set of techniques for collaborative, wide-area software development. Open source and peer-to-peer come full circle here. One of the key drivers of the early open source community was the peer-to-peer Usenet, which I'll discuss later in the chapter. Both open source and peer-to-peer are technologies that allow people to associate freely, end-to-end, and thus are great levelers and great hotbeds promoting innovation.

Napster also illustrates another guiding principle: tolerance for redundancy and unreliability. I was talking recently with Eric Schmidt, CEO of Novell, about lessons from peer-to-peer. He remarked on a conversation he'd had with his 13-year-old daughter. "Does it bother you," he asked, "that sometimes songs are there, and sometimes they aren't? Does it bother you that there are lots of

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copies of the same song, and that they aren't all the same?" Her answer—that neither of these things bothered her in the slightest—seemed to him to illustrate the gulf between the traditional computer scientist's concern for reliability and orthogonality and the user's indifference for these issues.

Another important lesson from Napster is that free riders, "super peers" providing more or better resources, and other variations in peer participation will ultimately decrease the system's decentralization. Experience is already showing that a hierarchy is starting to emerge. Some users turn off file sharing. Even among those who don't, some have more files, and some have better bandwidth. As in Orwell's *Animal Farm*, all animals are equal, but some are more equal than others. While this idea is anathema to those wedded to the theory of radical decentralization, in practice, it is this very feature that gives rise to many of the business opportunities in the peer-to-peer space. It should give great relief to those who fear that peer-to-peer will lead to the leveling of all hierarchy and the end of industries that depend on it. The most effective way for the music industry to fight what they fear from Napster is to join the trend, and provide sites that become the best source for high-quality music downloads.

Even on Gnutella, the concept of super peers is starting to emerge. The service DSS (Distributed Search Solutions) from Clip2.com, Inc. (*http://dss.clip2.com*) has developed a program that they call a Gnutella "Reflector." This is a proxy and index server designed to make Gnutella more scalable. According to Kelly Truelove of Clip2, "Multiple users connect to such a Reflector as they might connect to a Napster central server, yet, unlike such a central server, the Reflector itself can function as a peer, making outgoing connections to other peers on the network."

Mixing centralization and decentralization: Usenet, email, and IP routing

Not coincidentally, this evolution from a pure peer-to-peer network to one in which peer-to-peer and centralized architectures overlap echoes the evolution of Usenet. This history also shows that peer-to-peer and client/server (which can also be called decentralization and centralization) are not mutually exclusive.

Usenet was originally carried over the informal, peer-to-peer, dial-up network known as UUCPnet. Sites agreed to phone each other, and passed mail and news from site to site in a store-and-forward network. Over time, though, it became clear that some sites were better connected than others; they came to form a kind of de facto "Usenet backbone." One of the chief sites, seismo, a computer at the U.S. Geological Society, was run by Rick Adams. By 1987, the load on seismo had become so great that Rick formed a separate company,

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called UUnet (*http://www.uu.net*), to provide connectivity services for a monthly fee.

As the UUCPnet was replaced by the newly commercialized Internet, UUnet added TCP/IP services and became the first commercial Internet service provider. ISPs create a layer of hierarchy and centralization even though the IP routing infrastructure of the Internet is still peer-to-peer. Internet routers act as peers in finding the best route from one point on the Net to another, but users don't find each other directly any more. They get their Internet connectivity from ISPs, who in turn connect to each other in asymmetric hierarchies that are hidden from the end user. Yet beneath the surface, each of those ISPs still depends on the same peer-to-peer architecture.

Similarly, email is routed by a network of peered mail servers, and while it appears peer-to-peer from the user point of view, those users are in fact aggregated into clusters by the servers that route their mail and the organizations that operate those servers.

Centralization and decentralization are never so clearly separable as anyone fixated on buzzwords might like.

Maximizing use of far-flung resources: Distributed computation

Some of the earliest projects that excited the public about the potential for coordinating peers were distributed computation programs like SETI@home. This project is described by one of its founders in Chapter 5. Served from the Space Sciences Lab at U.C. Berkeley, SETI@home runs as a screensaver that uses the "spare cycles" from more than 1 million PCs to process radio telescope data in search of signs of extraterrestrial intelligence.

Viewed from one angle, distributed computation programs are not at all peer-to-peer. After all, they use an old-style, asymmetric, client/server architecture, in which the million independent computational clients download their data sets and upload their computed results to the central repository at the Space Sciences Lab. The clients don't peer with each other in any way.

But look a little deeper, and something else emerges: the clients are active participants, not just passive "browsers." What's more, the project uses the massive redundancy of computing resources to work around problems such as reliability and network availability of any one resource.

But even more importantly, look further down the development timeline when startups such as United Devices, Popular Power, Parabon, and others have their services in the market. At that point, the "ecology" of distributed computation is

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going to be much more complex. There will be thousands (and ultimately, perhaps millions) of compute-intensive tasks looking for spare cycles. At what point does it make sense to design a specialized architecture that facilitates a two-way flow of tasks and compute cycles?

Further, many of the key principles of Napster are also at play in distributed computation. Both Napster and SETI@home need to create and manage metadata about a large community of distributed participants. Both need to make it incredibly simple to participate.

Finally, both Napster and SETI@Home have tried to exploit what Clay Shirky (who contributed Chapter 2, *Listening to Napster*, to this book) memorably called "the dark matter of the Internet"—the hundreds of millions of interconnected PCs that have hitherto been largely passive participants in the network.

Already, startups like Mojo Nation (*http://www.mojonation.net*) are making a link between file sharing and distributed computation. In the end, both distributed file sharing and distributed computation are aspects of a new world where individual computer systems take on their most important role as part of a network—where the whole is much greater than the sum of its parts.

Immediate information sharing: The new instant messaging services

Napster could be characterized as a "brokered peer-to-peer system," in which a central addressing authority connects end points, and then gets out of the way.

Once you realize this, it becomes clear just how similar the Napster model is to instant messaging. In each case, a central authority manages an addressing system and a namespace that allows the unique identification of each user. These are employed by the system to connect end users. In some ways, Napster can be thought of as an instant messaging system in which the question isn't, "Are you online and do you want to chat?" but, "Are you online and do you have this song?"

Not surprisingly, a project like AIMster (*http://www.aimster.com*) makes explicit use of this insight to build a file-sharing network that uses the AOL Instant Messenger (AIM) protocol. This brings IM features such as buddy lists into the file-sharing arena.

The Jabber instant messaging platform (*http://www.jabbercentral.com*) takes things even further. An open source project, Jabber started out as a switching system between incompatible instant messaging protocols; it is evolving into a general XML routing system and a basis for applications that allow users and their computers to ask each other even more interesting questions.

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Ray Ozzie's Groove Networks (*http://www.groove.net*) is an even more mature expression of the same insight. It provides a kind of groupware dial tone or "LAN on demand" for ad hoc groups of peers. Like Jabber, it provides an XML routing infrastructure that allows for the formation of ad hoc peer groups. These can share not only files and chat, but a wide variety of applications. Replication, security, and so on are taken care of automatically by the underlying Groove system.

If systems like AIMster, Jabber, and Groove deliver what they promise, we can see peer-to-peer as a solution to the IT bottleneck, allowing users to interact more directly with each other in networks that can span organizational boundaries. Beyond the potential efficiency of such networks, peer-to-peer systems can help people share ideas and viewpoints more easily, ultimately helping the formation of online communities.

The writable Web

The Web started out as a participatory groupware system. It was originally designed by Tim Berners-Lee as a way for high-energy physicists to share their research data and conclusions. Only later was it recast into a publishing medium, in which sites seek to produce content that attracts millions of passive consumers.

To this day, there is a strong peer-to-peer element at the very heart of the Web's architecture: the hyperlink. A web hyperlink can point to any other site on the network, without any central intervention, and without the permission of the site being pointed to. What's more, hyperlinks can point to a variety of resources, not just web pages. Part of the Web's explosive growth, as compared to other early Internet information services, was that the web browser became a kind of universal client that was able to link to any kind of Internet resource. Initially, these resources were competing services such as FTP, Gopher, and WAIS. But eventually, through CGI, the Web became an interface to virtually any information resource that anyone wanted to make available. Mailto and news links even provide gateways to mail and Usenet.

There's still a fundamental flaw in the Web as it has been deployed, though. Tim Berners-Lee created both a web server and a web browser, but he didn't join them at the hip the way Napster did. And as the Buddhist Dhammapadda says, "If the gap between heaven and earth is as wide as a barleycorn, it is as wide as all heaven and earth." Before long, the asymmetry between clients and servers had grown wide enough to drive a truck through.

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Browsers were made freely available to anyone who wanted to download one, but servers were seen as a high-priced revenue opportunity, and were far less widely deployed. There were free Unix servers available (including the NCSA server, which eventually morphed into Apache), but by 1995, 95% of Web users were on Windows, and there was no web server at all available to them! In 1995, in an attempt to turn the tide, O'Reilly introduced Website. The first web server for Windows, it tried to push the market forward with the slogan "Everyone who has a web browser ought to have a web server." But by then, the market was fixated on the idea of the web server as a centralized publishing tool. Microsoft eventually offered PWS, or Personal Web Server, bundled with Windows, but it was clearly a low-powered, second-class offering.

Perhaps even more importantly, as several authors in this book point out, the rise of dynamic IP addressing made it increasingly difficult for individuals to publish to the Web from their desktops. As a result, the original "Two-Way Web" became something closer to television, a medium in which most of the participants are consumers, and only a relatively small number are producers.

Web site hosting services and participatory sites like GeoCities made it somewhat easier to participate, but these services were outside the mainstream of web development, with a consumer positioning and nonstandard tools.

Recently, there's been a new emphasis on the "writable Web," with projects like Dave Winer's EditThisPage.Com (*http://www.editthispage.com*), Dan Bricklin's Trellix (*http://www.trellix.com*), and Pyra's Blogger (*http://www.blogger.com*) making it easy for anyone to host their own site and discussion area. Wiki (*http://c2.com/cgi/wiki?WikiWikiWeb*) is an even more extreme innovation, creating web sites that are writable by anyone in an area set aside for public comment on a given topic. Wiki has actually been around for about six or seven years, but has suddenly started to catch on.

The writable Web is only one way that the Web is recapturing its peer-to-peer roots. Content syndication with Rich Site Summary (RSS), which I'll describe in the following section, and web services built with protocols like XML-RPC and SOAP allow sites to reference each other more fully than is possible with a hyperlink alone.

Web services and content syndication

I asked above, "At what point does it make sense to have an architecture that allows a two-way flow of tasks and compute cycles?" That's actually a pretty good description of SOAP and other web services architectures.

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The contribution of SOAP is to formalize something that sophisticated programmers have been doing for years. It's been relatively easy, using Perl and a library like libwww-perl, to build interfaces to web sites that do "screen scraping" and then reformulate and reuse the data in ways that the original web developers didn't intend. Jon Udell (co-author of Chapter 18, *Security*) demonstrated that one could even take data from one web site and pass it to another for further processing, in a web equivalent to the Unix pipeline.

SOAP makes this process more explicit, turning web sites into peers that can provide more complex services than simple CGI forms to their users. The next generation of web applications won't consist of single-point conversations between a single server and a single browser, but a multipoint conversation between cooperating programs.

One of the key issues that comes up, once you start thinking about more complex interactions between sites on the Net, is that metadata management is critical. In order for web clients and servers to use others as resources, they need a standard way to *discover* each other, the way Java-enabled devices discover each other through Jini. An initiative called Universal Description, Discovery, and Integration, or UDDI (*http://www.uddi.org*) represents a first step in this direction.

Similarly, content syndication formats like RSS allow web sites to cooperate in delivering content. By publishing RSS feeds, sites enable other sites to automatically pick up data about their stories. For instance, the O'Reilly Network home page is updated automatically out of a set of RSS news feeds from a web of cooperating sites.

Right now, RSS provides only the simplest of metadata about web pages, useful for simple syndication applications like creating news digest pages. But the RSS 1.0 proposal (*http://www.xml.com/pub/r/810*) will allow for more complex applications based on distributed data.

Peer-to-peer and devices

We've all heard popular descriptions of technologies such as BlueTooth and Jini. I walk into a room with my wireless laptop, and it queries other devices: "Hey, are there any printers here that can print a PostScript file?"

If this isn't peer-to-peer, what is? As we have billions of computing devices, some fixed, some mobile, some embedded in a variety of appliances (even in our clothing), we'll need technologies that allow the formation of ad hoc peer groups between devices.

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As you look at these technologies, you see a great deal of overlap between the kinds of problems that need to be solved for peer-to-peer devices and for peer-to-peer network applications ranging from web services to file sharing. Key technologies include resource discovery, reliability through redundancy, synchronization, and replication.

Strategic positioning and core competencies

The whirlwind tour of canonical projects we've just been through weaves a story about peer-to-peer that's very different from the one we started with. Not only is peer-to-peer fundamental to the architecture of the existing Internet, but it is showing us important directions in the future evolution of the Net. In some ways, you can argue that the Net is reaching a kind of critical mass, in which the network itself is the platform, more important than the operating system running on the individual nodes.

Sun first articulated this vision many years ago with the slogan "The Network is the Computer," but that slogan is only now coming true. And if the network is the computer, the projects under the peer-to-peer umbrella are collectively involved in defining the operating system for that emergent global computer.

That positioning guides technology developers. But there is a story for users too: you and your computer are more powerful than you think. In the peer-to-peer vision of the global network, a PC and its users aren't just passive consumers of data created at other central sites.

Since the most promising peer-to-peer applications of the near future are only beginning to be developed, it's crucial to provide a vision of the core competencies that peer-to-peer projects will need to bring to the table.

High on the list is metadata management, which is the subject of Chapter 13, *Metadata*. Whether you're dealing with networked devices, file sharing, distributed computation, or web services, users need to find each other and what they offer. While we don't have a clear winner in the resource discovery area, XML has emerged as an important component in the puzzle.

What do we mean by metadata? In the case of Napster, metadata means the combination of artist and song names that users search for. It also includes additional data managed by the central Napster server, such as the names and Internet addresses of users, the size of the music files, and the reported amount of bandwidth of the user's Internet link. (You can think of this information as the Napster "namespace," a privately-managed metadata directory that gives Napster the ability to link users and their files with each other.)

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In considering Napster, it's worth noting that it chose an easy information domain because the "namespace" of popular music is simple and well-known. The Napster model breaks down in cases where more complex metadata is required to find a given piece of data. For example, in the case of classical music, an artist/song combination is often insufficient, since the same piece may be performed by various combinations of artists.

A related observation, which Darren New of Invisible Worlds (*http://www.invisible.net*) made at the summit, is that Napster depends on the music industry itself to "market its namespace." Without preexisting knowledge of song titles and artists, there is nothing for the Napster user to search for. This will lead to additional centralization layers as unknown artists try to provide additional information to help users find their work. This is much the same thing that happened on the Web, as a class of portals such as Yahoo! grew up to categorize and market information about the peer-to-peer world of hyperlinked web pages.

It's easy to see, then, how understanding and managing namespaces and other forms of metadata becomes central to peer-to-peer applications. What's more, it is also the key to many peer-to-peer business models. Controlling namespaces and resource discovery has turned out to be one of the key battlegrounds of the Web. From Network Solutions (which largely controls DNS registration) to Yahoo! and search engines, identifying and capitalizing on the ways that centralization impacts even radically decentralized systems has turned out to be one key to financial success.

Instant messaging turns out to tell a similar story. The namespace of an instant messaging system, and the mapping of identity onto user addresses, is the key to those systems. You have only to witness the efforts of AOL to keep other instant messaging vendors from reaching its customers to understand just how important this is.

Note, however, that in the end, an open namespace with multiple providers will create a more powerful network than a closed one, just as the open Web trumped closed information services like AOL and MSN. AOL now succeeds for its customers as a "first among equals" rather than as a completely closed system.

In the case of a distributed computation application, metadata might mean some identifier that allows the distributed data elements to be reassembled, and the address of the user who is working on a particular segment. SETI@home tracks user identity as a way of providing a game-like environment in which users and companies compete to contribute the most cycles. Startups aiming to compensate users for their spare compute cycles will need to track how much is contributed.

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Depending on the type of problem to be computed, they might want to know more about the resources being offered, such as the speed of the computer, the amount of available memory, and the bandwidth of the connection. Some of the technical means used to track and reward users are explored in Chapter 16, *Accountability*.

We can see, then, that some of the key battlegrounds for peer-to-peer as a business proposition will be the standards for metadata, the protocols for describing and discovering network-based resources and services, and ownership of the namespaces that are used to identify those resources.

Returning to Napster, though, it's also clear that the core competencies required of successful peer-to-peer projects will include seamless communication and connectivity, facilities that support self-organizing systems, and the management of trust and expectations.

Ultimately, peer-to-peer is about overcoming the barriers to the formation of ad hoc communities, whether of people, of programs, of devices, or of distributed resources. It's about decoupling people, data, and services from specific machines, using redundancy to replace reliability of connections as the key to consistency. If we get it right, peer-to-peer can help to break the IT bottleneck that comes with centralized services. Decentralization and user empowerment enable greater productivity. Edge services allow more effective use of Internet resources.

We're just at the beginning of a process of discovery. To get this right, we'll need a lot of experimentation. But if we can learn lessons from Internet history, we also need to remember to focus on the *interoperability* of many systems, rather than treating this as a winner-takes-all game in which a single vendor can establish the standard for the network platform.

The peer-to-peer landscape is changing daily. New companies, applications, and projects appear faster than they can be catalogued. Especially with all the hype around peer-to-peer, the connections between these projects can be fairly tenuous. Is it marketing buzz or substance, when everyone tries to join the parade?

While there's a danger in casting the net too widely, there's also a danger in limiting it. I believe that the story I've told here gives us a good starting point in understanding an emergent phenomenon: the kind of computing that results when networking is pervasive, resources are abundant (and redundant), and the barriers are low to equal participation by any individual network node.

Microsoft Exhibit 1019

chapter four

The Cornucopia of the Commons

Dan Bricklin, Cocreator of Visicalc

Let's get to the bottom of the Napster phenomenon—why is this music trading service so popular? One could say, trivially, that Napster is successful because you can find what you want (a particular song) and get it easily. It's also pretty obvious that songs are easy to find because so many of them are available through Napster. If Napster let me get only a few popular songs, once I downloaded those I'd lose interest fast.

But what's the root cause? Why are so many songs available? Hint: *It has nothing to do with peer-to-peer.* Peer-to-peer is plumbing, and most people don't care about plumbing. While the "look into other people's computers and copy directly" method has some psychological benefit to people who understand what's going on (as indicated by thinkers such as Tom Matrullo and Dave Winer), I think the peer-to-peer aspects actually get in the way of Napster.

Let's be blunt: Napster would operate much better if, when you logged in, it uploaded all the songs from your disk that weren't already in the Napster database. If the songs were copied to a master server, rather than just the names of the songs and who was currently logged in, the same songs would be available for download provided by the same people, but at all times (not just when the "owner" happened to be connected to the Internet), and probably through more reliable and higher-speed connections to the Internet. (Akamai provides the kind of redundancy and efficiency that Napster currently relies on its worldwide network of users to provide.) Napster could at least maintain the list of who has what songs better than they do now.

Napster doesn't work this way partly because peer-to-peer may be more legal (or so they argue) and harder to litigate against. But other applications may not have

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Napster's legal problems and would therefore benefit from more centralized servers. While I'm a strong proponent for peer-to-peer for some things, I don't think architecture is the main issue driving new services.

The issue is whether you get what you want from the application: "Is the data I want in the database?" What's interesting about Napster is where its data ultimately comes from—the users—not when or how it's transferred. So in this chapter, I'm going to examine how a service can fill a database with lots of whatever people want.

Ways to fill shared databases

There are three common ways to fill a shared database: *organized manual*, *organized mechanical*, and *volunteer manual*.

The classic case of an organized manual database is the original Yahoo! directory. This database was filled by organizing an army of people to put in data manually. Another example is the old legal databases where armies of typists were paid to retype printed material into computers.

The original AltaVista is an example of an organized mechanical database. A program running on powerful computers followed links and domain names and spidered the Web, saving the information as it went. Many databases on the Web today are mechanically created by getting access to somebody else's data, sometimes for a fee. Examples include databases of street maps and the status of airline flights. Some of those databases are by-products of automated processes.

Finally, Usenet newsgroups and threaded discussions like Slashdot are examples of volunteer databases, where interested individuals provide the data because they feel passionate enough about doing so. Amazon.com's well-known reviews are created through a mixture of organized manual and volunteer manual techniques: the company recruits some reviews and readers spontaneously put up others.

CDDB: A case study in how to get a manually created database

The most interesting databases (for the purposes of this chapter) are the ones that involve manual creation. When we look closely at some of them, we find some very clever techniques for getting data that are very specific to the subjects they cover and the users they serve. Let's focus on one service that employs a

Microsoft Exhibit 1019

very unusual technique to aggregate its data: the CDDB service offered by Gracenote to organize information about music CDs (*http://www.cddb.com*).

The CDDB database has information that allows your computer to identify a particular music CD in the CD drive and list its album title and track titles. Their service is used by RealJukebox, MusicMatch, Winamp, and others. What's interesting is how they accumulate this information that so many users rely on without even thinking about it.

Most CDs do not store title information. The only information on the CD, aside from the audio tracks themselves, is the number of tracks (songs) and the length of each one. This is the information your CD player displays. What CDDB does is let the software on your PC take that track information, send a CD signature to CDDB through Internet protocols (if you're connected), and get back the titles.

CDDB works because songs are of relatively random length. The chances are good almost all albums are unique. To understand this point, figure there are about 10 songs on an album, and that they each run from about a minute and a half to about three and a half minutes in length. The times for each song therefore vary by 100 seconds. There are $100 \times 100 \times \ldots \times 100 = 100^{10} = 10^{11} = 100$ billion = an awful lot of possible combinations. So an album is identified by a signature that is a special arithmetic combination of the times of all the tracks.

You'd figure that CDDB just bought a standard database with all the times and titles. Well, there wasn't one. What they did was accept postings over the Internet that contained track timing information and titles typed in by volunteers. Software for playing music CDs on personal computers was developed that let people type in that information if CDDB didn't have it. As people noticed that their albums failed to come up with titles when they played them on their PCs, many cared enough to type in the information. They benefited personally from typing the information because they could then more easily make their own playlists, but in the process they happened also to update the shared database. The database could be built even if only one person was willing to do this for each album (even an obscure album).

If you loved your CD collection, you'd want all the albums represented—or at least some people did. Some people are the type who like to be organized and label everything. Not everybody needed to be this type, just enough people to fill the database. Also, the CDDB site needed this volunteer (user) labor only until the database got big enough that it was valuable enough for other companies to pay for access.

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CDDB is not run on a peer-to-peer architecture. Their database is on dedicated servers that they control. Their web site says:

> CDDB is now a totally secure and reliable service which is provided to users worldwide via a network of high availability, mirrored servers which each have multiple, high bandwidth connections to the Internet... boasting a database of nearly 620,000 album titles and over 7.5 million tracks.

So CDDB succeeded not through peer-to-peer networking—it succeeded by harnessing the energy of its users.

Napster: Harnessing the power of personal selfishness

Napster is a manually created database built on work by volunteers. It gets bigger when one of its users buys (or borrows) a copy of a CD, converts it to MP3, and stores it in his or her shared music directory. It can also be enlarged when somebody creates an MP3 of their own performance that they want to share. But Napster cleverly provides a short-circuit around the process of manually creating data: In both cases, *storing the copy in the shared music directory can be a natural by-product of the user's normal work with the songs.* It can be done as part of downloading songs to a portable music player or burning a personal mix CD. Whenever the users are connected to the Internet and to the Napster server, songs in the shared directory are then available to the world.

Of course, the user may not be connected to the Napster server all the time, so the song is not fully available to all who want it (a perennial problem with peer-to-peer systems). However, Napster overcomes this problem too, by exploiting the everyday activities of its users. Whenever someone downloads a song using Napster and leaves the file in his or her shared music directory, that person is increasing the number of Napster users who have that song, increasing the chances you will find someone with the song logged in to Napster when you want your copy. So again, the value of the database increases through normal use. (The same kind of replication is achieved in a more formal way by Freenet through its unique protocol, but Napster gets the same effect more simply—its protocol is just the decision of a user to do a download.)

The genius of Napster is that increasing the value of the database by adding more information is a natural by-product of each person using the tool for his or her own benefit. No altruistic sharing motives need be present, especially since sharing is the default. It isn't even like the old song about "leaving a cup with water by the pump to let the next person have something to prime it with." (I'll have to use Napster to find that song....) In other words, nobody has to think of being nice to the next guy or put in even a tiny bit of extra effort.

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As Internet analyst Kevin Werbach wrote in Release 1.0, a monthly report on technology trends:

> What made Napster a threat to the record labels was its remarkable growth. That growth resulted from two things: Napster's user experience and its focus on music... What makes Napster different is that it's drop-dead simple to use. Its interface isn't pretty, but it achieves that magic resonance with user expectations that marks the most revolutionary software developments.

I would add that, in using that simple, desirable user interface, you also are *adding to the value of the database without doing any extra work*. I'd like to suggest that one can predict the success of a particular system for building a shared database by how much the database is aided through normal, selfish use.

The commons

We've heard plenty about the tragedy of the commons—in fact, it pops up in several other chapters of this book. In the 1968 essay that popularized the concept, "The Tragedy of the Commons," Garrett Hardin wrote:

> Therein is the tragedy. Each man is locked into a system that compels him to increase his herd without limit—in a world that is limited. Ruin is the destination toward which all men rush, each pursuing his own best interest in a society that believes in the freedom of the commons. Freedom in a commons brings ruin to all.

In the case of certain ingeniously planned services, we find a contrasting *cornucopia of the commons*: use brings overflowing abundance. Peer-to-peer architectures and technologies may have their benefits, but I think the historical lesson is clear: concentrate on what you can get from users, and use whatever protocol can maximize their voluntary contributions. That seems to be where the greatest promise lies for the new kinds of collaborative environments.

Microsoft Exhibit 1019

part two

Projects

This part of the book offers a look at several current systems, giving a sense of what actual peer-to-peer systems look like and how they behave.

Microsoft Exhibit 1019

chapter five

SETI@home

David Anderson, SETI@home

It was January 1986, and I was sitting in a cafe on Berkeley, California's Telegraph Avenue. Looking up, I recognized a student in the graduate course I was teaching that semester at the university. We talked. His name was David Gedye, and he had just arrived from Australia. Our conversation revealed many common interests, both within and outside of computer science. This chance meeting led, twelve years later, to a project that may revolutionize computing and science: SETI@home.

Gedye and I became running partners. Our long forays into the hills above the Berkeley campus occasioned many far-ranging discussions about the universe and our imperfect understanding of it. I enjoyed these times. But all good things must end, and in 1989 Gedye left Berkeley with a master's degree. He worked in Silicon Valley for a few years, then moved to Seattle and started a family. I also left academia, but remained in the Bay Area.

In 1995 Gedye visited me in Berkeley, and we returned to the hills, this time for a leisurely walk. He was bursting with excitement about a new idea. It sounded crazy at first: He proposed using the computing power of home PCs to search for radio signals from extraterrestrial civilizations. But Gedye was serious. He had contacted Woody Sullivan, an astronomy professor at the University of Washington and an expert in the theory behind SETI, the Search for Extraterrestrial Intelligence. Woody had steered him to Dan Werthimer, a SETI researcher at UC Berkeley.

The four of us—Gedye, Werthimer, Sullivan, and I—met several times over the next year, trying to assess the viability of Gedye's idea. We decided that existing technology was sufficient, though just barely, for recording radio data and distributing it over the Internet. And if we managed to get 100,000 people to participate, the aggregate computing power would let us search for fainter signals,

Microsoft Exhibit 1019

and more types of signals, than had ever been done before. But could we get that many people interested? We decided to try it and find out.

Radio SETI

SETI is a scientific research area whose goal is to detect intelligent life outside the Earth. In 1959, Phil Morrison and Giuseppe Cocconi proposed listening for signals with narrow frequency bandwidth, like our own television and radar emissions, but unlike the noise emanating from stars and other natural sources. Such signals would be evidence of technology, and therefore of life.

The first radio SETI experiment was conducted in 1960 by Frank Drake, who pointed an 85-foot radio telescope in West Virginia at two nearby stars. Drake didn't detect an extraterrestrial signal, but he and other researchers have continued to listen. Since 1960 there have been tremendous advances in technology, especially in the digital technology at the heart of radio SETI. The systems that analyze radio signals use the Fast Fourier Transform (FFT), an algorithm that divides signals into their component frequencies. Most SETI projects have built special-purpose FFT supercomputers, but are limited to fairly simple types of analysis.

There are also larger and more sensitive radio telescopes. The largest is Arecibo, a 1,000-foot aluminum dish set into a natural hollow in the hills of northern Puerto Rico. A movable antenna platform is suspended 700 feet above the center of the dish. By moving the antenna, one can effectively point the telescope anywhere in a band of sky from the celestial equator to 38 degrees north. The telescope doesn't form an image like optical telescopes. It's more like a highly directional microphone. It sees a fuzzy disk (a *beam*) about 1/10 of a degree in diameter, or about 1/5 the diameter of the moon.

Arecibo's size and excellent electronics let it hear very faint signals. The telescope is used for many scientific purposes: looking for pulsars, imaging asteroids and planets by bouncing radio waves off them, and studying the upper atmosphere. Observation time on Arecibo is a precious commodity.

In 1992, Dan Werthimer devised a way for his SETI project, SERENDIP, to use Arecibo all the time—even while other projects are using it. He mounted a secondary antenna at the opposite end of the platform from the main antenna. While the main antenna tracks a fixed point in the sky (as it normally does) this secondary antenna moves slowly in an arc about 6 degrees away. SERENDIP observers have no control over where the scope points, but over long periods of time their beam covers the entire band of sky visible from Arecibo. SERENDIP is

Microsoft Exhibit 1019

thus a *sky survey*: It covers lots of stars but doesn't spend much time on each star. Other radio SETI projects use *targeted search*: they look at specific stars for longer periods, which gives them more sensitivity. No one knows which approach is better, or even if radio signals are the right thing to look for. The best bet, SETI experts agree, is to try everything.

How SETI@home works

We decided that SETI@home would use SERENDIP's antenna. Like all previous radio SETI projects, SERENDIP analyzes its signal using a dedicated supercomputer at the telescope; it doesn't record the signal. For SETI@home, we needed to digitally record the signal and transport it to our computers at Berkeley. The network connection from Arecibo to the mainland is too slow. Instead, we record the data on digital tapes and mail them to Berkeley. The largest-capacity digital tape available in 1998 was the 35-GB digital linear tape (DLT).

We had to decide what frequency range to record. Covering a wide range is good from a scientific point of view, but it means more tapes and more network bandwidth. We decided to record a 2.5 MHz frequency band. Using 1-bit samples, this gives a data rate of 5 Mbps, meaning that a tape fills up in about 16 hours. Like most radio SETI projects, we centered our band at the *hydrogen line*, 1.42 GHz. This is the resonant frequency of the hydrogen molecules that fill interstellar space. Since hydrogen is the most abundant element in the universe, we hope that if aliens are sending an intentional signal, they will use this frequency. Our 2.5 MHz band is wide enough to contain Doppler shifts (frequency shifts due to relative motion) corresponding to any likely velocity of a transmitter in our galaxy.

SETI@home and SERENDIP are complementary: SETI@home looks at a narrower frequency range than SERENDIP (2.5 MHz versus 140 MHz) but does better signal analysis. SETI@home will record data for two years, during which time we'll cover Arecibo's visible band about four times.

Every week about ten newly-recorded tapes arrive from Arecibo. These tapes are catalogued and stored. Next, the data is divided into *work units*, the pieces that are sent to clients. The data is divided along two dimensions: time and frequency. We decided that work units should be about 0.3 MB—large enough to keep a computer busy for a while, but small enough so that, even over a 28.8-Kbps modem, the transmission time is only a few minutes. We wanted each work unit to cover several times the *beam period* (the time it takes for the beam to move across a point in the sky, typically about 20 seconds). To accomplish this, we divide the data into 256 frequency bands, each about 10 KHz wide. We

Microsoft Exhibit 1019

then slice each band into pieces 256,000 samples long—about 107 seconds of recording time. Work units in a given band are overlapped in time by 20 seconds, ensuring that each beam period is contained entirely in at least one work unit.

The task of splitting data tapes into work units is itself computationally intensive—enough so that we considered making it a distributed task unto itself. In the end we assembled a group of six workstations, each equipped with a DLT tape drive, running the splitter program full-time.

Work units are stored on a computer with about 300 GB of disk space. Ideally, each work unit should remain in storage until a result for it has been returned. However, with 50 GB of data pouring in from Arecibo every day, and with some slow computers taking a week or more to complete a work unit, this can lead to a situation where we run out of space for new work units. Our current policy is to delete work units even if no result has been returned yet.

A relational database keeps track of everything: tapes, work units, results, users, and so on. This database has grown to several hundred gigabytes, and we place a tremendous load on it. Although we have spread it across two large server machines, it is frequently a performance bottleneck.

The most visible component of SETI@home is the client program. For Windows and Macintosh users, this program is a screensaver: it only does its work when the computer isn't being used. The client sets up an Internet connection to the SETI@home data distribution server, obtains a work unit, and closes the connection. It then processes the data; this may take anywhere from an hour to several days, depending on the speed of the computer. When it's finished, the client reconnects to the server, sends back the results, and gets a new work unit. Every few minutes the program writes a "checkpoint" file to disk, so that it can pick up where it left off in case the user turns off the computer.

The SETI@home data distribution server accepts connections from clients, collects their results, and sends them new data. The data server may send either new work units or previously sent work units that are still on disk. Many connections may arrive each second, and it may take several minutes (e.g., over a modem connection) to handle a request. So the server uses a large number of processes; in many respects it is like a web server such as Apache. The server system also uses several other programs, such as a "garbage collector" that removes work units for which results have been received.

If a transmitter and/or receiver is accelerated (e.g., because of planetary rotation or orbit), a signal sent at a constant frequency will be heard as drifting in frequency. SETI@home uses a technique called "coherent integration" for detect-

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ing drifting signals. SETI@home examines about 50,000 drift rates, ranging from –50 to +50 Hz/sec. For each drift rate, the client program transforms the data to remove the drift and then looks for signals at constant frequency. This gives about 10 times better sensitivity than looking for drifting signals directly.

For a given drift rate, the program uses 15 different FFT lengths, or frequency resolutions. A mathematical theorem called the Heisenberg Uncertainty Principle says that you can examine a signal with high frequency resolution or high time resolution, but not both. Since we don't know what characteristics an alien signal might have, we explore the full range of this trade-off.

For a given drift rate and FFT length, the program computes the time-varying power spectrum of the signal. This produces an array whose dimensions are time and frequency, and whose value is the power (the SETI@home graphics show a 3-D color graph of this array). The power array is analyzed, looking for several types of signals:

Spikes
: Power values much higher than the local average.

Gaussians
: Ridges in the data, along the time axis, whose shape matches the bell-shaped curve (called a "Gaussian") of the telescope beam.

Pulses
: Signals at a constant frequency that cycle on and off, with a Gaussian envelope. The pulse rate, phase, and duty cycle are not known in advance. We use an algorithm called Fast Folding, originally developed for finding pulsars, that efficiently covers a wide range of possibilities.

Triplets
: Groups of three evenly spaced spikes at the same frequency.

Signals that exceed predefined thresholds are returned to the server and added to the database. The client doesn't have a flashing light that goes off when an ET signal is found; this isn't possible. Man-made "radio frequency interference" (RFI), coming from TV stations, cell phones, and car ignitions, leaks into the radio telescope and is often indistinguishable from an ET signal. RFI rejection is a hard problem for radio SETI. Our approach is to check our database of candidate signals for two or more signals at the same frequency and sky position, but at different times. Man-made interference changes from one month to the next, but (hopefully) alien signals will remain unchanged.

So SETI@home's detection of an extraterrestrial signal, if it happens, will show up first on a computer screen at Berkeley, sometime towards the end of the

Microsoft Exhibit 1019

project. But our database will have a complete record of the users whose PCs contributed to the detection, and they'll share in the credit.

SETI@home's web site (*http://setiathome.berkeley.edu*) plays an important role in the project. At the site, people can download the client program, learn about SETI@home and radio SETI, and create and join teams. The web site also shows current statistics and "leader boards:" lists of users and teams, ordered by number of work units completed. These pages are generated by programs that obtain the latest information from the database.

Trials and tribulations

SETI@home has faced many difficulties and challenges. Server performance, for example, has been a major problem. As more and more people downloaded and ran the client, the stream of client requests grew from a trickle to a torrent. At first, our server system consisted of three pieces: an Informix database server, the data distribution server, and an Apache web server. These ran on three Sun workstations, which also served as our personal computers.

In the first week the server system quickly was overwhelmed. Client connections were being turned away, resulting in irritating error messages being displayed to users, and hence a torrent of email.

We scrambled to fix these problems by modifying the software. For example, we realized that much of the load on the database server was due to updating lots of accounting records (for countries, CPU types, teams, etc.) for each result received. We hastily revised the system to update the accounting records offline, combining thousands of database writes into a single write. This offline system quickly fell behind, producing yet another wave of irate email, but at least the data distribution server now kept up.

It quickly became clear that we needed more powerful server hardware. Sun Microsystems came to our rescue, and over the next year they donated several of their high-performance server machines. Even with these improvements, server performance continues to be an issue. Resources in general, especially funding and manpower, have been a problem. We've received funds from a variety of private donors and a grant from the University of California. This money has been enough to hire about three full-time employees. A project of similar magnitude in the private sector would probably employ 20 or 30 people. We've had to cut corners in many areas (for example, there is no customer support), and some tasks have fallen far behind schedule.

Microsoft Exhibit 1019

Another problem area involved processor-specific optimizations. The SETI@home client is written in C++, and we compile it using standard compilers such as Microsoft VC++ and Gnu's *gcc*. Performance-conscious users disassembled the inner loops of the program and figured out that it was doing FFTs and that the code was non-optimal on many processors. For example, several variants of the x86 architecture, such as AMD's 3DNow, have instructions that can do FFTs faster. This led to demands from 3DNow enthusiasts that we release a version optimized for 3DNow. Similar requests came from Altivec, MIPS, and Alpha owners.

We didn't have the manpower to maintain lots of processor-specific versions of the code. However, several people figured out how to replace the FFT routine at the heart of SETI@home with a faster routine. Some of them did this incorrectly, producing clients that returned incorrect results.

Doctored versions of the program were just one of many security challenges. Most of the problems involved "credit cheating" by, for example, returning the same result file over and over. People also doctored their result files, making it appear that their computers had found a strong signal. It's not clear what motivated these activities—after all, there are no financial rewards for work done. We invested a large amount of effort in making a more secure version of the client, which uses cryptographic checksumming to detect tampering with result files and with the program itself.

Some people feel that SETI@home should be an "open source" project, that we should distribute the source code and solicit the help of volunteer programmers to fix bugs and make enhancements. Indeed, we tried this for a short period and (perhaps due to our inexperience managing open source projects) were quickly inundated with code that, for various reasons, was unusable. We were also concerned that someone might substitute their own signal detection algorithm, announce a signal discovery, and destroy our project's credibility. When we launched the project as non–open source, a vocal group of critics created a web site calling for a boycott of SETI@home and attacking us for not being "free software." (Many people have interpreted this as meaning that we charge money for the client software, which is not the case.)

Human factors

Early in 1998, we launched a SETI@home web site describing the idea and letting people sign up. It was a good time to start a project like SETI@home. Public interest in SETI had been stirred by the movie *Contact*, which was released in

Microsoft Exhibit 1019

July 1997. This movie, based on a novel by Carl Sagan, describes radio SETI in reasonably accurate terms, and parts of it were filmed at Arecibo.

It became clear that there would be no shortage of participants—over 400,000 people signed up at the web site. After a long period of development and testing, we released the client software on May 17, 1999. In the first week after the launch, over 200,000 people downloaded and ran the client. This number has grown to 2,400,000 as of October 2000. People in 226 countries around the world run SETI@home. 50% of them are outside the U.S.; there are even 73 in Antarctica.

People have helped SETI@home in every way imaginable. People upgrade their computers, or buy new computers, just to run SETI@home faster. In Europe, people run SETI@home in spite of expensive Internet connection setup charges. Volunteers translated the web site into about 30 foreign languages. A number of people have written programs that track their work in elaborate detail. Graphic artists sent us dozens of banner and link graphics; one of these was so attractive that it replaced Gedye's original planet-and-wave image (which he threw together in PowerPoint) as our logo.

When it became clear that SETI@home was being widely embraced by the public, several questions arose. How was the word about SETI@home being spread? Why were people running SETI@home? Were they leaving their computers on longer, or buying faster computers, because of SETI@home?

We've heard the following "viral marketing" scenario from many sources: one person in an office starts running SETI@home; people see the screensaver graphics, ask about it, hear the explanation of the project, and try it themselves. Soon the entire office is running it.

In search of more quantitative information, we ran a poll on our web site, with questions involving demographics and attitudes about SETI and distributed computing. Some of the results were surprising; for example, only 7% of the respondents are female. We learned that our users are sober in their expectations: Only 10% think that a signal will be detected within the two-year duration of the project.

The world's most powerful computer

Scientific computations are often measured in units of *floating-point operations*—additions and multiplications of numbers with fractional parts, like 42.0 or 3.14159. A common unit of supercomputer speed is trillions of floating-point operations per second, or TFLOPS.

Microsoft Exhibit 1019

The 1.0 TFLOPS barrier has been broken only in the last year or so. The fastest supercomputer is currently the ASCI White, built by IBM for the U.S. Department of Energy. It costs $110 million, weighs 106 tons, and has a peak performance of 12.3 TFLOPS.

SETI@home is faster than ASCI White, at less than 1% of the cost. The FFT computations for each SETI@home work unit require 3.1 trillion floating-point operations. In a typical day, SETI@home clients process about 700,000 work units. This works out to over 20 TFLOPS. It has cost about $500,000, plus another $200,000 or so in donated hardware, to develop SETI@home and operate it for a year. Of course, the cost of the one million PCs running SETI@home greatly exceeds that of ASCI White—but these PCs were bought and paid for before SETI@home and would exist even without it.

As of October 2000, SETI@home has received 200 million results, for a total of 4×10^{20} floating-point operations. We believe that this is the largest computation ever performed. And in terms of the potential of the Internet for scientific computing, SETI@home is the tip of the iceberg. There are projected to be one billion Internet-connected computers by 2003. If 10% of them participate in distributed computing projects, there will be enough computing power for 100 projects the size of SETI@home.

To what range of problems is this power applicable? Certainly not all problems. It must be possible to factor the problem into a large number of pieces that can be handled in parallel, with few or no interdependencies between the pieces. The ratio between communication and computation must be fairly low: for example, it mustn't take an hour to transfer the data for one second of computing.

Surprisingly many problems meet these criteria. Some of them, such as mathematical problems, are of academic interest; others are in areas of commercial importance, such as genetic analysis. The range of feasible problems will increase along with communication speed and capacity; for example, it may soon be feasible to do computer graphics rendering for movies.

The peer-to-peer paradigm

In the brief history of computer technology, there have been several stages in the way computer systems are structured. The dominant paradigm today is called *client/server*: Information is concentrated in centrally located *server* computers and distributed through networks to *client* computers that act primarily as user interface devices. Client/server is a successor to the earlier *desktop computing* and *mainframe* paradigms.

Microsoft Exhibit 1019

Today's typical personal computer has a very fast processor, lots of unused disk space, and the ability to send data on the Internet—the same capabilities required of server computers. The sheer quantity of Internet-connected computers suggests a new paradigm in which tasks currently handled by central servers (such as supercomputing and data serving) are spread across large numbers of personal computers. In effect, the personal computer acts as both client and server. This new paradigm has been dubbed *peer-to-peer* (P2P). SETI@home and Napster (a program, released about the same time as SETI@home, that allows people to share sound files over the Internet) are often cited as the first major examples of P2P systems.

The huge number of computers participating in a P2P system can overcome the fact that individual computers may be only sporadically available (i.e., their owners may turn them off or disconnect them from the Internet). Software techniques such as data replication can combine a large number of slow, unreliable components into a fast, highly reliable system.

The P2P paradigm has a human as well as a technical side—it shifts power, and therefore control, away from organizations and toward individuals. This might lead, for example, to a music distribution system that efficiently matches musicians and listeners, eliminating the dilution and homogenization of mass marketing. For scientific computing, it could contribute to a democratization of science: a research project that needs massive supercomputing will have to explain its research to the public and argue the merit of the research. This, I believe, is a worthwhile goal and will be a significant accomplishment for SETI@home even if no extraterrestrial signal is found.

Microsoft Exhibit 1019

chapter six

Jabber

Conversational Technologies

Jeremie Miller, Jabber

Conversations are an important part of our daily lives. For most people, in fact, they are the most important way to acquire and spread knowledge during a normal working day.

Conversations provide a comfortable medium in which knowledge flows in both directions, and where contributors share an inherent context through their subjects and relationships. In addition to old forms of conversations—direct interaction and communication over the phone and in person—conversations are becoming an increasingly important part of the networked world. Witness the popularity of email, chat, and instant messaging, which enable users to increase the range and scope of their conversations to reach those that they may not have before.

Still, little attention has been paid in recent years to the popular Internet channels that most naturally support conversations. Instead, most people see the Web as the driving force, and they view it as a content delivery platform rather than as a place for exchanges among equals. The dominance of the Web has come about because it has succeeded in becoming a fundamentally unifying technology that provides access to content in all forms and formats. However, it tends toward being a traditional one-way broadcast medium, with the largest base of users being passive recipients of content.

Conversations have a stubborn way of reemerging in any human activity, however. Recently, much of the excitement and buzz around the Web have centered on sites that use it as a conversational medium. These conversations take place within a particular web site (Slashdot, eBay, Amazon.com) or an application (Napster, AIM/ICQ, Netshow).

Microsoft Exhibit 1019

And repeating the history of the pre-Web Internet, the new conversations sprout up in a disjointed, chaotic variety where the left hand doesn't know what the right hand is doing. The Web was a godsend for lowering the barrier to access information; it increased the value of all content by unifying the technologies that described and delivered that content. In the same way, Internet conversations stand to benefit significantly by the introduction of a common platform designed to support the rich dynamic and flexible nature of a conversation.

Jabber could well become this platform. It's not a single application (although Jabber clients can be downloaded and used right now) nor even a protocol. Instead, using XML, Jabber serves as a glue that can tie together an unlimited range of applications that tie together people and services. Thus, it will support and encourage the growth of diverse conversational systems—and this moment in Internet history is a ripe one for such innovations.

Conversations and peers

So what really is a conversation? A quick search using Dictionary.com reveals the following:

> con·ver·sa·tion (kän-ver-'sā-shən) *n.* 1. A spoken exchange of thoughts, opinions, and feelings; a talk. 2. An informal discussion of a matter by representatives of governments, institutions, or organizations. 3. *Computer Science.* A real-time interaction with a computer.

Essentially, a conversation is the rapid transfer of information between two or more parties. A conversation is usually characterized by three simple traits: it happens spontaneously, it is transient (lasting a short time), and it occurs among peers—that is, all sides are equal contributors.

Let's turn then to the last trait. The term "peer" is defined by Dictionary.com:

> peer (pîr) *n.* 1. A person who has equal standing with another or others, as in rank, class, or age; *children who are easily influenced by their peers.*

The Internet expands this definition to include both people (P) and applications (A). Inherently, when peers exchange information, it is a conversation, since both sides are equal and are transiently exchanging information with each other. Person-to-person conversations (P-P) include email, chat, and message boards. But crucial conversations also include application-to-application (A-A) ones such as web services, IP routing, and UUCP. Least common, but most intriguing for future possibilities, are person-to-application (P-A) conversations such as smart agents and bots.

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It's interesting to take a step back and look at the existing conversations happening on the Internet today. How well does each technology map to the kind of natural conversational style we know from real life? Let's identify a few important metrics to help evaluate these traditional forms of Internet communication as conversational channels:

Time
: The more rapidly messages can be created and delivered, and the more rapidly the recipient can respond, the more productive the conversation is for both participants.

P-A
: A technology provides greater potential for future innovation if it inherently supports applications as well as people.

Peers
: Participants in a conversation should be equal and the conversation bidirectional.

Distributed
: Conversations may be constrained if there is a central form of control or authority.

We can now evaluate a few technologies along some of the metrics just defined.

Email comes to mind first as the most popular form of conversation now happening on the Internet. It is relatively fast, each message taking typically between 30 seconds and a few days to deliver, but certainly not real-time. It is predominantly P-P, with some P-A applications, but it is not a very natural use for A-A, because it provides no structure for content. *Usenet* is similar to email but is focused on group discussions. Both are innately distributed, and participants are peers.

Internet Relay Chat (IRC) is a very popular conversational medium, primarily supporting real-time group discussions. As with email, it's primarily P-P with some P-A and very little A-A. Participants are peers. IRC is a distributed application within a network of groups, but it is restricted to that particular network—it does not extend beyond a single collection of groups.

The traditional *Web* is real-time, but in a strict sense it does not support conversations, because the participants are not peers. The content may be produced by a person, but it has a natural flow in only one direction. Applications that support conversations can be built and made available on the Web, but they are pretty rigid—each conversation is specific and centralized to that application.

Microsoft Exhibit 1019

The *next-generation Web*—also called the *Two-Way Web* by visionary developer Dave Winer—is represented by Microsoft's .NET; and it tries to solve the shortcomings in the evolution of the Web. It involves personal/fractional-horsepower (specialized) HTTP and DAV servers. These systems more naturally support peers and conversations than the traditional Web, but the conversations between these peers are still predominantly one-way (consumer or producer) and are often centralized based on the application or content.

Traditional *instant messaging* services, such as AOL Instant Messenger, ICQ, Yahoo! Messenger, and MSN Messenger, come the closest to a real-world conversation yet, and that is the reason for their soaring popularity. They unfortunately focus primarily on P-P. The most significant drawback is that they are commercial and completely centralized around a single closed service. You must be part of the service to communicate with others on it.

None of these existing technologies provides a common platform for Internet conversations as the Web does for content. Each is either limited in some important dimension or is specific to one application.

What could people do with an ideal, standardized conversational platform open to applications that can cross boundaries and access end user content? Here are some fanciful future possibilities:

- I could ask a coworker's word processor or source editor what documents they are editing and discuss revisions.
- My spell checker could ask the entire department to check the validity of unknown acronyms and project or employee names.
- Instead of trying to combine the details of everybody's lives in a central address book or schedule, each application that needs to discover this information could ask other peers for it. Different conversations could be with different communities I define, such as my department, my family (for holiday card or birthday lists), or my friends (for event invitations).
- My television set or video recorder could ask my friends what programs they are watching and use their recorders' extra space to save the programs in case I want to watch them too. With broadband, the television sets could have a conversation exchanging the actual video.
- My games could exchange scores and playing levels with my friends' games and schedule times to play collaboratively (possibly invoking some of the other peers above to schedule conversations). I could also ask another game to deliver an important message or to join a game.

Microsoft Exhibit 1019

- Businesses could reproduce some of the warmth and responsiveness of a phone conversation online, replacing the cold, faceless e-commerce store or customer support site that serves to drive us to our phones. The new sites could combine a rich context and content with the kind of conversational medium we all like to have.

Evolving toward the ideal

A look back at a bit of the World Wide Web's brief history proves quite interesting and enlightening. Back in its pioneering days, the Web was idealized as a revolutionary peer platform that would enable anyone on the Internet to become a publisher and editor. It empowered individuals to publish their unique collections of knowledge so that they were accessible by anyone. The vision was of a worldwide conversation where everyone could be both a voice and a resource. Here are a few quotes from Tim Berners-Lee to pique your interest:

> The World Wide Web was designed originally as an interactive world of shared information through which people could communicate with each other and with machines (*http://www.w3.org/People/Berners-Lee/1996/ppf.html*).

> I had (and still have) a dream that the web could be less of a television channel and more of an interactive sea of shared knowledge. I imagine it immersing us as a warm, friendly environment made of the things we and our friends have seen, heard, believe or have figured out. I would like it to bring our friends and colleagues closer, in that by working on this knowledge together we can come to better understandings (*http://www.w3.org/Talks/9510_Bush/Talk.html*).

Although the Web fulfills this vision for many people, it has quickly evolved into a traditional consumer/producer relationship. If it had instead evolved as intended, we might be in a different world today. Instead of passively receiving content, we might be empowered individuals collectively producing content, publishing parts of ourselves online to our family and friends, and collectively editing the shared knowledge within our communities.

So where did it go wrong in this respect? It could be argued that the problem was technological, in that the available tools were browsing-centric, and it wasn't easy to become an editor or publisher. A more thought-provoking answer might be that the problem was social, in that there was little demand for those empowering tools. Perhaps only a few people were ready to become individual publishers, and the rest of society wasn't ready to take that step.

Microsoft Exhibit 1019

The Web did not stagnate, however. It continued to evolve from a content distribution medium to an application distribution medium. Few users are publishing content, but a huge number of companies, groups, and talented individuals are building dynamic applications with new characteristics that reach beyond the original design of the Web. The most exciting of these exhibit characteristics of a peer medium and empower individuals to become producers as well as consumers. Examples include eBay, Slashdot, IMDB, and MP3.com. Although the applications provide a new medium for conversations between P-P peers, the mechanisms for doing so are application-specific. These new web-driven peer applications also have the drawbacks of being centralized, of not being real-time in the sense of a conversation, and of requiring their own form of internal addressing.

So instead of the Web being used primarily as a peer publishing medium, it has become a client/server application medium upon which a breed of peer applications are being built.

Elsewhere in the computer field we can find still other examples of systems that are incorporating greater interactivity. Existing desktop applications are evolving in that direction. They are becoming Internet-aware as they face competition from web sites, so that they can take advantage of the Internet in order to remain competitive and provide utility to the user. Thus, they are evolving from static, standalone, self-contained applications into dynamic, networked, componentized services.

Microsoft, recognizing the importance of staying competitive with online services, is pushing the evolution of desktop applications with their .NET endeavor. By turning applications into networked services, .NET blurs the lines even further between the desktop and the Internet.

The evolution of the Web and the desktop shows a definite trend towards applications becoming peers and having conversations with other applications, services, and people. The common language of conversations in both mediums is XML. As a way of providing a hierarchical structure and a meaningful context for data, XML is being adopted worldwide as the *de facto* language for moving this data between disparate applications. As Tim Bray puts it, "XML is the ASCII of the future."

Jabber is created

To fully realize the potential for unifying the conversations ranging throughout the Internet today, and enabling applications and services to run on top of a

Microsoft Exhibit 1019

common platform, a community of developers worldwide has developed a set of technologies collectively known as Jabber (*http://jabber.org*). Jabber was designed from the get-go for peer conversations, both P-P and particularly A-A, and for real-time as well as asynchronous/offline conversations. Jabber is fully distributed, while allowing a corporation or service to manage its own namespace. Its design is a response to the popularity of the closed IM services. We are trying to create a simple and manageable platform that offers the conversational traits described earlier in this chapter, traits that none of the existing systems come close to providing in full.

Jabber began in early 1998 out of a desire to create a truly open, distributed platform for instant messaging and to break free from the centralized, commercial IM services. The design began with XML, which we exploited for its extensibility and for its ability to encapsulate data, which lowers the barrier to accessing it. The use of XML is pervasive across Jabber, allowing new protocols to be transparently implemented on top of a deployed network of servers and applications. XML is used for the native protocol, translated to other formats as necessary in order to communicate between Jabber applications and other messaging protocols.

The Jabber project emerged from that early open collaboration of numerous individuals and companies worldwide. The name Jabber symbolizes its existence as numerous independent projects sharing common goals, each building a part of the overall architecture. These projects include:

- A modular open source server written in C
- Numerous open source and commercial clients for nearly every platform
- Gateways to most existing IM services and Internet messaging protocols
- Libraries for nearly every programming language
- Specialized agents and services such as RSS and language translations

Jabber is simply a set of common technologies that all of these projects agree on collaboratively when building tools for peer-to-peer systems. One important focus of Jabber is to empower conversations between both people *and* applications.

The Jabber team hopes to create an open medium in which the user has choice and flexibility in the software used to manage conversations, instead of being hindered by the features provided by a closed, commercial service. We hope to accelerate the development of peer applications built on an open foundation, by enabling them to have intelligent conversations with other people and applications, and by providing a common underlying foundation that facilitates conversations and the accessibility of dynamic data from different services.

Microsoft Exhibit 1019

The centrality of XML

Fundamentally, Jabber enables software to have conversations in XML. When people use Jabber-based software as a messaging platform to have conversations with other people, data exchanges use XML under the surface. Applications use Jabber as an XML storage and exchange service on behalf of their users.

XML is not only the core format for encoding data in Jabber; it is also the protocol, the transport layer between peers, the storage format, and the internal data model within most applications. XML permeates every conversation.

The Jabber architecture is also aware of XML namespaces, which permit different groups of people to define different sets of XML tags to represent data. Thus, using a namespace, one group (Dublin Core) has developed a set of tags for talking about the titles, authors, and other elements of a document. Another group might define a namespace for describing music. An instant messaging community using Jabber could combine the two namespaces to exchange information on books about music. Chapter 13, *Metadata*, looks at the promise of Dublin Core and other namespaces for peer-to-peer applications.

Here is a simple message using Jabber's XML format:

```
<message to="hamlet@denmark" from="horatio@denmark" type="chat">
  <body>Here, sweet lord, at your service.</body>
</message>
```

And here's a hypothetical message with additional data in a namespace included:

```
<message to="horatio@denmark" from="hamlet@denmark">
  <body>Angels and Ministers of Grace, defend us!</body>
  <prayer xmlns="http://www.grace.org">
    <verse>...</verse>
  </prayer>
</message>
```

By supporting namespaces, Jabber enables the inclusion of any XML data in any namespace anywhere within the conversation. This allows applications and services to include, intercept, and modify their own XML data at any point. Jabber is thus reduced to serving as a conduit between peers. Ironically, this lowly status provides the power that Jabber offers to Internet conversations.

Pieces of the infrastructure

While the goal of Jabber is to support other naming conventions and protocols, rather than to create brand-new ones, it depends on certain new concepts that require new types of syntax and binding technologies. These help create a common architecture.

Microsoft Exhibit 1019

Identity

Naming is at the heart of any system—each resource must have a unique identity. In Jabber, each resource is identified by a three-part name consisting of a *user*, a *server*, and a *resource*.

The user is often an individual, and the server is a system that runs a Jabber-based application. In a name, the user and server are formatted just like email, *user@server*. This provides a general way to pass identification between people that is already well understood and socially accepted. Since the server resolves the username, the format also allows a user's identity to be managed by a service or corporation the way America Online and Napster manage their usernames. This is an important point for Internet services that are providing a public utility to consumers or companies, and especially for corporations that want to or are required to manage their identities very carefully. This also allows any user to use a third party, such as Dynamic DNS Network Services (*http://dyndns.org*), for transient access to a permanent hostname so as not to be forced to rely on someone else's identity.

The server component of the identity could also provide a community aspect to naming, as it may be shared between a small group of friends, a family, or a special interest group. The name then stands out and identifies the user's relationship as part of that community.

The third part of the identity is the resource. As in a Unix filename or URL, the resource follows the server and is delimited by a slash, as in *user@server/resource*. Outside Jabber, the name is formatted like a combination of an email address and a web URL: *jabber://user@server/resource/data*.

This third aspect of the identity, the resource, allows any Jabber application to provide public access to any data within itself, analogous to a web server providing access to any file it can serve. It also serves to identify different applications that might be operating for a single user. For example, my Jabber ID is *jer@jabber.org*, and when I'm online at home my client application might be identified as *jer@jabber.org/desktop*.

Presence

Presence is a concept fundamental to conversations, because it supports the arbitrary coming and going of participants. Technically, presence is simply a state that a user or application is in. Traditional states in instant messaging include online, offline, and somewhere in between (away, do not disturb, sleeping, etc.). The Jabber architecture automatically manages presence information for users and applications, distributing the information as needed while strictly protecting

Microsoft Exhibit 1019

privacy. It is often this single characteristic that adds the most value to the peers in a conversation: just knowing that the other peer is available to have a conversation.

Presence can go beyond simple online/offline state information. XML could be used to convey location, activity, and contextual (work/project) or application-specific data. Presence information itself provides an inherent context for P-P conversations, as well as status and location context for A-A conversations.

Here is a simple presence example in XML:

```
<presence from="hamlet@denmark">
  <show>away</show>
  <status>Gone to England</status>
</presence>
```

Roster

Another powerful feature of a traditional instant messaging service is the *buddy list* or *roster*. The importance of this list is often underestimated. It is a valuable part of the user's reality that they've stored and made available to their applications.

In social terms, each user's roster is his or her community. It defines the participants in this community or relationships to larger communities. A roster is an actualization of personal trust and relationships with peers. Applications should use this list intelligently to share their functionality and filter conversations.

The circle of trust in which a user has chosen to include his or her computer is a starting point for applications to locate other devices the user utilizes. It should also be used for choosing to collaborate with the resources available from trusted peers. This single, simple feature begins to open the door to the future possibilities mentioned near the beginning of this chapter, and it forms a step toward the warm, friendly environment envisioned by Tim Berners-Lee for the World Wide Web.

Architecture

The Jabber architecture closely resembles email. Peers are connected and route data in a chain until it reaches the desired recipient. A client is connected to its server only, and its server is responsible for negotiating the delivery and receipt of that client's data with other servers or networks using whatever protocol is available. All data within the architecture is processed immediately and passed on to the next peer, or stored offline for immediate delivery once that peer is available again.

Microsoft Exhibit 1019

Peers can play traditional client and server roles within the Jabber architecture. Every server acts as a peer with respect to another server, using SRV DNS records to locate the actual server. Servers also use hostname dialback, independently contacting the sending server to validate incoming data. This prevents spoofing and helps ensure an overall more reliable and secure trust system.

All clients are peers with respect to other clients, and, after establishing a conversation with their servers, are able to establish real-time conversations in XML with any other client. Clients can also include or embed a server internally so that they can operate in any role and provide additional flexibility and security.

Protocols

Along with support for all major instant messaging services (AIM, ICQ, MSN, Yahoo!), Jabber is also protocol agnostic. It uses a variety of applications between the endpoints of the conversations to transparently translate the XML data to and from another protocol. In its immediate applications, Jabber's translation capabilities let it support P-P relationships across traditional instant messaging services, IRC, and email. But the same flexibility also allows the construction of A-A bridges, such as transparent access to SIP, IMXP, and PAM applications, as well as access to Jabber's native presence and messaging functionality from those protocols.

Finally, the protocol-agnostic design of Jabber allows it to participate in the exciting evolution of the Web mentioned earlier in the section "Evolving toward the ideal": An evolution including such technologies as WebDAV, the use of XML over HTTP in the SOAP protocol, the RSS service that broadcasts information about available content, and other web services. We hope to set up revolving door access so that HTTP applications can access native Jabber functionality and so that Jabber applications can transparently access conversations happening over HTTP.

Browsing

A recent addition to Jabber is browsing, which is similar to the feature of the same name in the Network Neighborhood on Microsoft systems. Browsing lets users retrieve lists of peers from other peers and establish relationships between peers. It can be used to see what services might be available from a server, as well as what applications and paths of communication a user has made available to other users and their applications.

Peers that a user might make available could include their normal instant messaging client (home, work, laptop, etc.), a pager transport, an offline inbox, a

Microsoft Exhibit 1019

cell phone, a PDA, a TV, a scheduling application, a 3-D game, or a word processor. Additionally, XML information can be made browsable by a user or application, so that a user's vCard (verification information), public key, personal recipes, music list, bookmarks, or other XML information could be read by both people and applications. Browsing also allows people and applications to locate public peers, such as other messaging gateways mentioned earlier, web services, group chats, and agents (searching, translation, fortune, announcements, Eliza).

Conversation management

By centralizing and coordinating all of your conversations via a central identity, the software managing that identity for you may be empowered to act upon incoming conversations and intelligently filter them. This feature can be used to modify the content of a transmission or, even more often, to make decisions about what to do with a conversation when you're not available (store it offline, copy it to a pager, forward it to another account, etc.). The same feature is also useful to manage the conversations between applications. For instance, if you maintain a personal peer and a work-scheduling peer, conversation management software can redirect incoming conversations to the correct agent based on the relationship to the sender stored in the roster. When you have all of your conversations managed by a common identity, they can be managed directly from one single point, enabling you to have more control over your conversations.

Conclusion

For more information about Jabber, or to become involved in the project (we openly welcome anyone interested), visit *http://jabber.org* or contact the core team at *team@jabber.org*. The 1.0 server was released in May of 2000 and rapidly evolved into a 1.2 release in October, due to popularity and demand. The development focus is now on helping the architecture mature and further developing many of the ideas mentioned here. The development team is collaborating to quickly realize the future possibilities described in this paper, so that they're not so "future" after all.

Microsoft Exhibit 1019

chapter seven

Mixmaster Remailers

Adam Langley, Freenet

Remailers are one of the older peer-to-peer technologies, but they have stood the test of time. Work done on them has helped or motivated much of the current work in the P2P field. Furthermore, they can be valuable to users who want to access many of the systems described in other chapters of this book by providing a reasonable degree of anonymity during this access, as explained in Chapter 15, *Trust*.

Anonymous remailers allow people to send mail or post to newsgroups while hiding their identities. There are many reasons why people might want to act anonymously. Maybe they fear for their safety if they are linked to what they post (a concern of the authors of the Federalist Papers), maybe they think people will prejudge what they have to say, or maybe they just prefer to keep their public lives separate from their private lives. Whatever the reason, anonymous posting is quite difficult on the Internet. Every email has, in its headers, a list of every computer it passed through. Armed with that knowledge, an attacker could backtrack an email to you. If, however, you use a good remailer network, you make that task orders of magnitude harder.

Mixmasters (also known as Type 2 remailers) are the most common type of remailer. The Type 1 remailers are technically inferior and no longer used, though Mixmasters provide backward compatibility with them. The first stable, public release of Mixmaster was on May 3, 1995, by Lance Cottrell. The current version is 2.0.3, released on July 4, 1996. Don't be put off by the old release date; Mixmasters are still the best remailers.

Microsoft Exhibit 1019

A simple example of remailers

In order to demonstrate the basics of remailers, I'll start with the Type 1 system. The Type 2 system builds on it, adding some extra assurances that messages cannot be traced.

If you wanted to mail something anonymously to *alice@world.net*, you could send the following message to a Mixmaster remailer:

```
::
Anon-To: alice@world.net
Latent-Time: +1:30

I have some important information for you. I hope you understand why
I've taken the precautions I have to keep my identity a secret.
```

The remailer would hold this message for one and a half hours—to throw off track anyone who might be sniffing traffic and trying to match your incoming message to the remailer's outgoing message—and then strip all the headers except the subject and forward the mail to Alice. Alice would see that the mail had come from the remailer and would have no idea who actually sent it.

However, this system does have problems. First, the remailer knows the destination and source of the message and could be compromised. Second, while your message is in transit to the remailer, anyone with privileged access to your local area network or an intervening mail hub can see that you are sending anonymous messages to Alice. Finally, Alice has no easy way to reply to you.

In order to hide the fact that you are sending anonymous messages to Alice, you can encrypt the message to the remailer. This assumes that you know the public key of the remailer, and while these public keys are widely known, key management is always a weak spot.

Encryption stops anyone who views the message in transit to the remailer from seeing the message and destination. (It should be noted that this doesn't hide the fact that you are sending anonymous messages, and even that snippet of information could land you in trouble in some places.) To anyone who saw it, the message would look like this:

```
::
Encrypted: PGP

-----BEGIN PGP MESSAGE-----
Version: 5
Comment: The following is encrypted data

mQGiBDmG74kRBACzWRoHjjbTrgGxp7275Caldaol72oWkPgj6xxHl2KNnDyvSyNi
D+PDQUk0W86EXTr9fR8mi8V8yDzSuUQCthoD8UPf7Kk/HtR//lCGWRhoN81ynrsm
FLVhGSR5n4lgf6oNUeIObKYYOWmXzjtKCkgAUtbsImOd8/5hm7zKCQl/LwCgveIW
```


Microsoft Exhibit 1019

```
3bcbQ+A02SM1rxUZcx4qCfUD/1RRuZsdsJFsX9N/tBDLclqtepGQbtwJG02QSCMa
ut81s+WEytb+1/jqBP/qN9Rry3YUtuRXmjjiYFQ813JWA5kd4VxzKP6nBTZfggEW
6BrGB8wDuhqTVL7SqivqrDdgB7S3WQIuZz17Vs1A1wzc37vDmHkw50wshTuvT0Pw
-----END PGP MESSAGE-----
```

This also solves the third problem of Alice needing to reply. You can give Alice a block, encrypted to the remailer, which contains your email address. If Alice then puts the encrypted block at the top of her reply and sends it to the same remailer, the remailer can decrypt it and forward it back to you. Alice can send messages to you without any way of knowing where they actually go. Thus, she has no way of tracing you.

That leaves the second problem, namely that the remailer is the weak link. If Alice, or anyone else, can compromise it, the whole project falls apart. The solution is a simple extension of the basic idea. Instead of the remailer sending the message to Alice, it sends it to another remailer. That remailer then sends it to another, and so on, until the last remailer in the chain sends it to Alice. Thus, no remailer in the chain knows both the source and the destination of the message.

Onion routing

If any remailer reads the contents of your message, it will know who is receiving it at the end. The solution to this involves a series of encryptions that hide the information from remailers in the middle.

Thus, when you send your message, you add an instruction to send it to *alice@world.net*, but you encrypt this recipient information using a key from the last remailer in the chain. So only this last remailer can determine her address. You then add instructions to send the mail to the last remailer and encrypt *that* information so that only the second-to-last remailer can read it, and so on. You thus form an "onion" of messages. Each remailer can remove a skin (one layer of encryption) and send the message to the next remailer, and no remailer knows anything more than what is under the skin they can remove. The layers are illustrated in Figure 7-1.

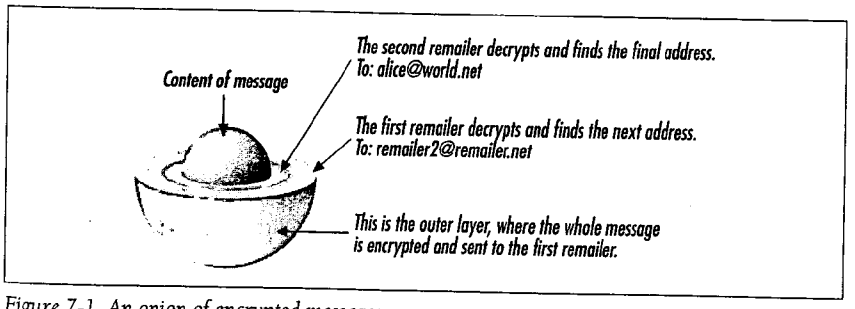

Figure 7-1. An onion of encrypted messages

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You construct a reply block for Alice in the same fashion, an onion of encrypted messages. Alice, or anyone else, would then need to compromise every remailer in the chain in order to remove every skin of the onion and trace you.

How Type 2 remailers differ from Type 1 remailers

Type 2 remailers were designed to fix some of the problems with the Type 1 system above. Even though the Type 1 system seems very good, there are a number of weaknesses that a powerful attacker could use. Most of these weaknesses come from being able to do traffic analysis.

Traffic analysis means capturing the bits that cross a communications channel so as to see every packet that passes around a network—where it came from and where it's going. It is not necessary for the snooper to be able to read the contents of every packet; a lot of useful information can be gathered just from TCP and IP headers sent in the clear, or, as you will see, just from incidental characteristics such as the length of a message.

In order to hide the connection between your incoming message and the Mixmaster's outgoing message, each message must appear to the attacker exactly the same as every other message in the system. The most basic difference between messages is their length. (Remember that the message is multiply encrypted, so the contents don't count.) If an attacker can see a certain sized message going into a remailer and then see a message of a very similar size going out again, he or she can follow the message. Even though the message changes size at each remailer because a skin is peeled off, this doesn't provide much protection. The change in size as the skins are removed is small and easily calculated.

In order to make all messages the same size and frustrate traffic analysis, every Mixmaster message is the same length. This is done by breaking the message into pieces and adding padding to the last part to make it the same size. Each part is sent separately and has enough information for the last remailer in the chain to reassemble them. Only the last remailer in the chain knows what messages go together, because the information is only on the last skin. To every other remailer, each part looks like a different message.

The next identifying mark that needs to be removed is the time. If a message enters a remailer and another leaves immediately after, an attacker knows where the message is going and can trace it. This is a more difficult problem to solve than it seems at first. Simply reordering messages, or delaying them for a time,

Microsoft Exhibit 1019

doesn't work. If the number of other messages is low, or if the attacker can stop other messages from reaching the remailer, your message will still stand out.

Mixmasters try to solve this problem by sending out a random selection of messages periodically, while always keeping a certain sized pool of messages. This makes it very difficult to match up outgoing messages with incoming ones, but still not impossible. However, if the traffic on the Mixmaster network is high enough, tracing the message over the whole chain of remailers becomes a massive challenge for an attacker.

Finally, an attacker can capture your message and attempt to replay it through a remailer. Since your message has the encrypted address of the next remailer, by sending many copies of it an attacker can watch for an unusually large number of outgoing messages to a certain address. That address is likely to be the next remailer in the chain (or the final destination). The attacker can then repeat this for each remailer in the chain.

To stop this, every skin has a random ID number. A remailer will not forward a message with the same ID number twice, so all the cloned messages will be dropped and no extra traffic will come out. An attacker cannot change the ID number of a message because it is encrypted along with everything else.

General discussion

Mixmasters have taken remailing to a fine art and are very good at it. They are an interesting study in peer-to-peer networks in which security is the absolute priority. Unlike many peer-to-peer networks, the Mixmaster user must have knowledge of the network in order to build the onion. This means that Mixmaster nodes are publicly known. It is possible to have a private remailer by simply not telling anyone about it, but this would leave the traffic level very low and thus reduce security.

Unfortunately, Mixmasters themselves are often the target of attacks by people who, for one reason or another, disagree that people have a right to anonymity. It has been known for people to send death threats to themselves to try to get remailers shut down. The public nature of remailers makes such attacks easier.

Life can be very hard for a Mixmaster administrator, because he has to explain to angry people why he can't give them the email address of someone who has used his remailer. This goes some way to explaining why there are only about 20–30 active Mixmasters and serves as a warning to other peer-to-peer projects that provide anonymity.

Microsoft Exhibit 1019

chapter eight

Gnutella

Gene Kan, Gnutella and GoneSilent.com

> When forced to assume [self-government], we were novices in its science. Its principles and forms had entered little into our former education. We established, however, some, although not all its important principles.
>
> —*Thomas Jefferson, 1824*

> Liberty means responsibility. That is why most men dread it.
>
> —*George Bernard Shaw*

Gnutella is among the first of many decentralized technologies that will reshape the Internet and reshape the way we think about network applications. The traditional knee-jerk reaction to create a hierarchical client/server system for any kind of networked application is being rethought. Decentralized technologies harbor many desirable qualities, and Gnutella is a point of proof that such technologies, while young, are viable.

It is possible that Gnutella has walked the Earth before. Certainly many of the concepts it uses—even the unconventional ones—were pioneered long ago. It's tricky to determine what's brand-new and what's not, but this is for certain: Gnutella is the successful combination of many technologies and concepts at the right time.

Gnutella in a gnutshell

Gnutella is a citizen of two different worlds. In the popular consciousness, Gnutella is a peer-to-peer, techno-chic alternative to Napster, the popular Internet music swapping service. To those who look past the Napster association, Gnutella is a landscape-altering technology in and of itself. Gnutella turned every

Microsoft Exhibit 1019

academically correct notion of computer science on its head and became the first large-scale, fully decentralized system running on the wild and untamed public Internet.

Roughly, Gnutella is an Internet potluck party. The virtual world's equivalents of biscuits and cheese are CPU power, network capacity, and disk space. Add a few MP3s and MPEGs, and the potluck becomes a kegger.

On the technical side, Gnutella brings together a strange mix of CDMA, TCP/IP, and lossy message routing over a reliable connection. It's a really strange concept.

Contrary to popular belief, Gnutella is not branded software. It's not like Microsoft Word. In fact, Gnutella is a language of communication, a protocol. Any software that speaks the language is Gnutella-compatible software. There are dozens of flavors of Gnutella compatibles these days, each catering to different users. Some run on Windows, others on Unix, and others are multiplatform Java or Perl. And as Gnutella's name implies, many of the authors of these Gnutella compatibles have contributed to the open source effort by making the source code of their projects freely available.

A brief history

Besides its impact on the future of intellectual property and network software technology, Gnutella has an interesting story, and it's worth spending a little time understanding how something this big happens with nobody writing any checks.

Gnutella's first breath

Gnutella was born sometime in early March 2000. Justin Frankel and Tom Pepper, working under the dot-com pen name of Gnullsoft, are Gnutella's inventors. Their last life-changing product, Winamp, was the beginning of a company called Nullsoft, which was purchased by America Online (AOL) in 1999. Winamp was developed primarily to play digital music files. According to Tom Pepper, Gnutella was developed primarily to share recipes.

Gnutella was developed in just fourteen days by two guys without college degrees. It was released as an experiment. Unfortunately, executives at AOL were not amenable to improving the state of recipe sharing and squashed the nascent Gnutella just hours after its birth. What was supposed to be a GNU General Public License product when it matured to Version 1.0 was never allowed to grow beyond Version 0.56. Certainly if Gnutella were allowed to develop further under the hands of Frankel and Pepper, this chapter would look a lot different.

Microsoft Exhibit 1019

At least Gnutella was born with a name. The neologism comes from ramming GNU and Nutella together at high speed. GNU is short for GNU's Not Unix, the geekish rallying cry of a new generation of software developers who enjoy giving free access to the source code of their products. Nutella is the hazelnut and chocolate spread produced by Italian confectioner Ferrero. It is typically used on dessert crepes and the like. I think it's great, and chocolate is my nemesis.

Anyway, Gnutella was declared an "unauthorized freelance project" and put out to pasture like a car that goes a hundred miles on a gallon of gas. Or maybe like a technology that could eliminate the need for a physical music distribution network. Cast out like a technology that could close the books on a lot of old-world business models? Well, something like that, anyway.

Open source to the rescue

It was then, in Gnutella's darkest hour, that open source developers intervened. Open source developers did for Gnutella what the strange masked nomads did for George Clooney and friends in *Three Kings*. Bryan Mayland, with some divine intervention, reverse engineered Gnutella's communication language (also known as "Gnutella protocol") and posted his findings on Gnutella's hideout on the Web: *gnutella.nerdherd.net*. Ian Hall-Beyer and Nathan Moinvaziri created a sort of virtual water cooler for interested developers to gather around. Besides the protocol documentation, probably the most important bit of information on the Nerdherd web site was the link to Gnutella's Internet Relay Chat (IRC) channel, *#gnutella*. *#gnutella* had a major impact on Gnutella development, particularly when rapid response among developers was required.

What makes Gnutella different?

Gnutella has that simple elegance and minimalism that marks all great things. Like Maxwell's equations, Gnutella has no extraneous fluff. The large amount of Gnutella-compatible software available is testimony to that: Gnutella is small, easy, and accessible to even first-time programmers.

Unlike the Internet that we are all familiar with, with all its at signs, dots, and slashes, Gnutella does not give meaningful and persistent identification to its nodes. In fact, the underlying structure of the Internet on which Gnutella lives is almost entirely hidden from the end user. In newer Gnutella software (Gnotella, Furi, and Toadnode, for example), the underlying Internet is completely hidden from view. It simply isn't necessary to type in a complex address to access infor-

Microsoft Exhibit 1019

mation on the Gnutella system. Just type in a keyword and wait for the list of matching files to trickle in.

Also unlike standard Internet applications such as email, Web, and FTP, which ride on the bare metal of the Internet, Gnutella creates an application-level network in which the infrastructure itself is constantly changing. Sure, the wires stay in the ground and the routers don't move from place to place, but which wires and which routers participate in the Gnutella network changes by the second. The Gnutella network comprises a dynamic *virtual infrastructure* built on a fixed physical infrastructure.

What makes Gnutella different from a scientific perspective is that Gnutella does not rely on any central authority to organize the network or to broker transactions. With Gnutella, you need only connect to one arbitrary host. Any host. In the early days, discovery of an initial host was done by word of mouth. Now it is done automatically by a handful of "host caches." In any case, once you connect with one host, you're in. Your Gnutella node mingles with other Gnutella nodes, and pretty soon you're in the thick of things.

Contrast that to Napster. Napster software is programmed to connect to *www.Napster.com*. At *www.Napster.com* is a farm of large servers that broker your every search and mouse click. This is the traditional client/server model of computing. Don't get me wrong: client/server is great for many things. Among its positive qualities are easy-to-understand scalability and management. The downside is that by being the well-understood mainstay of network application science, client/server is boring, inflexible, and monolithic. Those are bad words in the Internet lexicon.

Gnutella works like the real world

So far, we know that Gnutella is an Internet potluck. We know it's impossible to stop. But how does it actually work all this magic?

In its communication, it's like finding the sushi tray at a cocktail party. The following is a loose description of the interaction on the Gnutella network.

A Gnutella cocktail party

The concepts introduced in this example, primarily the idea that a request is repeated by a host to every other host known by that host, is critical to understanding how Gnutella operates. In any case, you can see that Gnutella's communication concepts closely reflect those of the real world:

Microsoft Exhibit 1019

Cocktail party	Gnutella
You enter at the foyer and say hello to the closest person.	You connect to a Gnutella host and issue a PING message.
Shortly, your friends see you and come to say hello.	Your PING message is broadcast to the Gnutella hosts in your immediate vicinity. When they receive your PING, they respond with a PONG, essentially saying, "Hello, pleased to meet you."
You would like to find the tray of sushi, so you ask your nearby friends.	You would like to find the recipe for strawberry rhubarb pie, so you ask the Gnutella nodes you've encountered.
None of your drunken friends seem to know where the sushi is, but they ask the people standing nearby. Those people in turn ask the people near them, and so on, until the request makes its way around the room.	One of the Gnutella nodes you're connected to has a recipe for strawberry rhubarb pie and lets you know. Just in case others have a better recipe, your request is passed on to other hosts, which repeat the question to all hosts known to them. Eventually the entire network is canvassed.
A handful of partygoers a few meters away have the tray. They pass back the knowledge of its location by word of mouth.	You get several replies, or "hits," routed back to you.
You walk over to the keepers of the tray and partake of their sushi.	There are dozens of recipes to choose from. You double-click on one and a request is issued to download the recipe from the Gnutella node that has it.

A client/server cocktail party

In contrast, centralized systems don't make much sense in the real world. Napster is a good example of a client/server system, so let's look at how things would be if there were a real-life cocktail party that mimicked Napster's system:

Cocktail party	Napster
You enter at the foyer and the host of the party greets you. Around him are clustered thirty-five million of his closest friends.	You connect to Napster and upload a list of files that you are sharing. The file list is indexed and stored in the memory of the party host: the central server.
Your only friend at this party is the host.	The Napster server says, "File list successfully received."
You would like to find the tray of sushi, so you find your way back to the foyer and ask the host where exactly the tray has gone.	You would like to find the recipe for strawberry rhubarb pie. So you type "rhubarb" into the search box, and the request is delivered to the central server.

Microsoft Exhibit 1019

Cocktail party	Napster
The host says, "Oh, yes. It's over there."	You get several replies, or "hits," from the Napster server that match your request.
You hold the tray and choose your favorite sushi.	You decide which MP3 file you want to download and double-click. A request is issued to the Napster server for the file. The Napster server determines which file you desire and whose computer it is on, and brokers a download for you. Soon the download begins.

As you can see, the idea of a central authority brokering all interaction is very foreign to us. When I look at what computer science has espoused for decades in terms of real-world interactions, I wonder how we got so far off track. Computer science has defined a feudal system of servers and slaves, but technologies like Gnutella are turning that around at long last.

Client/server means control, and control means responsibility

As it relates to Napster, the server is at once a place to plant a business model and the mail slot for a summons. If Napster threw the switch for Napster subscriptions, they could force everyone to pay to use their service. And if the RIAA (Recording Industry Association of America) wins its lawsuit, Napster just might have to throw the switch the other way, stranding thirty-five million music swappers. We'll see how that suit goes, but whether or not Napster wins in United States Federal Court, it will still face suits in countless municipalities and overseas. It's the Internet equivalent of tobacco: the lawsuits will follow Napster like so many cartoon rain clouds.

Gnutella, on the other hand, is largely free of these burdens. In a decentralized world, it's tough to point fingers. No one entity is responsible for the operation of the Gnutella network. Any number of warrants, writs, and summons can be executed, and Gnutella will still be around to help you find recipes for strawberry rhubarb pie and "Oops, I Did It Again" MP3s.

Thomas Hale, CEO of WiredPlanet, said, "The only way to stop Gnutella is to turn off the Internet." Well, maybe it's not the only way, but it's really hard to think of a way to eliminate every single cell of Gnutella users, which is truly the only way to wipe Gnutella off the planet.

Microsoft Exhibit 1019

The client is the server is the network

Standard network applications comprise three discrete modules. There is the server, which is where you deposit all the intelligence—the equivalent of the television studio. There is the client, which typically renders the result of some action on the server for viewing by the user—the equivalent of the television. And there is the network, which is the conduit that connects the client and the server—the equivalent of the airwaves.

Gnutella blends all that into one. The client is the server is the network. The client and server are one, of course. That's mainly a function of simplification. There could be two processes, one to serve files and another to download files. But it's just easier to make those two applications one; easier for users and no more difficult for developers.

The interesting thing is that the network itself is embedded in each Gnutella node. Gnutella is an internet built on top of the Internet, entirely in software. The Gnutella network expands as more nodes connect to the network, and, likewise, it does not exist if no users run Gnutella nodes. This is effectively a software-based network infrastructure that comes and goes with its users. Instead of having specialized routers and switches and hubs that enable communication, Gnutella marries all those things into the node itself, ensuring that the communication facilities increase with demand. Gnutella makes the network's users the network's operators.

Distributed intelligence

The underlying notion that sets Gnutella apart from all other systems is that it is a system of distributed intelligence. The queries that are issued on the network are requests for a response, any kind of response.

Suppose you query the Gnutella network for "strawberry rhubarb pie." You expect a few results that let you download a recipe. That's what we expect from today's Gnutella system, but it actually doesn't capture the unique properties Gnutella offers. Remember, Gnutella is a distributed, real-time information retrieval system wherein your query is disseminated across the network in its raw form. That means that every node that receives your query can interpret your query however it wants and respond however it wants, in free form. In fact, Gnutella file-sharing software does just that.

Each flavor of Gnutella software interprets the search queries differently. Some Gnutella software looks inside the files you are sharing. Others look only at the filename. Others look at the names of the parent directories in which the file is

Microsoft Exhibit 1019

contained. Some Gnutella software interprets multiword queries as conjunctions, while others look at multiword queries as disjunctions. Even the results returned by Gnutella file-sharing software are wildly different. Some return the full path of the shared file. Others return only the name of the file. Yet others return a short description extracted from the file. Advertisers and spammers took advantage of this by returning URLs to web sites completely unrelated to the search. Creative and annoying, yet demonstrative of Gnutella's power to aggregate a collective intelligence from distributed sources.

To prove the point once and for all that Gnutella could be used to all kinds of unimagined benefit, Yaroslav Faybishenko, Spencer Kimball, Tracy Scott, and I developed a prototype search engine powered by Gnutella that we called InfraSearch. The idea was that we could demonstrate Gnutella's broad power by building a search engine that accessed data in a nontraditional way while using nothing but pure Gnutella protocol. At the time, InfraSearch was conceived solely to give meat to what many Gnutella insiders were unable to successfully convey to journalists interested in Gnutella: that Gnutella reached beyond simple file swapping. To illustrate, I'll use the examples we used in our prototype.

InfraSearch was accessed through the World Wide Web using a standard web browser. Its interface was familiar to anyone who had used a traditional web search engine. What happened with the query was all Gnutella. When you typed a search query into InfraSearch, however, the query was not answered by looking in a database of keywords and HTML files. Instead, the query was broadcast on a private Gnutella network comprising a few nodes. The nodes themselves were a hodgepodge of variegated data sources. A short list of the notables: Online Photo Lab's image database, a calculator, a proxy for Yahoo! Finance, and an archive of MoreOver.com's news headlines.

When you typed in "MSFT" the query would be broadcast to all the nodes. Each node would evaluate the query in relation to its knowledge base and respond only if the node had relevant information to share. Typically, that would mean that the Yahoo! Finance node would return a result stating Microsoft's current stock price and the MoreOver.com node would return a list of news stories mentioning Microsoft. The results were just arbitrary snippets of HTML. The HTML fragments would be stitched together by a Gnutella node, which also doubled as a web server, and forwarded on to the web browser. Figure 8-1 shows the results of a search for "rose."

The real power of this paradigm showed itself when one entered an algebraic expression into the search box, say, "1+1*3" for instance. The query would be disseminated and most nodes would realize that they had nothing intelligent to

Microsoft Exhibit 1019

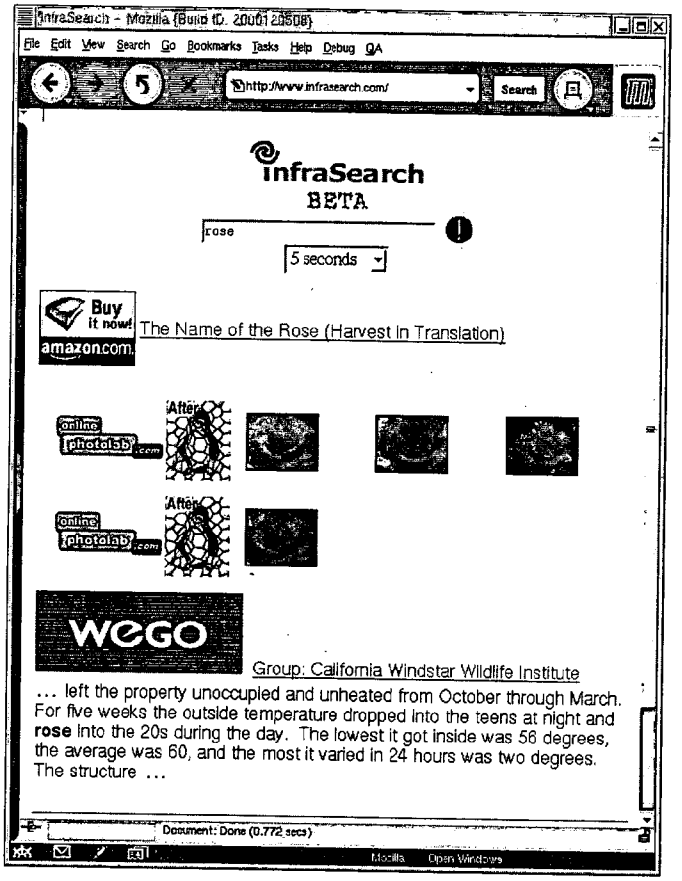

Figure 8-1. Results displayed from Gnutella search

say about such a strange question. All except the calculator node. The calculator was a GNU *bc* calculator hacked to make it speak Gnutella protocol. Every time the calculator received a query, it parsed the text to see if it was a valid algebraic expression. If it was not, then the calculator remained silent. If the query was an algebraic expression, however, the calculator evaluated the expression and returned the result. In this case, "1+1*3 = 4" would be the result.*

* Some creative users would search on ridiculously complex algebraic expressions, causing the calculator node to become overburdened. Gnutella would then simply discard further traffic to the calculator node until it recovered from figuring out what "98791283741984719798797123 4*1234183743748845765" was. The other nodes continued on unaffected.

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One potential application of this is to solve the dynamic page problem on the World Wide Web. Instead of trying to spider those pages as web search crawlers currently do, it would be possible to access the information databases directly and construct a response based upon data available at the time the query was issued. Possibilities that reach even further are within sight. The query could become structured or parameterized, making a huge body of data available through what effectively becomes a unified query interface. The possibilities for something like that in the enterprise are enormous. When peer-to-peer systems take off, accessing data across heterogeneous information stores will become a problem that Gnutella has already demonstrated it can solve.

What we realized is that this aggregation of intelligence maps very closely to the real world. When you ask a question of two different people, you expect two different answers. Asking a question about cars of a mechanic and a toy shop clerk would expectedly yield two very different answers. Yet both are valid, and each reflects a different sort of intelligence in relation to the topic. Traditional search technologies, however, apply only one intelligence to the body of data they search. Distributed search technologies such as Gnutella allow the personality of each information provider and software developer to show through undiluted.

Different from Freenet

Oftentimes Gnutella and Freenet are lumped together as decentralized alternatives to Napster. True, Gnutella and Freenet are decentralized. And it's true that one can share MP3 files using either Gnutella or Freenet. The technical similarities extend further in various ways, but the philosophical division between Gnutella and Freenet picks up right about here.

Freenet can really be described as a bandwidth- and disk space–sharing concept with the goal of promoting free speech. Gnutella is a searching and discovery network that promotes free interpretation and response to queries. With Freenet, one allocates a certain amount of one's hard drive to the task of carrying files which are in the Freenet. One shares bandwidth with others to facilitate the transport of files to their optimal localities in the Freenet. In a sense, Freenet creates a very large and geographically distributed hard drive with anonymous access. The network is optimized for computerized access to those files rather than human interaction. Each file is assigned a complex unique identification that is obscure in its interpretation. The only way to search for files is by searching via that unique identification code.

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In contrast, Gnutella is a distributed searching system with obvious applications for humans and less obvious applications for automatons. Each Gnutella node is free to interpret the query as it wants, allowing Gnutella nodes to give hits in the form of filenames, advertising messages, URLs, graphics, and other arbitrary content. There is no such flexibility in the Freenet system. The Japanese Gnutella project, *http://jnutella.org*, is deploying Gnutella on i-Mode mobile phones, where the results of a search are tailored to mobile phone interfaces. Freenet's highly regimented system of file location based upon unique identification is about cooperative distribution of files. There is nothing wrong with this. It's just a different approach with different effects which I'll leave to Freenet's authors to explain.

Gnutella's communication system

With the basic understanding that Gnutella works the way real-world interpersonal communication works, let's take a look at the concepts that make it all possible in the virtual world. Many of these concepts are borrowed from other technologies, but their combination into one system makes for interesting results and traffic jams.

Message-based, application-level routing

Traditional application-level networks are circuit-based, while Gnutella is message-based. There is no idea of a persistent "connection," or circuit, between any two arbitrary hosts on the Gnutella network. They are both on the network but not directly connected to each other, and not even indirectly connected to each other in any predictable or stable fashion. Instead of forcing the determinism provided by circuit-based routing networks, messages are relayed by a computerized bucket-brigade which forms the Gnutella network. Each bucket is a message, and each brigadier is a host. The messages are handed from host to host willy-nilly, giving the network a unique interconnected and redundant topology.

TCP broadcast

Another unconventional approach that Gnutella uses is a broadcast communication model over unicast TCP. Contrast this to a traditional system such as Napster, where communication is carefully regulated to minimize traffic to its absolute lowest levels, and even then to only one or two concerned parties. Traditional networking models are highly regimented and about as natural as formal gardens.

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The broadcast mechanism is extremely interesting, because it maps very closely to our everyday lives. Suppose you are standing at a bus stop and you ask a fellow when the next bus is to arrive: "Oi, mate! When's the next bus?" He may not know, but someone nearby who has heard you will hopefully chime in with the desired information. That is the strength behind Gnutella: it works like the real world.

One of the first questions I asked upon learning of Gnutella's TCP-based broadcast was, "Why not UDP?" The simple answer is that UDP is a pain. It doesn't play nicely with most firewall configurations and is tricky to code. Broadcasting on TCP is simple, and developers don't ask questions about how to assess "connection" status. Let's not even start on IP multicast.

Message broadcasting

Combining the two concepts of message-based routing and broadcast gives us what I'll term message broadcasting. Message broadcasting is perfect for situations where more than one network participant can provide a valid response to a request. This same sort of thing happens all the time. Auctions, for example, are an example of message broadcasting. The auctioneer asks for bids, and one person's bid is just as good as another's.

Gnutella's broadcasting mechanism elegantly avoids continuous echoing. Messages are assigned unique identifiers (128-bit unique identifiers, or UUIDs, as specified by Leach and Salz's 1997 *UUIDs and GUIDs Informational Draft* to the IETF). With millions of Gnutella nodes running around, it is probably worth answering the question, "How unique is a UUID?" Leach and Salz assert uniqueness until 3400 A.D. using their algorithm. Anyway, it's close enough that even if there were one or two duplicated UUIDs along the way nobody would notice.

Every time a message is delivered or originated, the UUID of the message is memorized by the host it passes through. If there are loops in the network then it is possible that a host could receive the same message twice. Normally, the host would be obligated to rebroadcast the message just like any other that it received. However, if the same message is received again at a later time (it will have the same UUID), it is not retransmitted. This explicitly prevents wasting network resources by sending a query to hosts that have already seen it.

Another interesting idea Gnutella implements is the idea of decay. Each message has with it a TTL* number, or time-to-live. Typically, a query starts life

* TTL is not unique to Gnutella. It is present in IP, where it is used in a similar manner.

Microsoft Exhibit 1019

with a TTL of 7. When it passes from host to host, the TTL is decremented. When the TTL reaches 0, the request has lived long enough and is not retransmitted again. The effect of this is to make a Gnutella request fan out from its originating source like ripples on a pond. Eventually the ripples die out.

Dynamic routing

Message broadcasting is useful for the query, but for the response, it makes more sense to route rather than to broadcast. Gnutella's broadcast mechanism allows a query to reach a large number of potential respondents. Along the way, the UUIDs that identify a message are memorized by the hosts it passes through. When Host A responds to a query, it looks in its memory and determines which host sent the query (Host B). It then responds with a reply message containing the same UUID as the request message. Host B receives the reply and looks in its memory to see which host sent the original request (Host C). And on down the line until we reach Host X, which remembers that it actually originated the query. The buck stops there, and Host X does something intelligent with the reply, like display it on the screen for the user to click on (see Figure 8-2).

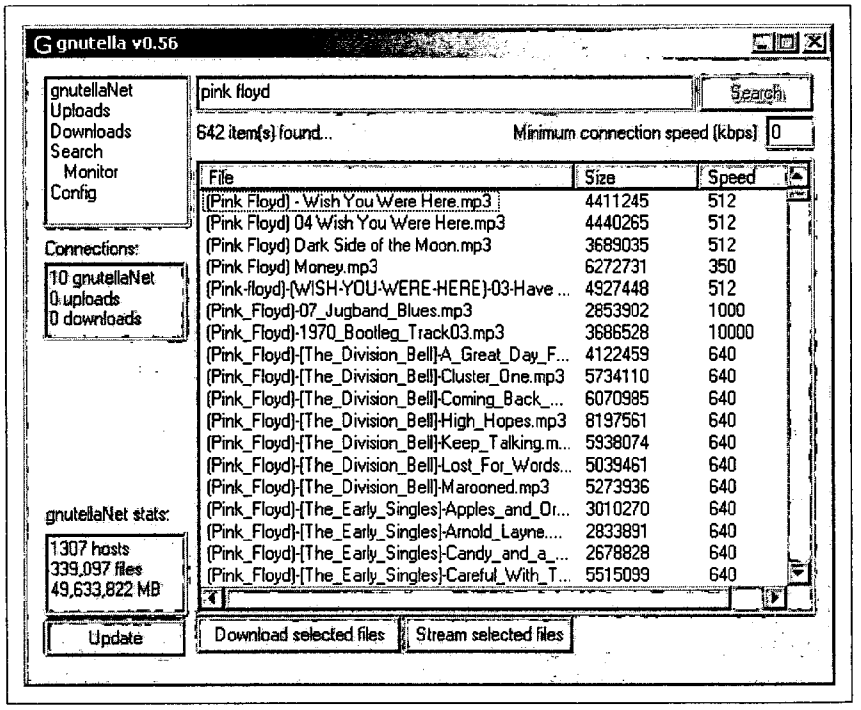

Figure 8-2. Results displayed from a Gnutella query

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The idea to create an ephemeral route as the result of a broadcast for discovery is not necessarily novel, but it is interesting. Remember, a message is identified only by its UUID. It is not associated with its originator's IP address or anything of the sort, so without the UUID-based routes, there is no way for a reply to be delivered to the node that made the request.

This sort of dynamic routing is among the things that make Gnutella the intriguing technology that it is. Without it, there would need to be some kind of fixed Gnutella infrastructure. With dynamic routing, the infrastructure comes along with the nodes that join the network, in real time. A node brings with it some network capacity, which is instantly integrated into the routing fabric of the network at large.

When a node leaves the network, it does not leave the network at large in shambles, as is typical for the Internet. The nodes connected to the departing node simply clean up their memories to forget the departed node, and things continue without so much as a hiccup. Over time, the network adapts its shape to long-lived nodes, but even if the longest-lived, highest-capacity node were to disappear, there would be no lasting adverse effects.

Lossy transmission over reliable TCP

A further unconventional notion that is core to Gnutella's communication mechanisms is that the TCP connections that underlie the Gnutella network are not to be viewed as the totally reliable transports they are typically seen as. With Gnutella, when traffic rises beyond the capacity that a particular connection can cope with, the excess traffic is simply forgotten. It is not carefully buffered and preserved for future transmission as is typically done. Traffic isn't coddled on Gnutella. It's treated as the network baggage that it is.

The notion of using a reliable transport to unreliably deliver data is notable. In this case, it helps to preserve the near-real-time nature of the Gnutella network by preventing an overlong traffic backlog. It also creates an interesting problem wherein low-speed Gnutella nodes are at a significant disadvantage when they connect to high-speed Gnutella nodes. When that happens, it's like drinking from a fire hose, and much of the data is lost before it is delivered.

On the positive side, loss rates provide a simple metric for relative capacity. If the loss rate is consistently high, then it's a clear signal to find a different hose to drink from.

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Organizing Gnutella

One of the ways Gnutella software copes with constantly changing infrastructure is by creating an ad hoc backbone. There is a large disparity in the speeds of Internet connections. Some users have 56-Kbps modems, and others have, say, T3 lines. The goal is that, over time, the T3-connected nodes migrate toward the center of the network and carry the bulk of the traffic, while the 56-Kbps nodes simultaneously move out toward the fringes of the network, where they will not carry as much of the traffic.

In network terms, the placement of a node on the network (in the middle or on the fringes) isn't determined geographically. It's determined in relation to the topology of the connections the node makes. So a high-speed node would end up being connected to potentially hundreds of other high-speed Gnutella nodes, acting as a huge hub, while a low-speed node would hopefully be connected to only a few other low-capacity nodes.

Over time this would lead the Gnutella network to have a high concentration of high-speed nodes in the middle of the network, surrounded by rings of nodes with progressively decreasing capacities.

Placing nodes on the network

When a Gnutella node connects to the network, it just sort of parachutes in blindly. It lands where it lands. How quickly it is able to become a productive member of Gnutella society is determined by the efficacy of its network analysis algorithms. In the same way that at a cocktail party you want to participate in conversations that interest you, that aren't too dull and aren't too deep, a Gnutella node wants to quickly determine which nodes to disconnect from and which nodes to maintain connections to, so that it isn't overwhelmed and isn't too bored.

It is unclear how much of this logic has been implemented in today's popular Gnutella client software (Gnotella, Furi, Toadnode, and Gnutella 0.56), but this is something that Gnutella developers have slowly educated themselves about over time. Early Gnutella software would obstinately maintain connections to nodes in spite of huge disparities in carrying capacity. The effect was that modem nodes acted as black holes into which packets were sent but from which nothing ever emerged.

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One of the key things that we* did to serve the surges of users and new client software was to run high-speed nodes that were very aggressive in disconnecting nodes which were obviously bandwidth disadvantaged. After a short time, the only active connections were to nodes running on acceptably high-speed links. This kind of feedback system created an effective backbone that was captured in numerous early network maps. A portion of one is shown in Figure 8-3.

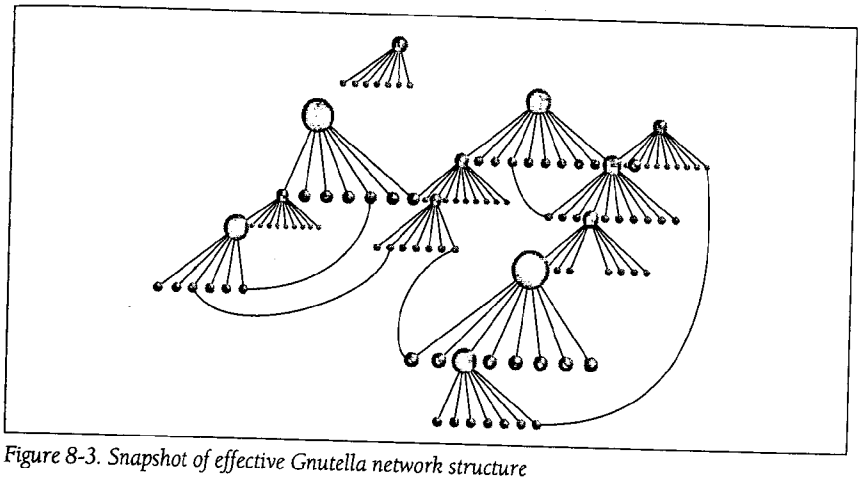

Figure 8-3. Snapshot of effective Gnutella network structure

Gnutella's analogues

The first thing that technologists say when they think about how Gnutella works is, "It can't possibly scale." But that is simply not the case. Gnutella is an unconventional system and as such requires unconventional metrics. Millions of users may be using Gnutella simultaneously, but they will not all be visible to one another. That is the basic nature of a public, purely peer-to-peer network. Because there is no way to guarantee the quality of service throughout the network, it is impossible to guarantee that every node on the network can be reached by every other node on the network. In spite of that, Gnutella has many existing analogues.

Of all the analogues that exist, the most interesting two are cellular telephony and Ethernet.

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* Bob Schmidt, Ian Hall-Beyer, Nathan Moinvaziri, Tom Camarda, and countless others came to the rescue by running software which made the network work in its times of need. This software ranged from standard Gnutella software to host caches to so-called Mr. Clean nodes, which aggressively removed binary detritus from the network.

Microsoft Exhibit 1019

The Gnutella horizon

In Gnutella, there is a concept of a horizon. This is simply a restatement of the effect the TTL has on how far a packet can go before it dies, the attenuation of ripples on a pond. Gnutella's standard horizon is seven hops. That means that from where you stand, you can see out seven hops. How far is that? Typically, a seven-hop radius combined with network conditions means about ten thousand nodes are within sight.

When Gnutella was younger, and the pond analogy hadn't yet crossed my mind, I explained this effect as a horizon, because it was just like what happens when you are at the beach and the world seems to disappear after some distance (approximately five kilometers if you're two meters tall). Of course, that is due to the curvature of the earth, but it seemed like a pretty good analogy.

A slightly better one is what happens in a mob. Think first day of school at UC Berkeley, or the annual Love Parade in Germany. You stand there in the middle of the mob, and you can only see for a short distance around you. It's obvious that there are countless more people outside your immediate vision, but you can't tell how many. You don't even really know where you are in relation to the crowd, but you're certainly in the thick of it. That's Gnutella.

Each node can "see" a certain distance in all directions, and beyond that is a great unknown. Each node is situated slightly differently in the network and as a result sees a slightly different network. Over time, as nodes come and go and the network shifts and morphs, your node gets to see many different nodes as the network undulates around it. If you've used Gnutella, you've seen this happen. Initially, the host count increases very rapidly, but after a minute or two, it stabilizes and increases much more slowly than it did at the outset. That is because in the beginning your node discovers the network immediately surrounding it: the network it can see. Once that is done, your node discovers only the nodes that migrate through its field of view.

Cellular telephony and the Gnutella network

In the technological world, this concept is mirrored exactly by cellular telephony cell sites (cellular telephony towers). Each site has a predetermined effective radius. When a caller is outside that radius, his telephone cannot reach the site and must use another if a nearer one is available. And once the caller is outside the operating radius, the site cannot see the caller's telephone either. The effect is the irksome but familiar "no coverage" message on your phone.

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Cellular network operators situate cell sites carefully to ensure that cell sites overlap one another to prevent no-coverage zones and dropped calls. A real coverage map looks like a Venn diagram gone mad. This is, in fact, a very close analogue of the Gnutella network. Each node is like a cell site in the sense that it has a limited coverage radius, and each node's coverage area overlaps with that of the nodes adjacent to it. The key to making cellular telephony systems scale is having enough cells and enough infrastructure to connect the cells. It's a similar story with Gnutella.

Cell sites are not all that one needs to build a successful cellular network. Behind all those cell towers is a complex high-bandwidth packet switching system, also much like Gnutella. In the cellular world, this network is very carefully thought out and is a piece of physical infrastructure. As with everything else, the infrastructure comes and goes in the Gnutella network, and things are constantly changing shape.

So then the goal is to find a way to create cells that are joined by a high-speed backbone. This is entirely what would happen in the early Gnutella network. Gnutella nodes would gather around a local hub, forming a cluster. There were numerous clusters interconnected by high-speed lines. All this happened in an unplanned and dynamic way.

Ethernet

Gnutella is also similar in function to Ethernet. In fact, Ethernet is a broadcast network where each message has a unique identifier. Like Gnutella, its scalability metrics are unconventional. The question most people ask about Gnutella is, "How many users are on Gnutella?" The answer is complicated.

Millions of users have Gnutella on their computers. One node can only see about ten thousand others from where it stands in the network. So what is the answer? Ten thousand, or several million?

We could ask the same question about Ethernet, and we'd get the same duality in answer. Hundreds of millions of computers have Ethernet, yet only a few dozen can share an Ethernet "segment" before causing network gridlock. The solution for Ethernet was to develop specialized hardware in the form of Ethernet bridges, switches, and routers. With that hardware, it became possible to squeeze all those millions of computers onto the same network: the Internet.

Microsoft Exhibit 1019

Cultivating the Gnutella network

Similar development is underway for Gnutella. Fundamentally, each Gnutella node can contain enough logic to make the Gnutella network grow immensely. Broadening the size of a Gnutella cell, or segment, is only a matter of reducing the network traffic. A minor reduction by each node can translate into a huge reduction in traffic over all nodes. That is what happens with distributed systems: a minor change can have a huge effect, once multiplied over the number of nodes.

There is at least one effort underway to create a specialized Gnutella node which outwardly mimics a standard Gnutella node but inwardly operates in a dramatically different manner. It is known as Reflector and is being developed by a company called Clip2. The Reflector is effectively a miniature Napster server. It maintains an index of the files stored on nodes to which it is connected. When a query is issued, the Reflector does not retransmit it. Rather, it answers the query from its own memory. That causes a huge reduction in network use.*

Anyone can run a Reflector, making it an ideal way to increase the size of a Gnutella cluster. Connecting Reflectors together to create a super high-capacity backbone is the obvious next step. Gnutella is essentially an application-level Internet, and with the development of the Gnutella equivalent of Cisco 12000s, Gnutella will really become what it has been likened to so many times: an internet on the Internet.

Gnutella's traffic problems

One place where the analogy drawn between Gnutella and cellular telephony and Ethernet holds true down to its last bits is how Gnutella suffers in cases of high traffic. We know this because the public Gnutella network at the time of this writing has a traffic problem that is systemic, rather than the standard transient attack. Cellular telephones show a weakness when the cell is too busy with active calls. Sometimes there is crosstalk; at other times calls are scratchy and low quality. Ethernet similarly reaches a point of saturation when there is too much traffic on the network, and, instead of coping gracefully, performance just degrades in a downward spiral. Gnutella is similar in almost every way.

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* Depending on your view, the benefit, or unfortunate downside, of Reflector is that it makes Gnutella usable only in ways that Reflector explicitly enables. To date, Reflector is chiefly optimizing the network for file sharing, and because it removes the ability for hosts to respond free-form and in real time, it sacrifices one of the key ideas behind Gnutella.

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In terms of solutions, the bottom line is that when too many conversations take place in one cell or segment the only way to stop the madness is to break up the cell.

On the Gnutella network, things started out pretty peacefully. First a few hundred users, then a few thousand, then a few hundred thousand. No big deal. The network just soldiered along. The real problem came along when host caches came into wide use.

Host caches

In the early days of Gnutella, the way you found your way onto the network was by word of mouth. You got onto IRC and asked for a host address to connect to. Or you checked one of the handful of web pages which maintained lists of hosts to connect to. You entered the hosts into your Gnutella software one by one until one worked. The the software took care of the rest. It was tedious, but it worked for a long while.

Before host caches, it was fairly random what part of the network you connected to. Ask two different people, and they would direct you to connect to hosts on opposite sides of the Gnutella network. Look at two different host lists, and it was difficult to find any hosts in common. Host lists encouraged sparseness and small clusters. It was difficult for too many new hosts to be concentrated into one cell. The cells were sparsely connected with one another, and there wasn't too much crosstalk. That created a nearly optimal network structure, where the Gnutella network looked like a land dotted by small cities and townships interconnected by only a few roads.

Users eventually became frustrated by the difficulties of getting onto Gnutella. Enter Bob Schmidt and Josh Pieper. Bob Schmidt is the author of GnuCache, a host caching program. Josh Pieper also included host caching logic in his popular Gnut software for Unix. Host caches provide a jumping off spot for Gnutella users, a host that's always up and running, that gives a place for your Gnutella software to connect to and find the rest of the Gnutella network.* The host cache greets your node by handing off a list of other hosts your node should connect to. This removes the uncertainty from connecting to Gnutella and provides a more friendly user experience. We were all very thankful for Schmidt and Pieper's efforts until host caches became a smashing success.

* Actually, Gnutella was born with a ready host cache located at *findshit.gnutella.org*. Unfortunately, the same people who took away Gnutella also took away *findshit.gnutella.org*, leaving us with a host-cacheless world until GnuCache and Pieper's Gnut software came along.

Microsoft Exhibit 1019

An unexpected consequence evidenced itself when waves of new Gnutella users logged on in the wake of the Napster injunction on July 26, 2000. Everyone started relying on host caches as their only means of getting onto the Gnutella network. Host caches were only telling new hosts about hosts they saw recently. By doing that, host caches caused Gnutella nodes to be closely clustered into the same little patch of turf on the Gnutella network. There was effectively only one tightly clustered and highly interconnected cell, because the host caches were doling out the same list of hosts to every new host that connected. What resulted was overcrowding of the Gnutella airwaves and a downward spiral of traffic.

Oh well. That's life in the rough-and-tumble world of technology innovation.

To draw an analogy, the Gnutella network became like a crowded room with lots of conversations. Sure, you can still have a conversation, but maybe only with one or two of your closest friends. And that is what has become frustrating for Gnutella users. Whereas the network used to have a huge breadth and countless well-performing cells of approximately ten thousand nodes each, the current network has one big cell in which there is so much noise that queries only make it one or two hops before drowning in overcrowded network connections.

Effectively, a crowded network means that cells are only a few dozen hosts in size. That makes the network a bear to use and gives a disappointing user experience.

Returning the network to its natural state

Host caches were essentially an unnatural addition to the Gnutella network, and the *law of unintended consequences* showed that it could apply to high technology, too. Improving the situation requires a restoration of the network to its original state, where it grew organically and, at first glance, inefficiently. Sometimes, minor inefficiency is good, and this is one of those cases.

Host lists, by enforcing a sparse network, made it so that the communities of Gnutella nodes that did exist were not overcrowded. Host caches created a tightly clustered network, which, while appearing more efficient, in fact led to a major degradation in overall performance. For host caches to improve the situation, they need only to encourage the sparseness that we know works well.

Sort of. An added complication is that each Gnutella host maintains a local host *catcher*, in which a long list of known hosts (all hosts encountered in the node's travels) is deposited for future reference. The first time one logs into the Gnutella network, a host list or a host cache must be used. For all future logins,

Microsoft Exhibit 1019

Gnutella softwares refer to their host catcher to connect into the Gnutella network. This creates a permanent instability in the network as nodes log on and connect to hosts they remember, irrespective of the fact that those hosts are often poor choices in terms of capacity and topology. The problem is compounded by the reluctance of most Gnutella software to "forget" hosts that are unsuitable.

Turning the network around is technically easy. Host caches can listen to the levels of traffic on each cell they want to serve and distribute new hosts among those cells until the traffic levels become high enough to warrant establishment of new cells. By purposely separating nodes into distinct cells, traffic in each cell can be reduced to a manageable level. With those well-planted seeds and periodic resets of the collective memories of host catchers to allow smart host caches to work their magic, the network can be optimized for performance. The trouble with host catchers, though, is that they are seldom reset, because that requires manual intervention as well as some understanding of the reset mechanism.

Private Gnutella networks

One feature that some Gnutella client software implements is the notion of private Gnutella networks. To join a private network, one needs to know the secret handshake or password. That enforces manageably-sized networks with a predetermined community membership, and it is a pretty good way to ensure a high quality of service no matter what is happening out in the wilds of the public Gnutella network.

Reducing broadcasts makes a significant impact

Broadcasts are simultaneously the most powerful and the most dangerous feature of the Gnutella protocol. Optimally, there are two broadcast packet types: PING and QUERY. PING packets are issued when a node greets the network and wants to learn what other nodes are available to connect to. QUERY packets are issued when a search is conducted. Some Gnutella developers also implemented the PUSH REQUEST packet as a broadcast packet type. PUSH REQUEST packets are used to request files from hosts which are protected by firewalls. The concept is one of the unsung heroes of Gnutella, making it work in all but the most adverse of Internet environments (the double firewall, arch enemy of productivity).

Unfortunately, the PUSH REQUEST packet should be implemented as a routed packet rather than a broadcast packet. At times, PUSH REQUEST packets com-

Microsoft Exhibit 1019

prised 50% of all Gnutella network traffic. Simply routing those packets rather than broadcasting them would reduce the overall network traffic dramatically.

Reductions can also be made in the number of queries that are broadcast to large expanses of the network by intelligently caching results from similar searches. Clip2's Reflector software is an example of such a product. Portions of Reflector can be integrated into each Gnutella client, leading to a small increase in the software's internal complexity (the user need not concern herself with this behind-the-scenes activity) in exchange for a massive improvement in network performance.

The final broadcast packet that was the carrier of some early abuses is the PING packet. In early Gnutella software, PING packets could have a payload, even though it was not clear what that payload might contain. It was subsequently abused by script kiddies to debilitate the Gnutella network. Gnutella developers responded immediately by altering their software to discard PING packets with payloads, causing a several thousand–fold traffic reduction on the Gnutella network and simultaneously foiling what amounted to a denial of service attack.

What developers have been debating ever since is how to reduce the level of traffic usurped by PING packets. Suggestions have ranged from eliminating PING packets to reducing the allowed number of retransmissions of a PING packet. Personally, I favor something in the middle, where every host on the network behaves as a miniature host cache for its locality, returning proxied greetings for a few nearby hosts in response to a PING and only occasionally retransmitting the PING. That would allow the PING to continue to serve its valuable duty in shaping the network's structure and connectivity, while reducing the network's traffic levels dramatically. Figure a thousand-fold reduction in traffic.

One thing to consider in distributed applications is that, no matter how difficult the code is to write and how much it bloats the code (within reason), it's worth the trouble, because the savings in network utilization pay dividends on every packet. As an example, just consider that if a PUSH REQUEST packet is broadcast, it may reach 1,000 hosts. If it is routed, it may reach four or five. That is a 200-fold reduction in traffic in exchange for a dozen lines of code.

The analogy that Gnutella is like an Internet potluck rings true. Everyone brings a dish when they join the Gnutella network. At a minimum, the one dish that everyone brings is network capacity. So then there is definitely enough bandwidth to go around. The only matter is to organize the combined capacity and manage the traffic to make sure the network operates within its limits.

Microsoft Exhibit 1019

We just covered what is probably Gnutella's biggest problem, so if this was a corporate memorandum, this would be the perfect point at which to introduce the engineering organization. This isn't a memo, but let's do it anyway. Keep in mind the Thomas Jefferson quotation at the beginning of the chapter: none of us knew what we were doing, but we got our hands dirty and took responsibility for what we did.

The policy debates

Napster and Gnutella have really been at the center of the policy debate surrounding the new breed of peer-to-peer technologies. For the moment, let's forget about the debate that's burning in the technology community about what is truly peer-to-peer. We'll get back to that later and tie all these policy questions back to the technology.

Napster wars

There is only one thing that gets people more riled up than religion, and that is money. In this case, the squabble is over money that may or may not be lost to online music swaps facilitated by services such as Napster and systems like the Gnutella network. This war is being fought by Napster and the RIAA, and what results could change the lives of everyone, at least in the United States.

Well, sort of. The idea that lawsuit or legislature can stop a service that everyone enjoys is certainly a false one. Prohibition was the last real effort (in the U.S.) by the few against the many, and it was a dismal failure that gave rise to real criminal activity and the law's eventual embarrassing repeal. We have an opportunity to see all that happen again, or the recording industry could look at what's coming down the road and figure out a way to cooperate with Napster before the industry gets run down by next-generation peer-to-peer technologies.

Napster, at least, provides a single place where file swappers can be taxed. With Gnutella and Freenet, there is no place to tax, no person to talk to about instituting a tax, and no kinds of controls. The recording industry may hope that Gnutella and Freenet will "just go away," but that hope will probably not materialize into reality.

Anonymity and peer-to-peer

One of the big ideas behind peer-to-peer systems is their potential to provide a cloak under which users can conduct information exchanges without revealing their identities or even the information they are exchanging. The possibility of

Microsoft Exhibit 1019

anonymity in nearly every case stems from the distribution of information across the entire network, as well as the difficulty in tracking activities on the network as a whole.

Gnutella provides some degree of anonymity by enabling an essentially anonymous searching mechanism. It stops there, though. Gnutella reveals the IP address of a downloading host to the uploading host, and vice versa.

Gnutella pseudoanonymity

Gnutella is a prime example of peer-to-peer technology. It was, after all, the first successful, fully decentralized, peer-to-peer system. But in the policy debate, that's not a huge matter. What does matter is that Gnutella's message-based routing system affords its users a degree of anonymity by making request and response packets part of a crowd of similar packets issued by other participants in the network.

In most messages that are passed from node to node, there is no mention of anything that might tie a particular message to a particular user. On the Internet, identity is established using two points of data: An IP address and the time at which the packet containing the IP address was seen. Most Gnutella messages do not contain an IP address, so most messages are not useful in identifying Gnutella users. Also, Gnutella's routing system is not outwardly accessible. The routing tables are dynamic and stored in the memory of the countless Gnutella nodes for only a short time. It is therefore nearly impossible to learn which host originated a packet and which host is destined to receive it.

Furthermore, Gnutella's distributed nature means that there is no one place where an enforcement agency can plant a network monitor to spy on the system's communications. Gnutella is spread throughout the Internet, and the only way to monitor what is happening on the Gnutella network is to monitor what is happening on the entire Internet. Many are suspicious that such monitoring is possible, or even being done already. But given the vastness of today's Internet and its growing traffic, it's pretty unlikely.

What Gnutella does subject itself to, however, are things such as Zeropaid.com's Wall of Shame. The Wall of Shame, a Gnutella Trojan Horse, was an early attempt to nab alleged child pornography traffickers on the Gnutella network. This is how it worked: a few files with very suggestive filenames were shared by a special host. When someone attempted to download any of the files, the host would log the IP address of the downloader to a web page on the Wall of Shame. The host obtained the IP address of the downloader from its connection information.

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That's where Gnutella's pseudoanonymity system breaks down. When you attempt to download, or when a host returns a result, identifying information is given out. Any host can be a decoy, logging that information. There are systems that are more interested in the anonymity aspects of peer-to-peer networking, and take steps such as proxied downloads to better protect the identities of the two endpoints. Those systems should be used if anonymity is a real concern.

The Wall of Shame met a rapid demise in a rather curious and very Internet way. Once news of its existence circulated on IRC, Gnutella users with disruptive senses of humor flooded the network with suggestive searches in their attempts to get their IP addresses on the Wall of Shame.

Downloads, now in the privacy of your own direct connection

So Gnutella's message-based routing system and its decentralization both give some anonymity to its users and make it difficult to track what exactly is happening. But what really confounds any attempt to learn who is actually sharing files is that downloads are a private transaction between only two hosts: the uploader and the downloader.

Instead of brokering a download through a central authority, Gnutella has sufficient information to reach out to the host that is sharing the desired file and grab it directly. With Napster, it's possible not only to learn what files are available on the host machines but what transactions are actually completed. All that can be done easily, within the warm confines of Napster's machine room.

With Gnutella, every router and cable on the Internet would need to be tapped to learn about transactions between Gnutella hosts or peers. When you double-click on a file, your Gnutella software establishes an HTTP connection directly to the host that holds the desired file. There is no brokering, even through the Gnutella network. In fact, the download itself has nothing to do with Gnutella: it's HTTP.

By being truly peer-to-peer, Gnutella gives no place to put the microscope. Gnutella doesn't have a mailing address, and, in fact, there isn't even anyone to whom to address the summons. But because of the breakdown in anonymity when a download is transacted, Gnutella could not be used as a system for publishing information anonymously. Not in its current form, anyway. So the argument that Gnutella provides anonymity from search through response through download is impossible to make.

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Anonymous Gnutella chat

But then, Gnutella is not exclusively a file-sharing system. When there were fewer users on Gnutella, it was possible to use Gnutella's search monitor to chat with other Gnutella users. Since everyone could see the text of every search that was being issued on the network, users would type in searches that weren't searches at all: they were messages to other Gnutella users (see Figure 8-4).

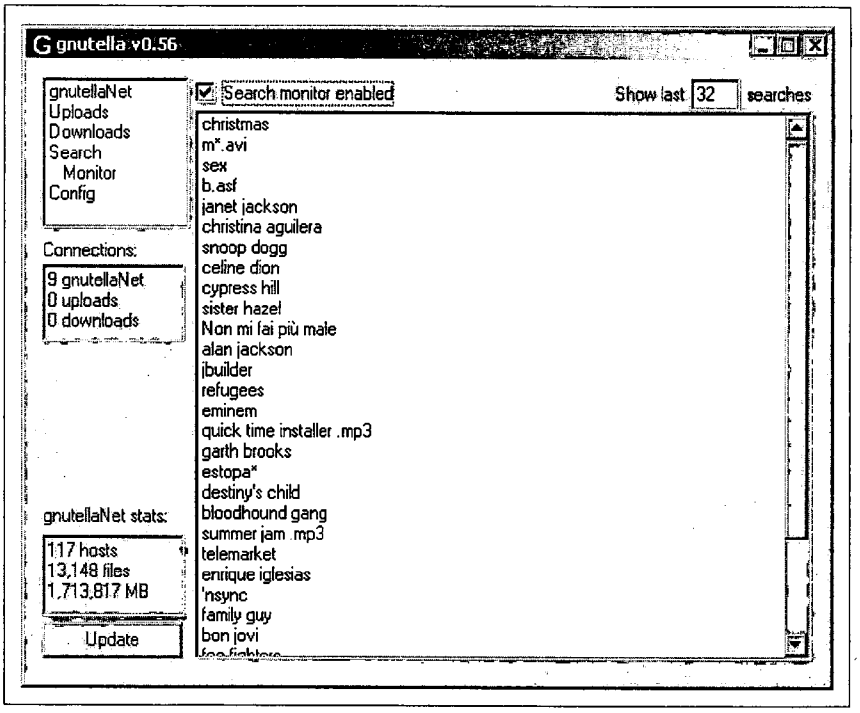

Figure 8-4. Gnutella search monitor

It was impossible to tell who was saying what, but conversations were taking place. If you weren't a part of the particular thread of discussion, the messages going by were meaningless to you. This is an excellent real-world example of the ideas behind Rivest's "Chaffing and Winnowing."* Just another message in a sea of messages. Keeping in mind that Gnutella gives total anonymity in searching, this search-based chat was in effect a totally anonymous chat! And we all thought we were just using Gnutella for small talk.

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* Ronald L Rivest (1998), "Chaffing and Winnowing: Confidentiality without Encryption," *http://www.toc.lcs.mit.edu/~rivest/chaffing.txt*.

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Next-generation peer-to-peer file-sharing technologies

No discussion about Gnutella, Napster, and Freenet is complete without at least a brief mention of the arms race and war of words between technologists and holders of intellectual property. What the recording industry is doing is sensitizing software developers and technologists to the legal ramifications of their inventions. Napster looked like a pretty good idea a year ago, but today Gnutella and Freenet look like much better ideas, technologically and politically. For anyone who isn't motivated by a business model, true peer-to-peer file-sharing technologies are the way to go.

It's easy to see where to put the toll booths in the Napster service, but taxing Gnutella is trickier. Not impossible, just trickier. Whatever tax system is successfully impösed on Gnutella, if any, will be voluntary and organic—in harmony with Gnutella, basically. The same will be true for next-generation peer-to-peer file-sharing systems, because they will surely be decentralized.

Predicting the future is impossible, but there are a few things that are set in concrete. If there is a successor to Gnutella, it will certainly learn from the lessons taught to Napster. It will learn from the problems that Gnutella has overcome and those that frustrate it today. For example, instead of the pseudoanonymity that Gnutella provides, next generation technologies may provide true anonymity through proxying and encryption. In the end, we can say with certainty that technology will outrun policy. It always has. The question is what impact that will have.

Gnutella's effects

Gnutella started the decentralized peer-to-peer revolution.* Before it, systems were centralized and boring. Innovation in software came mainly in the form of a novel business plan. But now, people are seriously thinking about how to turn the Internet upside down and see what benefits fall out.

Already, the effects of the peer-to-peer revolution are being felt. Peer-to-peer has captured the imagination of technologists, corporate strategists, and venture

* The earliest example of a peer-to-peer application that I can come up with is Zephyr chat, which resulted from MIT's Athena project in the early 1990s. Zephyr was succeeded by systems such as ICQ, which provided a commercialized, graphical, Windows-based instant messaging system along the lines of Zephyr. Next was Napster. And that is the last notable client/server-based, peer-to-peer system. Gnutella and Freenet were next, and they led the way in decentralized peer-to-peer systems.

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capitalists alike. Peer-to-peer is even getting its own book. This isn't just a passing fad.

Certain aspects of peer-to-peer are mundane. Certain other aspects of it are so interesting as to get notables including George Colony, Andy Grove, and Marc Andreessen excited. That doesn't happen often. The power of peer-to-peer and its real innovation lies not just in its file-sharing applications and how well those applications can fly in the face of copyright holders while flying under the radar of legal responsibility. Its power also comes from its ability to do what makes plain sense and what has been overlooked for so long.

The basic premise underlying all peer-to-peer technologies is that individuals have something valuable to share. The gems may be computing power, network capacity, or information tucked away in files, databases, or other information repositories, but they are gems all the same. Successful peer-to-peer applications unlock those gems and share them with others in a way that makes sense in relation to the particular applications.

Tomorrow's Internet will look quite different than it does today. The World Wide Web is but a little blip on the timeline of technology development. It's only been a reality for the last six years! Think of the Web as the Internet equivalent of the telegraph: it's very useful and has taught us a lot, but it's pretty crude. Peer-to-peer technologies and the experience gained from Gnutella, Freenet, Napster, and instant messaging will reshape the Internet dramatically.

Unlike what many are saying today, I will posit the following: today's peer-to-peer applications are quite crude, but tomorrow's applications will not be strictly peer-to-peer or strictly client/server, or strictly anything for that matter. Today's peer-to-peer applications are necessarily overtly peer-to-peer (often to the users' chagrin) because they must provide application and infrastructure simultaneously due to the lack of preexisting peer-to-peer infrastructure. Such infrastructure will be put into place sooner than we think. Tomorrow's applications will take this infrastructure for granted and leverage it to provide more powerful software and a better user experience in much the same way modern Internet infrastructure has.

In the short term, decentralized peer-to-peer may spell the end of censorship and copyright. Looking out, peer-to-peer will enable crucial applications that are so useful and pervasive that we will take them for granted.

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chapter nine

Freenet

Adam Langley, Freenet

Freenet is a decentralized system for distributing files that demonstrates a particularly strong form of peer-to-peer. It combines many of the benefits associated with other peer-to-peer models, including robustness, scalability, efficiency, and privacy.

In the case of Freenet, decentralization is pivotal to its goals, which are the following:

- Prevent censorship of documents
- Provide anonymity for users
- Remove any single point of failure or control
- Efficiently store and distribute documents
- Provide plausible deniability for node operators

Freenet grew out of work done by Ian Clarke when he was at the University of Edinburgh, Scotland, but it is now maintained by volunteers on several continents.

Some of the goals of Freenet are very difficult to bring together in one system. For example, efficient distribution of files has generally been done by a centralized system, and doing it with a decentralized system is hard.

However, decentralized networks have many advantages over centralized ones. The Web as it is today has many problems that can be traced to its client/server model. The Slashdot effect, whereby popular data becomes less accessible because of the load of the requests on a central server, is an obvious example.

Centralized client/server systems are also vulnerable to censorship and technical failure because they rely on a small number of very large servers.

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Finally, privacy is a casualty of the structure of today's Web. Servers can tell who is accessing or posting a document because of the direct link to the reader/poster. By cross-linking the records of many servers, a large amount of information can be gathered about a user. For example, DoubleClick, Inc., is already doing this. By using direct marketing databases and information obtained through sites that display their advertisements, DoubleClick can gather very detailed and extensive information. In the United States there are essentially no laws protecting privacy online or requiring companies to handle information about people responsibly. Therefore, these companies are more or less free to do what they wish with the data.

We hope Freenet will solve some of these problems.

Freenet consists of nodes that pass messages to each other. A node is simply a computer that is running the Freenet software, and all nodes are treated as equals by the network. This removes any single point of failure or control. By following the Freenet protocol, many such nodes spontaneously organize themselves into an efficient network.

Requests

In order to make use of Freenet's distributed resources, a user must initiate a request. Requests are messages that can be forwarded through many different nodes. Initially the user forwards the request to a node that he or she knows about and trusts (usually one running on his or her own computer). If a node doesn't have the document that the requestor is looking for, it forwards the request to another node that, according to its information, is more likely to have the document. The messages form a chain as each node forwards the request to the next node. Messages time out after passing through a certain number of nodes, so that huge chains don't form. (The mechanism for dropping requests, called the hops-to-live count, is a simple system similar to that used for Internet routing.) The chain ends when the message times out or when a node replies with the data.

The reply is passed back though each node that forwarded the request, back to the original node that started the chain. Each node in the chain may cache the reply locally, so that it can reply immediately to any further requests for that particular document. This means that commonly requested documents are cached on more nodes, and thus there is no Slashdot effect whereby one node becomes overloaded.

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The reply contains an address of one of the nodes that it came through, so that nodes can learn about other nodes over time. This means that Freenet becomes increasingly connected. Thus, you may end up getting data from a node you didn't even know about. In fact, you still might not know that that node exists after you get the answer to the request—each node knows only the ones it communicates with directly and possibly one other node in the chain.

Because no node can tell where a request came from beyond the node that forwarded the request to it, it is very difficult to find the person who started the request. This provides anonymity to the users who use Freenet.

Freenet doesn't provide perfect anonymity (like the Mixmaster network discussed in Chapter 7, *Mixmaster Remailers*) because it balances paranoia against efficiency and usability. If someone wants to find out exactly what you are doing, then given the resources, they will. Freenet does, however, seek to stop mass, indiscriminate surveillance of people.

A powerful attacker that can perform traffic analysis of the whole network could see who started a request, and if they controlled a significant number of nodes so that they could be confident that the request would pass through one of their nodes, they could also see what was being requested. However, the resources needed to do that would be incredible, and such an attacker could find better ways to snoop on users.

An attacker who simply controlled a few nodes, even large ones, couldn't find who was requesting documents and couldn't generate false documents (see "Key Types," later in this chapter). They couldn't gather information about people and they couldn't censor documents. It is these attackers that Freenet seeks to stop.

Detail of requests

Each request is given a unique ID number by the node that initiates it, and this serves to identify all messages generated by that request. If a node receives a message with the same unique ID as one it has already processed, it won't process it again. This keeps loops from forming in the network, which would congest the network and reduce overall system performance.

The two main types of requests are the *InsertRequest* and the *DataRequest*. The *DataRequest* simply asks that the data linked with a specified key is returned; these form the bulk of the requests on Freenet. *InsertRequests* act exactly like *DataRequests* except that an *InsertReply*, not a *TimedOut* message, is returned if the request times out.

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This means that if an attacker tries to insert data which already exists on Freenet, the existing data will be returned (because it acts like a *DataRequest*), and the attacker will only succeed in spreading the existing data as nodes cache the reply.

If the data doesn't exist, an *InsertReply* is sent back, and the client can then send a *DataInsert* to actually insert the new document. The insert isn't routed like a normal message but follows the same route as the *InsertRequest* did. Intermediate nodes cache the new data. After a *DataInsert*, future *DataRequests* will return the document.

The data store

The major tasks each node must perform—deciding where to route requests, remembering where to return answers to requests, and choosing how long to store documents—revolve around a stack model. Figure 9-1 shows what a stack could contain.

KEY	DATA	ADDRESS
8e0109xb87wkhkujhs98k	99usbkjhgd7333khjgs763	tcp/5.34.27.4:6473
uushs89763kjhx7w732722	yy6254231gsyw4GGcwhgs	tcp/89.34.36.1:24855
kjhks872228x0982876jjhd	TTRas384hgygduybv1111n	tcp/194.44.62.66:9897
878772kx762776xbv8622		tcp/64.28.67.48:43653
222764kjhx8t63wkbkjs77w		tcp/4.18.49.35:65466
57765xkjhd72729jnbck01kj		tcp/55.18.4.1:3895

Figure 9-1. Stack used by a Freenet node

Each key in the data store is associated with the data itself and an address to the node where the data came from. Below a certain point the node no longer stores the data related to a key, only the address. Thus the most often requested data is kept locally. Documents that are requested more often are moved up in the stack, displacing the less requested ones. The distance that documents are moved is linked to the size, so that bigger documents are at a disadvantage. This gives people an incentive not to waste space on Freenet and so compress documents before inserting.

When a node receives a request for a key (or rather the document that is indexed by that key), it first looks to see if it has the data locally. If it does, the

Microsoft Exhibit 1019

request is answered immediately. If not, the node searches the data store to find the key closest to the requested key (as I'll explain in a moment). The node referenced by the closest key is the one that the request is forwarded to. Thus nodes will forward to the node that has data closest to the requested key.

The exact closeness function used is complex and linked to details of the data store that are beyond this chapter. However, imagine the key being treated as a number, so that the closest key is defined as the one where the absolute difference between two keys is a minimum.

The closeness operation is the cornerstone of Freenet's routing, because it allows nodes to become biased toward a certain part of the keyspace. Through routine node interactions, certain nodes spontaneously emerge as the most often referenced nodes for data close to a certain key. Because those nodes will then frequently receive requests for a certain area of the keyspace, they will cache those documents. And then, because they are caching certain documents, other nodes will add more references to them for those documents, and so on, forming a positive feedback.

A node cannot decide what area of the keyspace it will specialize in because that depends on the references held by other nodes. If a node could decide what area of the keyspace it would be asked for, it could position itself as the preferred source for a certain document and then seek to deny access to it, thus censoring it.

For a more detailed discussion of the routing system, see Chapter 14, *Performance*. The routing of requests is the key to Freenet's scalability and efficiency. It also allows data to "move." If a document from North America is often requested in Europe, it is more likely to soon be on European servers, thus reducing expensive transatlantic traffic. (But neighboring nodes can be anywhere on the Internet. While it makes sense for performance reasons to connect to nodes that are geographically close, that is definitely not required.)

Because each node tries to forward the request closer and closer to the data, the search is many times more powerful than a linear search and much more efficient than a broadcast. It's like looking for a small village in medieval times. You would ask at each village you passed through for directions. Each time you passed through a village you would be sent closer and closer to your destination. This method (akin to Freenet's routing closer to data) is much quicker than the linear method of going to every village in turn until you found the right one. It also means that Freenet scales well as more nodes and data are added. It is also better than the Gnutella-like system of sending thousands of messengers to all the villages in the hope of finding the right one.

Microsoft Exhibit 1019

The stack model also provides the answer to the problem of culling data. Any storage system must remove documents when it is full, or reject all new data. Freenet nodes stop storing the data in a document when the document is pushed too far down the stack. The key and address are kept, however. This means that future requests for the document will be routed to the node that is most likely to have it.

This data-culling method allows Freenet to remove the least requested data, not the least agreeable data. If the most unpopular data was removed, this could be used to censor documents. The Freenet design is very careful not to allow this.

The distinction between unpopular and unwanted is important here. Unpopular data is disliked by a lot of people, and Freenet doesn't try to remove that because that would lead to a tyranny of the majority. Unwanted data is simply data that is not requested. It may be liked, it may not, but nobody is interested in it.

Every culling method has problems, and on balance this method has been selected as the best. We hope that the pressure for disk space won't be so high that documents are culled quickly. Storage capacity is increasing at an exponential rate, so Freenet's capacity should also. If an author wants to keep a document in Freenet, all he or she has to do is request or reinsert it every so often.

It should be noted that the culling is done individually by each node. If a document (say, a paper at a university) is of little interest globally, it can still be in local demand so that local nodes (say, the university's node) will keep it.

Keys

As has already been noted, every document is indexed by a key. But Freenet has more than one type of key—each with certain advantages and disadvantages.

Since individual nodes on Freenet are inherently untrusted, nodes must not be allowed to return false documents. Otherwise, those false documents will be cached and the false data will spread like a cancer. The main job of the key types is to prevent this cancer. Each node in a chain checks that the document is valid before forwarding it back toward the requester. If it finds that the document is invalid, it stops accepting traffic from the bad node and restarts the request.

Every key can be treated as an array of bytes, no matter which type it is. This is important because the closeness function, and thus the routing, treats them as equivalent. These functions are thus independent of key type.

Microsoft Exhibit 1019

Key types

Freenet defines a general Uniform Resource Indicator (URI) in the form:

freenet:keytype@data

where binary data is encoded using a slightly changed Base64 scheme. Each key type has its own interpretation of the *data* part of the URI, which is explained with the key type.

Documents can contain metadata that redirects clients to another key. In this way, keys can be chained to provide the advantages of more than one key type. The rest of this section describes the various types of keys.

Content Hash Keys (CHKs)

A CHK is formed from a hash of the data. A hash function takes any input and produces a fixed-length output, where finding two inputs that give the same output is computationally impossible. For further information on the purpose of hashes, see the section on "Message digest functions" in Chapter 15, *Trust*.

Since a document is returned in response to a request that includes its CHK, a node can check the integrity of the returned document by running the same hash function on it and comparing the resulting hash to the CHK provided. If the hashes match, it is the correct document. CHKs provide a unique and tamperproof key, and so the bulk of the data on Freenet is stored under CHKs. CHKs also reduce the redundancy of data, since the same data will have the same CHK and will collide on insertion. However, CHKs do not allow updating, nor are they memorable.

A CHK URI looks like the following example:

freenet:CHK@
DtqiMnTj8YbhScLp1BQoW9In9C4DAQ,2jmj7l5rSw0yVb-vlWAYkA

Keyword Signed Keys (KSKs)

KSKs appear as text strings to the user (for example, "text/books/1984.html"), and so are easy to remember. A common misunderstanding about Freenet, arising from the directory-like format of KSKs, is that there is a hierarchy. There isn't. It is only by convention that KSKs look like directory structures; they are actually freeform strings.

KSKs are transformed by clients into a binary key type. The transformation process makes it impractical to recover the string from the binary key. KSKs are

Microsoft Exhibit 1019

based on a public key system where, in order to generate a valid KSK document, you need to know the original string. Thus, a node that sees only the binary form of the KSK does not know the string and cannot generate a cancerous reply that the requestor would accept.

KSKs are the weakest of the key types in this respect, as it is possible that a node could try many common human strings (such as "Democratic" and "China" in many different sentences) to find out what string produced a given KSK and then generate false replies.

KSKs can also clash as different people insert different data while trying to use the same string. For example, there are many versions of the Bible. Hopefully the Freenet caching system should cause the most requested version to become dominant. Tweaks to aid this solution are still under discussion.

A KSK URI looks like this:

freenet:KSK@text/books/1984.html

Signature Verification Keys (SVKs)

SVKs are based on the same public key system as KSKs but are purely binary. When an SVK is generated, the client calculates a private key to go with it. The point of SVKs is to provide something that can be updated by the owner of the private key but by no one else.

SVKs also allow people to make a subspace, which is a way of controlling a set of keys. This allows people to establish pseudonyms on Freenet. When people trust the owner of a subspace, documents in that subspace are also trusted while the owner's anonymity remains protected. Systems like Gnutella and Napster that don't have an anonymous trust capability are already finding that attackers flood the network with false documents.

Named SVKs can be inserted "under" another SVK, if one has its private key. This means you can generate an SVK and announce that it is yours (possibly under a pseudonym), and then insert documents under that subspace. People trust that the document was inserted by you, because only you know the private key and so only you can insert in that subspace. Since the documents have names, they are easy to remember (given that the user already has the base SVK, which is binary), and no one can insert a document with the same key before you, as they can with a KSK.

Microsoft Exhibit 1019

An SVK URI looks like this:

freenet:SVK@
XChKB7aBZAMIMK2cBArQRo7v05ECAQ,7SThKCDy~QCuODt8xP=KzHA

or for an SVK with a document name:

freenet:SSK@ U7MyLl0mHrjm6443k1svLUcLWFUQAgE/text/books/1984.html

Keys and redirects

Redirects use the best aspects of each kind of key. For example, if you wanted to insert the text of George Orwell's *1984* into Freenet, you would insert it as a CHK and then insert a KSK like "Orwell/1984" that redirects to that CHK. Recent Freenet clients will do this automatically for you. By doing this you have a unique key for the document that you can use in links (where people don't need to remember the key), and a memorable key that is valuable when people are either guessing the key or can't get the CHK.

All documents in Freenet are encrypted before insertion. The key is either random and distributed by the requestor along with the URI, or based on data that a node cannot know (like the string of a KSK). Either way, a node cannot tell what data is contained in a document. This has two effects. First, node operators cannot stop their nodes from caching or forwarding content that they object to, because they have no way of telling what the content of a document is. For example, a node operator cannot stop his or her node from carrying pro-Nazi propaganda, no matter how anti-Nazi he or she may be. It also means that a node operator cannot be responsible for what is on his or her node.

However, if a certain document became notorious, node operators could purge that document from their data stores and refuse to process requests for that key. If enough operators did this, the document could be effectively removed from Freenet. All it takes to bypass explicit censorship, though, is for an anonymous person to change one byte of the document and reinsert it. Since the document has been changed, it will have a different key. If an SVK is used, they needn't even change it at all because the key is random. So trying to remove documents from Freenet is futile.

Because a node that does not have a requested document will get the document from somewhere else (if it can), an attacker can never find which nodes store a document without spreading it. It is currently possible to send a request with a hops-to-live count of 1 to a node to bypass this protection, because the message goes to only one node and is not forwarded. Successful retrieval can tell the requestor that the document must be on that node.

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Future releases will treat the hops-to-live as a probabilistic system to overcome this. In this system, there will be a certain probability that the hops-to-live count will be decremented, so an attacker can't know whether or not the message was forwarded.

Conclusions

In simulations, Freenet works well. The average number of hops for requests of random keys is about 10 and seems largely independent of network size. The simulated network is also resilient to node failure, as the number of hops remains below 20 even after 10% of nodes have failed. This suggests that Freenet will scale very well. More research on scaling is presented in Chapter 14.

At the time of writing, Freenet is still very much in development, and a number of central issues are yet to be decided. Because of Freenet's design, it is very difficult to know how many nodes are currently participating. But it seems to be working well at the moment.

Searching and updating are the major areas that need work right now. During searches, some method must be found whereby requests are routed closer and closer to the answer in order to maintain the efficiency of the network. But search requests are fuzzy, so the idea of routing by key breaks down here. It seems at this early stage that searching will be based on a different concept. Searching also calls for node-readable metadata in documents, so node operators would know what is on their nodes and could then be required to control it. Any searching system must counter this breach as best it can.

Even at this early stage, however, Freenet is solving many of the problems seen in centralized networks. Popular data, far from being less available as requests increase (the Slashdot effect), becomes more available as nodes cache it. This is, of course, the correct reaction of a network storage system to popular data. Freenet also removes the single point of attack for censors, the single point of technical failure, and the ability for people to gather large amounts of personal information about a reader.

Microsoft Exhibit 1019

chapter ten

Red Rover

Alan Brown, Red Rover

The success of Internet-based distributed computing will certainly cause headaches for censors. Peer-to-peer technology can boast populations in the tens of millions, and the home user now has access to the world's most advanced cryptography. It's wonderful to see those who turned technology against free expression for so long now scrambling to catch up with those setting information free. But it's far too early to celebrate: What makes many of these systems so attractive in countries where the Internet is not heavily regulated is precisely what makes them the wrong tool for much of the world.

Red Rover was invented in recognition of the irony that the very people who would seem to benefit the most from these systems are in fact the least likely to be able to use them. A partial list of the reasons this is so includes the following:

The delivery of the client itself can be blocked

The perfect stealth device does no good if you can't obtain it. Yet, in exactly those countries where user secrecy would be the most valuable, access to the client application is the most guarded. Once the state recognized the potential of the application, it would not hesitate to block web sites and FTP sites from which the application could be downloaded and, based on the application's various compressed and encrypted sizes, filter email that might be carrying it in.

Possession of the client is easily criminalized

If a country is serious enough about curbing outside influence to block web sites, it will have no hesitation about criminalizing possession of any application that could challenge this control. This would fall under the ubiquitous legal category "threat to state security." It's a wonderful advance for technology that some peer-to-peer applications can pass messages even the

Microsoft Exhibit 1019

CIA can't read. But in some countries, being caught with a clever peer-to-peer application may mean you never see your family again. This is no exaggeration: in Burma, the possession of a modem—*even a broken one*—could land you in court.

Information trust requires knowing the origin of the information

Information on most peer-to-peer systems permits the dissemination of poisoned information as easily as it does reliable information. Some systems succeed in controlling disreputable transmissions. On most, though, there's an information free-for-all. With the difference between freedom and jail hinging on the reliability of information you receive, would you really trust a Wrapster file that could have originated with any one of 20 million peer clients?

Non-Web encryption is more suspicious

Encrypted information can be recognized because of its unnatural entropy values (that is, the frequencies with which characters appear are not what is normally expected in the user's language). It is generally tolerated when it comes from web sites, probably because no country is eager to hinder online financial transactions. But especially when more and more states are charging ISPs with legal responsibility for their customers' online activities, encrypted code from a non-Web source will attract suspicion. Encryption may keep someone from reading what's passing through a server, but it never stops him from logging it and confronting the end user with its existence. In a country with relative Internet freedom, this isn't much of a problem. In one without it, the cracking of your key is not the only thing to fear.

I emphasize these concerns because current peer-to-peer systems show marked signs of having been created in relatively free countries. They are not designed with particular sensitivity to users in countries where stealth activities are easily turned into charges of subverting the state. States where privacy is the most threatened are the very states where, for your own safety, you must not take on the government: if they want to block a web site, you need to let them do so for your own safety.

Many extant peer-to-peer approaches offer other ways to get at a site's information (web proxies, for example), but the information they provide tends to be untrustworthy and the method for obtaining it difficult or dangerous.

Red Rover offers the benefits of peer-to-peer technology while offering a clientless alternative to those taking the risk behind the firewall. The Red Rover anticensorship strategy does not require the information seeker to download any software, place any incriminating programs on her hard drive, or create any two-

Microsoft Exhibit 1019

way electronic trails with information providers. The benefactor of Red Rover needs only to know how to count and how to operate a web browser to access a web-based email account.

Red Rover is technologically very "open" and will hopefully succeed at traversing censorship barriers not by electronic stealth but by simple brute force. The Red Rover distributed clients create a population of contraband providers which is far too large, changing, and growing for any nation's web-blocking software to keep up with.

Architecture

Red Rover is designed to keep a channel of information open to those behind censorship walls by exploiting some now mundane features of the Internet, such as dynamic IP addresses and the unbalanced ratio of Red Rover clients to censors. Operating out in the open at a low-tech level helps keep Red Rover's benefactors from appearing suspicious. In fact, Red Rover makes use of aspects of the current Internet that other projects consider liabilities, such as the impermanent connections of ordinary Internet users and the widespread use of free, web-based email services. The benefactors, those behind the censorship barrier (hereafter, "subscribers"), never even need to see a Red Rover client application: users of the client are in other countries.

The following description of the Red Rover strategy will be functional (i.e., top-down) because that is the best way to see the rationale behind decisions that make Red Rover unique among peer-to-peer projects. It will be clear that the Red Rover strategy openly and necessarily embraces human protocols, rather than performing all of its functions at the algorithmic level. The description is simplified in the interest of saving space.

The Red Rover application is not a proxy server, not a site mirror, and not a gate allowing someone to surf the Web through the client. The key elements of the system are hosts on ordinary dial-up connections run by Internet users who volunteer to download data that the Red Rover administrator wants to provide. Lists of these hosts and the content they offer, changing rapidly as the hosts come and go over the course of a day, are distributed by the Red Rover hub to the subscribers. The distribution mechanism is done in a way that minimizes the risk of attracting attention.

It should be clear, too, that Red Rover is a strategy, not just the software application that bears the name. Again, those who benefit the most from Red Rover will never see the program. The strategy is tripartite and can be summarized as

Microsoft Exhibit 1019

follows. (The following sentence is deliberately awkward, for reasons explained in the next section.)

3 simple layers: the hub, the client, & sub scriber.

The hub

The hub is the server from which all information originates. It publishes two types of information.

First, the hub creates packages of HTML files containing the information the hub administrator wants to pass through the censorship barrier. These packages will go to the clients at a particular time. Second, the hub creates a plain text, email notification that explains what material is available at a particular time and which clients (listing their IP addresses) have the material. The information may be encoded in a nontraditional way that avoids attracting attention from software sniffers, as described later in this chapter.

The accuracy of these text messages is time-limited, because clients go on- and offline. A typical message will list perhaps 10 IP addresses of active clients, selected randomly from the hub's list of active clients for a particular time.

The hub distributes the HTML packages to the clients, which can be done in a straightforward manner. The next step is to get the text messages to the subscribers, which is much trickier because it has to be done in such a way as to avoid drawing the attention of authorities that might be checking all traffic.

The hub would never send a message directly to any subscriber, because the hub's IP address and domain name are presumed to be known to authorities engaged in censorship. Instead, the hub sends text messages to clients and asks them to forward them to the subscribers. Furthermore, the client that forwards this email would never be listed in its own outgoing email as a source for an HTML package. Instead, each client sends mail listing the IP addresses of *other* clients. The reason for this is that if a client sent out its own IP address and the subscriber were then to visit it, the authorities could detect evidence of two-way communication. It would be much safer if the notification letter and the subscriber's decision to surf took different routes.

The IP addresses on these lists are "encrypted" at the hub in some nonstandard manner that doesn't use hashing algorithms, so that they don't set off either entropy or pattern detectors. For example, that ungrammatical "3 simple layers" sentence at the end of the last section would reveal the IP address 166.33.36.137 to anyone who knew the convention for decoding it. The convention is that each digit in an IP address is represented by the number of letters in a word, and

Microsoft Exhibit 1019

octets are separated by punctuation marks. Thus, since there is 1 letter in "3," 6 in "simple," and 6 in "layers," the phrase "3 simple layers" yields the octet 166 to someone who understands the convention.

Sending a list of 10 unencoded IP addresses to someone could easily be detected by a script. But by current standards, high-speed extraction of any email containing a sentence with bad grammar would result in an overwhelming flood of false positives. The "encryption" method, then, is invisible in its overtness. Practical detection would require a great expenditure of human effort, and for this reason, this method should succeed by its pure brute force. The IP addresses will get through.

The hub also keeps track of the following information about the subscriber:

- Her web-based email address, allowing her the option of proxy access to email and frequent address changes without overhead to the hub.
- The dates and times that she wishes to receive information (which she could revise during each Red Rover client visit, perhaps via SSL, in order to avoid identifiable patterns of online behavior).
- Her secret key, in case she prefers to take her chances with encrypted list notifications (an option Red Rover would offer).

The clients

The clients are free software applications that are run on computers around the world by ordinary, dial-up Internet users who volunteer to devote a bit of their system usage to Red Rover. Clients run in the background and act as both personal web servers and email notification relays. When the user on the client system logs on, the client sends its IP address to the hub, which registers it as active. For most dial-up accounts, this means that, statistically, the IP will differ from the one the client had for its last session. This simple fact plays an important role in client longevity, as discussed below.

Once the client is registered, the hub sends it two things. The first is an HTML package, which the client automatically posts for anyone accessing the IP address through a browser. (URL encryption would be a nice feature to offer here, but not an essential one.)

The second message from the hub is an email containing the IP list, plus some filler to make sure the size of the message is random. This email will be forwarded automatically from the receiving Red Rover client to a subscriber's web-based email account. These emails will be generated in random sizes as an added frustration to automated censors which hunt for packet sizes.

Microsoft Exhibit 1019

The email list, with its unhashed encryption of the IP addresses, is itself fully encrypted at the hub and decrypted by a client-specific key by the client just before mailing it to the subscriber. This way, the client user doesn't know anything about who she's sending mail to. The client will also forward the email with a spoofed originating IP address so that if the email is undelivered, it will not be returned to the sender. If it did return, it would be possible for a malicious user of the client (censors and police, for example) to determine the subscriber's email address simply by reading it off of the route-tracing information revealed by any of a variety of publicly available products. Together with the use of web-based accounts for subscriber email, rather than ISP accounts, subscriber privacy will benefit from these precautions.

The subscribers

The subscriber's role requires a good deal of caution, and anyone taking it on must understand how to make the safest use of Red Rover as well as the legal consequences of getting caught. The subscriber's actions should be assumed, after all, to be entirely logged by the state or its agents from start to finish.

The first task of the subscriber is to use a side channel (a friend visiting outside the country, for instance, or a phone call or postal letter) to give the hub the information needed to maintain contact. She also needs to open a free web-based email account in a country outside the area being censored. Then, after she puts in place any other optional precautions she feels will help keep her under the authorities' digital radar (and perhaps real-life radar), she can receive messages and download controversial material. Figure 10-1 shows how information travels between the hub, clients, and servers.

In particular, it is wise for subscribers to change their notification times frequently. This decreases the possibility of the authorities sending false information or attempting to entrap a subscriber by sending a forged IP notification email (containing only police IPs) at a time they suspect the subscriber expects notification. If the subscriber is diligent and creates new email addresses frequently, it is far less likely that a trap will succeed. The subscriber is also advised to ignore any notification sent even one second different from her requested subscription time. Safe subscription and subscription-changing protocols involve many interesting options, but these will not be detailed here.

When the client is closed or the computer disconnected, the change is registered by the hub, and that IP address is no longer included on outgoing notifications. Those subscribers who had already received an email with that IP address on it would find it did not serve Red Rover information, if indeed it worked at

Microsoft Exhibit 1019

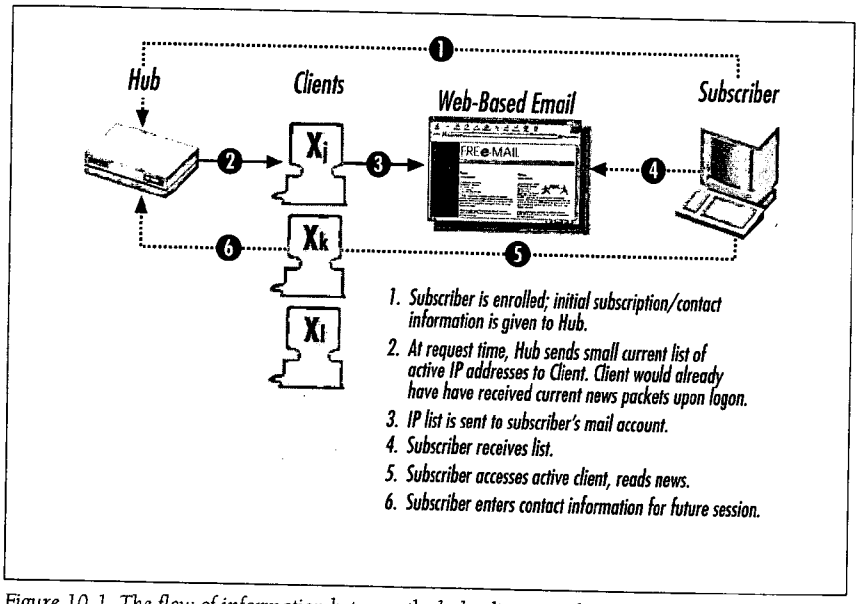

Figure 10-1. The flow of information between the hub, clients, and servers

all from the browser. The subscribers would then try the other IP addresses on the list. The information posted by the hub is identical on all clients, and the odds that the subscriber would find one that worked before all the clients on the list disconnect are quite high.

Client life cycle

Every peer-to-peer system has to deal with the possibility that clients will disappear unexpectedly, but senescence is actually assumed for Red Rover clients. Use it long enough and, just as with tax cheating, they'll probably catch up with you. In other words, the client's available IPs will eventually all be blocked by the authorities.

The predominant way nations block web sites is by IP address. This generally means all four octets are blocked, since C-class blocking (blocking any of the possibilities in the fourth octet of the IP address) could punish unrelated web sites. Detection has so far tended to result not in prosecution of the web visitor, but only in the blocking of the site. In China, for example, it will generally take several days, and often two weeks, for a "subversive" site to be blocked.

The nice thing about a personal web server is that when a user logs on to a dial-up account, the user will most likely be assigned a fourth octet different from the one she had in previous sessions. With most ISPs, odds are good of getting a

Microsoft Exhibit 1019

different third octet as well. This means that a client can sustain a great number of blocks before becoming useless, and, depending on the government's methods (and human workload), many events are likely to evade any notice whatsoever. But whenever the adversary is successful in completely blocking a Red Rover client's accessible IP addresses, that's the end of that client's usefulness—at least until the user switches ISPs. (Hopefully she'll choose a new ISP that hasn't been blocked due to detection of *another* Red Rover client.) Some users can make their clients more mobile, and therefore harder to detect, by subscribing to a variety of free dial-up services.

A fact in our favor is that it is considered extremely unlikely that countries will ever massively block the largest ISPs. A great deal of damage to both commerce and communication would result from a country blocking a huge provider like, for example, America Online, which controls nearly a quarter of the American dial-up market. This means that even after many years of blocking Red Rovers, there will still always be virgin IPs for them. Or so we hope.

The Red Rover strategy depends upon a dynamic population. On one level, each user can stay active if she has access to abundant, constantly changing IP addresses. And at another level, Red Rover clients, after they become useless or discontinued, are refreshed by new users, compounding the frustration of would-be blockers.

The client will be distributed freely at software archives and partner web sites after its release, and will operate without user maintenance. A web site (see "Acknowledgments") is already live to provide updates and news about the strategy, as well as a downloadable client.

Putting low-tech "weaknesses" into perspective

Red Rover creates a high-tech relationship between the hub and the client (using SL and strong encryption) and a low-tech relationship between the client and the subscriber. Accordingly, this latter relationship is inherently vulnerable to security-related difficulties. Since we receive many questions challenging the viability of Red Rover, we present below in dialogue form our responses to some of these questions in the hope of putting these security "weaknesses" into perspective.

Skeptic:

> I understand that the subscriber could change subscription times and addresses during a Red Rover visit. But how would anyone *initially* subscribe? If subscription is done online or to an email site, nothing would pre-

Microsoft Exhibit 1019

vent *those* sites from being blocked. The prospective subscriber may even be at risk for *trying* to subscribe.

Red Rover:

True, the low-tech relationship between Red Rover and the client means that Red Rover must leave many of the steps of the strategy to the subscriber. As we've said above, another channel such as a letter or phone call (not web or email communication) will eventually be necessary to initiate contact since the Red Rover site and sites which mirror it will inevitably be victims of blocking. But this requirement is no different than other modern security systems. SSL depends on the user downloading a browser from a trusted location; digital signatures require out-of-band techniques for a certificate authority to verify the person requesting the digital signature.

This is *not* a weakness; it is a strength. By permitting a diversity of solutions on the part of the subscribers, we make it much harder for a government to stop subscription traffic. It also lets the user determine the solution ingredients she believes are safest for her, whether public key cryptography (legal, for now, in many blocking countries), intercession by friends who are living in or visiting countries where subscribing would not be risky, proxy-served requests to forward email to organizations likely to cooperate, etc.

We are confident that word of mouth and other means will spread the news of the availability of Red Rover. It is up to the subscriber, though, to first offer her invitation to crash the censorship barrier. For many, subscribing may not be worth the risk. But for every subscriber who gets information from Red Rover, word of mouth can also help hundreds to learn of the content.

If this response is not as systematic as desired, remember that prospective subscribers face vastly different risks based on their country, profession, technical background, criminal history, dependents, and other factors. Where a problem is not recursively enumerable, the best solution to it will rarely be an algorithm. A variety of subscription opportunities, combined with non-patterned choices by each subscriber, leads to the same kind of protection that encryption offers in computing: Both benefit from increased entropy.

Skeptic:

What is to stop a government from cracking the client and cloning their own application to entrap subscribers or send altered information?

Red Rover:

Red Rover has to address this problem at both the high-tech and low-tech levels. I can't cover all strategies available to combat counterfeiting, but I can lay out what we've accomplished in our design.

Microsoft Exhibit 1019

At the high-tech level, we have to make sure the hub can't be spoofed, that the client knows if some other source is sending data and pretending to be the hub. This is a problem *any* secure distributed system must address, and a number of successful peer-to-peer systems have already led the way in solving this problem. Red Rover can adopt one of these solutions for the relationship between the hub and clients. This aspect of Red Rover does not need to be novel.

Addressing this question for the low-tech relationship is far more interesting. An alert subscriber will know, to the second, what time she is to receive email notifications. This information is sent and recorded using an SSL-like solution, so if that time (and perhaps other clues) isn't present on the email, the subscriber will know to ignore any IP addresses encoded in it.

Skeptic:

Ah, but what stops the government from intercepting the IP list, altering it to reflect different IP addresses, and *then* forwarding it to the subscriber? After all, you don't use standard encryption and digest techniques to secure the list.

Red Rover:

First, we have taken many precautions to make it hard for surveillance personnel to actually notice or suspect the email containing the IP list. Second, remember that we told the subscribers to choose web-based email accounts outside the boundaries of the censoring country. If the email is waiting at a web-based site in the United States, the censoring government would have to intercept a message during the subscriber's download, determine that it contained a Red Rover IP address (which we've encoded in a low-tech manner to make it hard to recognize), substitute their own encoded IP address, and finish delivering the message to the subscriber. All this would have to be done in the amount of time it takes for mail to download, so as not to make the subscriber suspicious. It would be statistically incredible to expect such an event to occur.

Skeptic:

But the government could hack the web-based mail site and change the email content without the subscriber knowing. So there wouldn't be any delay.

Red Rover:

Even if this happened, the government wouldn't know when to expect the email to arrive, since this information was passed from the subscriber to the client via SSL. And if the government examined and counterfeited every unread email waiting for the subscriber, the subscriber would know from our instructions that any email which is not received "immediately" (in

Microsoft Exhibit 1019

some sense based on experience) should be distrusted. It is in the subscriber's interest to be prompt in retrieving the web pages from the clients anyway, since the longer the delay, the greater the chance that the client's IP address will become inactive. Still, stagnant IP lists are far more likely to be useless than dangerous.

Skeptic:

A social engineering question, then. Why would anyone want to run this client? They don't get free music, and it doesn't phone E.T. Aren't you counting a little too much on people's good will to assume they'll sacrifice their valuable RAM for advancing human rights?

Red Rover:

This has been under some debate. Options always include adding file server functions or IRC capability to entice users into spending a lot of time at the sponsor's site. Another thought was letting users add their own, client-specific customized page to the HTML offering, one which would appear last so as not to interfere with the often slow downloading of the primary content by subscribers in countries with stiff Internet and phone rates and slow modems. This customized page could be pictures of their dog, editorials, or, sadly but perhaps crucially, advertising. Companies could even pay Red Rover users to post their ads, an obvious incentive. But many team members are rightfully concerned that if Red Rover becomes viewed as a mercantile tool, it would repel both subscribers and client users. These discussions continue.

Skeptic:

Where does the name "Red Rover" come from?

Red Rover:

Red Rover is a playground game analogous to the strategy we adopted for our anti-censorship system. Children form two equal lines, facing each other. One side invites an attacker from the other, yelling to the opposing line: "Red Rover, Red Rover, send Lena right over." Lena then runs at full speed at the line of children who issued the challenge, and her goal is to break through the barrier of joined arms and cut the line. If Lena breaks through, she takes a child back with her to her line; if she fails, she joins that line. The two sides alternate challenges until one of the lines is completely absorbed by the other.

It is a game, ultimately, with no losers. Except, of course, the kid who stayed too rigid when Lena rammed him and ended up with a dislocated shoulder.

We hope Red Rover leads to similar results.

Microsoft Exhibit 1019

Acknowledgments

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Red Rover was unveiled in April 2000 at *Outlook on Freedom*, Moscow, sponsored by the Human Rights Organization (Russia) and the National Press Institute, in a talk entitled "A Functional Strategy for an Online Anti-Blocking Remedy," delivered by the author. Red Rover's current partners include Anonymizer, Free Haven, Quintessenz, and VIP Reference. Updates about production progress and contact information about Red Rover will be posted at *http://redrover.org*.

Microsoft Exhibit 1019

chapter eleven

Publius

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Publius is a web-based publishing system that resists censorship and tampering. A file published with Publius is replicated across many servers, making it very hard for any individual or organized group to destroy the document. Distributing the document also provides resistance to so-called distributed denial of service (DDoS) attacks, which have been used in highly publicized incidents to make a resource unavailable. Another key feature of Publius is that it allows an individual to publish a document without providing information that links the document to any particular computer. Therefore, the publisher of a document can remain anonymous.

Publius has been designed with ease of access for end users in mind. HTML pages, images, or any other type of file can be published with the system. Documents published with Publius can be read with a standard web browser in combination with an HTTP proxy that can run locally or remotely. Files published with Publius are assigned a URL that can be entered into a web browser or embedded in a hyperlink.

The current architecture of the World Wide Web does not lend itself easily to censorship-resistant, anonymous publication. Published documents have a URL that can be traced back to a specific Internet host and usually a specific file owner. However, there are many reasons why someone might wish to publish something anonymously. Among the nobler of these reasons is political dissent or "whistleblowing." It is for these reasons that we designed Publius. Chapter 12 covers Free Haven, a project with some similarities, and provides more background on anonymity.

Anonymous publishing played an important role in the early history of the United States. James Madison, Alexander Hamilton, and John Jay collectively

Microsoft Exhibit 1019

wrote the Federalist Papers under the pen name Publius. This collection of 85 articles, published pseudonymously in New York State newspapers from October 1787 through May 1788, was influential in convincing New York voters to ratify the proposed United States Constitution. It is from these distinguished authors that our system gets its name.

Like many of the other systems in this book, Publius is seen from the outside as a unified system that works as a monolithic service, not as a set of individual Internet hosts. However, Publius consists of a set of servers that host content. These servers are collectively referred to as Publius Servers. The Publius Servers are independently owned and operated by volunteers located throughout the world. The system resists attack because Publius as a whole is robust enough to continue serving files even when many of the hosts go offline.

Publius uses two main pieces of software. The first is the server software, which runs on every Publius server. The second piece of software is the client software. This software consists of a special HTTP proxy that interfaces with a web browser and allows an individual to publish and retrieve files. In this chapter we use the terms proxy and client software interchangeably, as they both refer to the HTTP proxy. In order to use Publius an individual runs the proxy on their computer or connects to a proxy running on someone else's computer.

Why censorship-resistant anonymous publishing?

The publication of written words has long been a tool for spreading new (and sometimes controversial) ideas, often with the goal of bringing about social change. Thus the printing press, and more recently, the World Wide Web, are powerful revolutionary tools. But those who seek to suppress revolutions possess powerful tools of their own. These tools give them the ability to stop publication, destroy published materials, or prevent the distribution of publications. And even if they cannot successfully censor the publication, they may intimidate and physically or financially harm the author or publisher in order to send a message to other would-be revolutionaries that they would be well advised to consider an alternative occupation. Even without a threat of personal harm, authors may wish to publish their works anonymously or pseudonymously because they believe they will be more readily accepted if not associated with a person of their gender, race, ethnic background, or other characteristics.

Microsoft Exhibit 1019

Publius and other systems in this book

The focus of this book is peer-to-peer systems. While Publius is not a pure peer-to-peer system, it does share many characteristics with such systems. In addition, Publius provides unique and useful solutions to many of the problems faced by users and designers of such systems.

Distributed publishing tools and peer-to-peer file-sharing tools are still in their infancy. Many of these systems are changing very rapidly—each system continually gains new features or improves on old ones. This complicates any sort of direct comparison. However, in certain areas Publius does have some advantages over other file-sharing systems described in this book, such as Gnutella and Freenet. This is not to say that Publius is necessarily better than other systems. Indeed, in certain areas other systems offer marked advantages over Publius. Each system has its strengths and weaknesses.

One of Publius' strengths is that it allows a publisher (and only the publisher) to update previously published material in such a way that anyone retrieving the old version is automatically redirected to the newly updated document. Publius also allows a publisher to delete a published document from all of the servers it is stored on. Safeguards are in place to prevent anyone but the publisher from deleting or updating the published document. A tamper-check mechanism is built into the Publius URL. This allows the Publius client to verify that a retrieved document has not been tampered with.

Publius is one of a handful of file-sharing and publishing systems that are entirely implemented on top of the standard HTTP protocol. This makes Publius portable and simplifies installation as it easily interfaces with a standard web browser. By portable we mean that Publius can run on a variety of different operating systems with little or no modification. Of course, as with everything in life, there is a trade-off. Implementing Publius over HTTP means that Publius is not as fast as it could be. There is a slight overhead in using HTTP as opposed to implementing the communication between server and browser directly.

System architecture

The Publius system consists of a collection of web servers called Publius Servers. The list of web servers, called the Publius Server List, is known to all Publius clients. An individual can publish a document using the client software.

The first part of the publication process involves using the Publius client software to encrypt the document with a key. This key is split into many pieces,

Microsoft Exhibit 1019

called shares, such that only a small number of shares are required to form the key. For example, the key can be split into 30 shares such that any 3 of these shares can be used to form the key. But anyone combining fewer than 3 shares has no hint as to the value of the key. The choice of 3 shares is arbitrary, as is the choice of 30. The only constraint is that the number of shares required to form the key must be less than or equal to the total number of shares.

The client software then chooses a large subset of the servers listed in the Publius Server List and uploads the document to each one. It places the complete encrypted document and a single share on each server; each server has a different share of the key. The encrypted file and a share are typically stored on at least 20 servers. Three shares from any of these servers are enough to form the key.

A special URL called the Publius URL is created for each published document. The Publius URL is needed to retrieve the document from the various servers. This URL tells the client software where to look for the encrypted document and associated shares.

Upon receiving a Publius URL, the client software randomly retrieves three shares from the servers indicated by the URL. The shares are then combined to form the key. The client software also retrieves one copy of the encrypted file from one of the servers. The key is used to decrypt the file and a tamper check is then performed. If the document successfully passes the tamper check, it is displayed in the browser; otherwise, a new set of shares and a new encrypted document are retrieved from another set of servers.

The encryption prevents Publius server administrators from reading the documents stored on their servers. It is assumed that if server administrators don't know what is stored on their servers they are less likely to censor them. Only the publisher knows the Publius URL—it is formed by the client software and displayed in the publisher's web browser. Publishers can do what they wish with their URLs. They can post them to Usenet news, send them to reporters, or simply place them in a safe deposit box. To protect their identities, publishers may wish to use anonymous remailers when communicating these URLs.

The Publius client software is implemented as an HTTP proxy. Most web browsers can be configured to send web requests to an HTTP proxy, which retrieves the requested document (usually performing some extra service, such as caching, in the process) and returns it to the web browser. The HTTP proxy may be located on the user's computer or on some other computer on the Internet. In the case of Publius, the HTTP proxy is able to interpret Publius URLs, fetch the

Microsoft Exhibit 1019

necessary shares and encrypted documents, and return a decrypted document to the user's web browser.

Cryptography fundamentals

Before describing the Publius operations, we briefly introduce some cryptographic topics that are essential to all Publius operations. For more information about these cryptographic topics see an introductory cryptography text.*

Encryption and decryption

Encryption is the process of hiding a message's true content. An unencrypted message is called a *plaintext*, while a message in encrypted form is called a *ciphertext*.

A *cipher* is a function that converts plaintext to ciphertext or ciphertext back to plaintext. Rijndael, the Advanced Encryption Standard, is an example of a well-known cipher. Decryption is the process of converting ciphertext back to plaintext. The encryption and decryption processes require a key. Trying to decrypt a message with the wrong key results in gibberish, but when the correct key is used, the original plaintext is revealed. Therefore it is important to keep the key secret and to make sure it is virtually impossible for an adversary to guess.

Ciphers that use the same key to encrypt and decrypt messages are called *symmetric ciphers*. These are the type of ciphers used in Publius.

Secret sharing

A message can be divided into a number of pieces in such a way that combining only a fraction of those pieces results in the original message. Any combination of pieces is sufficient, so long as you have the minimum number required.

An algorithm that divides data in such a manner is called a secret sharing algorithm. The secret sharing algorithm takes three parameters: the message to divide, the number of pieces to divide the message into, and the number of pieces needed to reconstruct the message. The individual pieces are called shares. Publius uses Shamir's secret sharing algorithm. Other secret sharing algorithms also exist.

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* See, for example, Bruce Schneier (1996), *Applied Cryptography Protocols, Algorithms, and Source Code in C, 2nd Edition*, John Wiley & Sons.

Microsoft Exhibit 1019

Hash functions

A hash function takes a variable-length input and returns a fixed-length output. Publius uses the cryptographically strong hash functions MD5 and SHA-1. Cryptographically strong hash functions possess two properties. First, the hash function is hard to invert—that is, if someone is told the hash value, it is hard to find a message that produces that hash value. Second, it is hard to find two messages that produce the same hash value. By hard we mean that it is not feasible, even using massive amounts of computing power, to accomplish the specified task.

The slightest change to a file completely changes the value of the hash produced. This characteristic makes hash functions ideal for checking whether the content of a message has been changed. The MD5 hash function produces a 128-bit output and SHA-1 produces a 160-bit output.

Publius operations

Given the previous description of Publius-related cryptographic functions, we now describe the Publius operations Publish, Retrieve, Update, and Delete.

Publish operation

Suppose that we wish to publish the file *homepage.html* with Publius. The accompanying sidebar outlines the steps the Publius proxy follows to publish a file. First, a key is created from the MD5 and SHA-1 hash of the contents of the file *homepage.html*. This key is then used to encrypt the file, producing a new file we will call *homepage.enc*. Using Shamir's secret sharing algorithm, the key is split into 30 shares such that any 3 of these shares can be used to reconstruct the key. The first share is named *Share_1*, the second *Share_2*, and so on. The MD5 hash of the contents of *homepage.html* and *Share_1* is calculated. This MD5 hash results in a 128-bit number. An operation is performed on this number to determine an index into the Publius Server List. The Publius Server List is essentially just a numbered table of web servers, each running the Publius server software. The index is used to locate a particular server. For instance, the index value 5 corresponds to the 5th entry in the Publius Server List. You will recall that all Publius client software has the same list, and therefore the 5th server is the same for everyone.

For the sake of argument let's assume that our index number is 5 and that the 5th server is named *www.nyu.edu*. The proxy now attempts to store the file *homepage.enc* and *Share_1* on *www.nyu.edu*. The files are stored in a directory

Microsoft Exhibit 1019

derived from the previously calculated MD5 hash of *homepage.html* and *Share_1*. The file *homepage.enc* is stored in a file named *file* and *Share_1* is stored in a file named *share*. These same two names are used for every piece of content published with Publius, regardless of the type of the file. One of the reasons for storing *homepage.enc* as *file* rather than as *homepage.enc* is that we don't want to give anyone even a hint as to the type of file being stored. The neutrality of the name, along with the use of encryption so that no one can read the file without the key, allows Publius server administrators to plausibly deny any knowledge of the content of the files being hosted on the Publius server. While each server possesses a part of the encryption key, it is of no value by itself for decrypting the file. We thus expect that server administrators have little motive to delete, and thereby censor, files stored on their servers.

The whole process of performing the MD5 hash and storing the files on a Publius server is repeated for each of the 30 shares. A file is stored on a particular server only once—if Publius generates the same index number more than once, the corresponding share is discarded.

Each time a file and share are stored on a Publius server, the file and share's corresponding MD5 hash (calculated in line 5 of the sidebar "Process for publishing the file homepage.html in Publius") is used in the formation of the Publius URL. A Publius URL has the following form:

http://!publius!/ options MD5_hash MD5_hash MD5_hash...MD5_hash

where each *MD5_hash* is the hash defined in line 5 of the sidebar. Each *MD5_hash* is Base64-encoded to generate an ASCII representation of the hash value. Here is an example of a Publius URL:

http://!publius!/010310023/
VYimRS+9ajc=B20wYdxGsPk=kMCiu9dzSHg=xPTuzOyUnNk=/
O5uFb3KaC8I=MONUMmecuCE=P5WY8LS8HGY=KLQGrFwTcuE=/
kJyiXge4S7g=6I7LBrYWAV0=

The *options* part of the Publius URL is made up of several flags that specify how the proxy should interpret the URL. The options section includes a "do not update" flag, the number of shares needed to form the key, and the version number of the Publius client that published the URL.

The version number allows us to add new features to future versions of Publius while at the same time retaining backward compatibility.

The update flag determines whether the update operation can be performed on the Publius content represented by the URL. If the update flag is 1, the retrieval of updated content may be performed when update URLs are discovered. If the

Microsoft Exhibit 1019

Process for publishing the file homepage.html in Publius

1. Generate a key.
2. Using the key, encrypt file *homepage.html* to produce *homepage.enc*.
3. Perform Shamir's secret sharing algorithm on the key.
   This produces *Share_1*, *Share_2*...*Share_30*.
   Any three shares can be used to form the key.
4. Set *share* to *Share_1*.
5. Set *h* as the MD5 hash of *share* appended to content of file *homepage.html*.
6. Set *index* to *h* mod (the number of entries in the Publius Server List).
7. Set *server* to the Publius server at the location specified by *index*.
8. On *server*: Create a directory derived from *h*. In this directory store the contents of *homepage.enc* into a file named *file* and *share* into a file named *share*.

Repeat steps 4 through 8 once for each of the remaining shares (*Share_2*... *Share_30*), setting the variable *share* appropriately before each repetition.

update flag is 0, however, the client ignores update URLs sent by Publius servers in response to share and encrypted file requests.

The options part of the Publius URL also includes a number that indicates the size of the Publius Server List at the time the file was published. The Publius Server List is not static—it can grow over time. Servers can be added without affecting previously published files. The index calculation performed on line 6 of the Publius Publish algorithm (see the sidebar "Process for publishing the file homepage.html in Publius") depends on the size of the Publius Server List. Changes to this value change the computed index location. Therefore it is necessary to store this value in the URL. When interpreting a given Publius URL, the proxy essentially ignores all entries in the server list with index greater than the one stored in the Publius URL. This ensures that the proxy will calculate the correct index value for every server hosting the shares and encrypted file.

Retrieve operation

Upon receiving a request to retrieve a Publius URL, the proxy first breaks the URL into its MD5 hash components. As the size of each MD5 hash is exactly 128 bits, this is an easy task. As you may recall, each of these hash values determines which servers in the Publius Server List are storing the encrypted file and

Microsoft Exhibit 1019

a share. In order to retrieve the encrypted file and share, the proxy randomly selects one of the hash values and performs the same operation performed by the Publish operation (line 6 in the sidebar). The value returned is used as an index into the Publius Server List, revealing the name of the server. The proxy retrieves the encrypted file and share file from the server. Recall that the file named *file* contains the encrypted version of the published file and the file named *share* contains a single share. In order to form the key, the proxy needs to find two additional shares. Thus, the client selects two other MD5 hash values randomly from the Publius URL and performs the same operation as before on each. This reveals two other servers that in turn lead to two more shares. The 3 shares can now be combined to form the key used to encrypt the file.

During the Publish operation, the key was broken into 30 shares. Assume that after testing each of these shares, the proxy ends up storing the encrypted file and a corresponding share on 20 servers. This means that 20 MD5 hashes appear in the Publius URL. During the retrieval process only 3 of these 20 shares are needed. Publius derives its fault-tolerant and censorship-resistant properties from the storage of these additional shares and encrypted files. By fault tolerant we mean that if for some reason several servers are unavailable, the proxy can still successfully retrieve the Publius document. In fact, if the file is stored on 20 servers, even if 17 servers are unavailable we can successfully retrieve the Publius document. However, if 18 Publius servers are unavailable, the Publius document cannot be retrieved because 2 shares are not enough to form the key needed to decrypt the content.

The additional copies also provide censorship resistance—if several Publius server administrators decide to delete the encrypted files and shares corresponding to a particular Publius file, the file can still be retrieved if at least three servers still contain the shares and encrypted file. With Publius servers located throughout the world, it becomes increasingly difficult to force Publius server administrators to delete files corresponding to a particular Publius URL, by legal or other means.

Many of the other systems in this book also have fault-tolerant features. However, most of these systems focus on maintaining a network of nodes with variable connectivity and temporary network addresses. Publius does not address the use of servers with temporary network addresses.

Once the key has been reconstructed from the shares, it can be used to decrypt the file. The decrypted file can now be displayed in the web browser. However, just before the file is displayed in the web browser, a tamper check is initiated. The tamper check verifies that the file has not changed since the time it

Microsoft Exhibit 1019

was initially published. The MD5 hashes stored in the URL are used to perform the tamper check. The hash was formed from the unencrypted file and a share—both of which are now available. Therefore, the client recalculates the MD5 hash of the unencrypted file and of each share (as in line 5 in the sidebar). If the calculated hashes do not match the corresponding hashes stored in the URL, the file has been tampered with or corrupted. In this case, the proxy simply throws away the encrypted file and shares and tries another set of encrypted files and shares. If a tamper check is successfully performed, the file is sent to the web browser. If the proxy runs out of share and encrypted file combinations, a message appears in the browser stating that the file could not be retrieved.

Update operation

Files, especially web pages, change over time. An individual may find a particular web document interesting and add it to his collection of bookmarks or link to it from a web page. The problem with linking to a Publius URL is that if anyone changes the document and tries to republish it, a new Publius URL is generated for the document. Therefore, anyone linking to the old document may never learn that the document has been updated because the link or bookmark still points to the older Publius document.

To remedy this situation, Publius supports an Update operation. The operation allows the publisher of a document to replace an older version of the Publius document with a newer one while still retaining the old URL. This is accomplished by allowing a Publius URL to be stored in a file called *update* in the same directory where the old version of the file resided.

For example, let's say that one encrypted file and share are stored on *www.nyu.edu* in directory *pubdir*. Upon receiving the update command, the proxy contacts the server *www.nyu.edu*, deletes the files named *file* and *share* from the directory *pubdir,* and places the new Publius URL in a file named *update*. Of course, the Update command is issued to all servers holding copies of the file to be updated.

Now, whenever *www.nyu.edu* receives a request for the encrypted file or share in the directory *pubdir,* the server sends the new Publius URL found in the *update* file. If several of the queried Publius servers also respond with this same Publius URL, the proxy retrieves the document referenced by the new Publius URL. Therefore, whenever a proxy requests the old file it is automatically redirected to the updated version of the file.

Of course, we want only the publisher of the document to be able to perform the Update command. In order to enforce this, the Publish operation allows a

Microsoft Exhibit 1019

password to be specified. This password is stored in the file *password* and is checked by the server during an Update operation. In order for this scheme to work, the password must be stored on each server so that the server can check that the password sent with the Update command matches the stored password. However, simply storing the password on the server would be dangerous, because it would permit Publius server administrators to update the document on all servers if they discover the corresponding URL. This is essentially a form of censorship, as the original file would no longer be accessible. So instead of simply storing the password, we store the MD5 hash of the password appended to the domain name of the particular server. The server stores this value in the password file associated with the particular document. The hash by itself provides no clues as to the actual value of the password, so it cannot be used to update the document on all of the servers.

Delete operation

There are circumstances in which a publisher may wish to delete a document from Publius. Publius therefore supports the Delete operation. Only the publisher may delete the document. The same password that controls the Update operation also ensures that only the publisher can perform the Delete operation.

The ability to delete Publius documents gives an adversary the option of trying to force the publisher of a Publius document to delete it. In order to prevent this scenario, Publius provides a "do not delete" option during the Publish operation. This option allows someone to publish a document in such a way that Publius servers deny requests to delete the document.

Of course, nothing stops a Publius server administrator from deleting the document from her own server, but the safeguards in this section do prevent a single person from deleting the Publius file from all the servers at once.

Both the Delete and Update commands attempt to make the required changes on all of the relevant servers. For example, the Update command tries to update every server storing a particular document. However, this may not always be possible due to a server being down or otherwise unavailable. This could lead to an inconsistent state in which some servers are updated and others are not. Although Publius does not currently deal with the problem of an inconsistent state, it does report the names of the servers on which the operation failed. At a later time, the Update command can be executed again in an attempt to contact the servers that failed to get updated. The same is true for the Delete command.

Microsoft Exhibit 1019

Publius implementation

Publius is a working system that has been in operation since August 2000. In the following sections, we describe several important aspects of the implementation. As you will recall, Publius consists of both client and server software. All Publius servers run the server software. The client software consists of a special HTTP proxy that interfaces with any standard web browser. This special proxy handles all Publius commands and therefore interacts with the Publius servers. Upon connecting to the proxy, the web browser displays the Publius User Interface. This user interface is essentially an HTML form that allows an individual to select a Publius operation (Delete, Publish, or Update). This form is not required for the Retrieve operation as it is the default operation.

User interface

The web browser interface, as shown in Figure 11-1, allows someone to select the Publius operation (Delete, Publish, or Update) and enter the operation's required parameters such as the URL and password. Each Publius operation is bound to a special *!publius!* URL that is recognized by the proxy. For example, the Publish URL is *http://!publius!PUBLISH*. The operation's parameters are sent in the body of the HTTP POST request to the corresponding *!publius!* URL. The proxy parses the parameters and executes the corresponding Publius operation. An HTML message indicating the success or failure of the operation is returned. If the Retrieve operation is requested and is successful, the requested document is displayed in a new web browser window.

Server software

To participate as a Publius server, one needs to install the Publius CGI script on a system running an HTTP server. The client software communicates with the server by executing an HTTP POST operation on the URL corresponding to the server's CGI script. The requested operation (Retrieve, Update, Publish, or Delete), the filename, the password, and any other required information is passed to the server in the body of the POST request.

Client software

The client software consists of the special HTTP proxy. The proxy transparently sends non-Publius URLs to the appropriate servers and passes the returned content back to the browser. Upon receiving a request for a Publius URL, the proxy retrieves the encrypted document and shares, as described in the "Retrieve operation" section earlier. The proxy also handles the Delete, Publish, and Update commands.

Microsoft Exhibit 1019

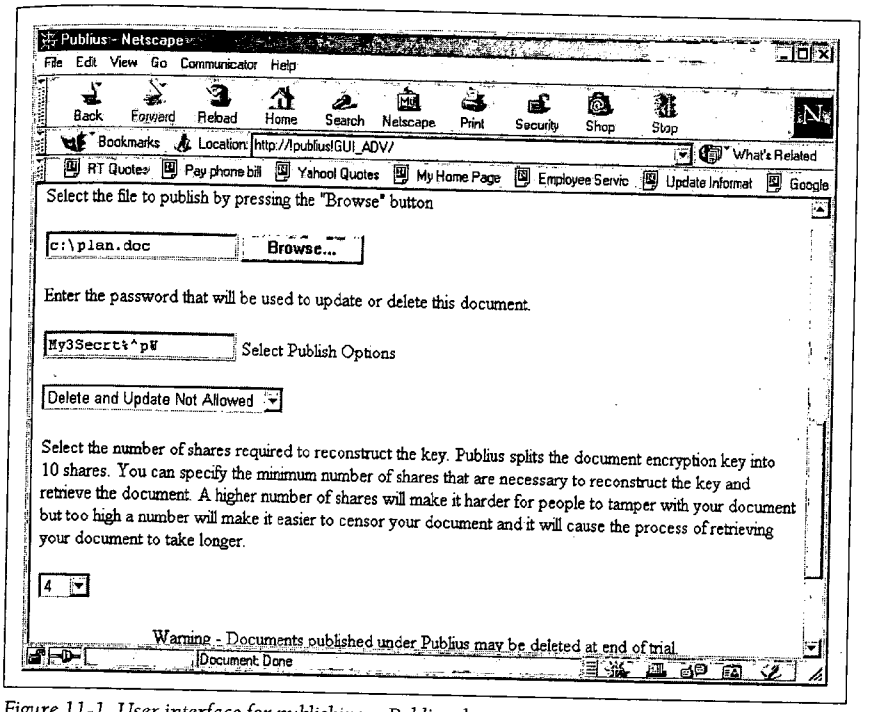

Figure 11-1. User interface for publishing a Publius document

Publius MIME type

The filename extension of a particular file usually determines the way in which a web browser or other software interprets the file's content. For example, a file that has a name ending with the extension *.html* usually contains HTML. Similarly, a file that has a name ending with the extension *.jpg* usually contains a JPEG image. The Publius URL does not retain the file extension of the file it represents. So the Publius URL gives no hint to the browser, or anyone else for that matter, as to the type of file it points to. However, in order for the browser to correctly interpret the byte stream sent to it by the proxy, the proxy must properly identify the type of data it is sending. Therefore, before publishing a file, Publius prepends the first three-letters of the file's name extension to the file. The file is then published as described earlier, in the section "Publish operation." When the proxy is ready to send the requested file back to the browser, the three-letter extension is removed from the file and checked to determine an appropriate MIME type for the document. The MIME type is sent in an HTTP Content-type header. If the three-letter extension is not helpful in determining the MIME type, a default type of *text/plain* is sent for text files. The default MIME type for binary files is *octet/stream*.

Microsoft Exhibit 1019

Publius in a nutshell

Documents are published in a censorship-resistant manner
This is partially achieved by storing the encrypted document and a share on a large number of servers.

Retrieved documents can be tamper-checked
The Publius URL is made up of MD5 hashes that allow the document to be checked for changes since publication.

Published documents can be updated
Any requests for the previous document are redirected to the new document.

Published documents can be securely deleted
A password mechanism is utilized for the Delete and Update commands.

A document can be anonymously published
Once the document is published there is no way to directly link the document to the publisher. However, indirect mechanisms of identification may exist, so one may wish to use an anonymizing proxy or publish the file in a cyber café or library.

The stored document is resistant to distributed denial of service attacks
The published file can still be retrieved even if a large number of servers are unavailable.

The Publius web site is *http://www.cs.nyu.edu/waldman/publius*. The source code, a technical paper describing Publius, and instructions for using Publius are available at this site.

Microsoft Exhibit 1019

chapter twelve

Free Haven

Roger Dingledine, Reputation Technologies, Inc., Michael J. Freedman, MIT, and David Molnar, Harvard University

The Free Haven Project is dedicated to designing a system of anonymous storage that resists the attempts of powerful adversaries to find or destroy any stored data. Our goals include the following:

Anonymity
: We try to meet this goal for all parties: the publishers that insert documents, the readers that retrieve documents, and the servers that store documents.

Persistence
: The publisher of a document—not the servers holding the document—determines its lifetime.

Flexibility
: The system functions smoothly as servers are added or remove themselves.

Accountability
: We apply a reputation system to servers that attempts to limit the damage done by those that misbehave.

In this chapter, we'll show how Free Haven tries to meet these goals. We spend a particularly large amount of time on anonymity. It is not adequate to speak of "anonymity" as a monolithic concept. In the section "Anonymity for anonymous storage," we'll enumerate the many different kinds of anonymity that are important to protect participants in the system.

Free Haven differs from the other projects in this book in the wide range of difficult goals we have taken on. We try to assure anonymity, server accountability,

Microsoft Exhibit 1019

and persistent storage for data independent of its popularity, all at the same time. Here are some comparisons to other projects:

Gnutella

The strength of Gnutella is its extremely flexible network design. But when a search is performed, servers respond with an external IP address or URL where the user can download the document. Since this actual retrieval is done without any privacy protection, using Gnutella is not a good choice if publishers or readers want anonymity. Further, documents in the Gnutella network last only as long as their host servers; when a user logs out for the night, all of his files leave with him.

Freenet and Mojo Nation

These systems make files highly accessible and offer some level of anonymity. But since the choice to drop a file is a purely local decision, and since files that aren't requested for some time tend to disappear automatically, these systems don't guarantee a specified lifetime for a document. Indeed, we expect that Freenet will provide a very convenient service for porn and popular audio files, but anything less popular will be driven off the system.

Publius

This project is closest to ours, because it addresses file storage rather than easy accessibility. But Publius provides no smooth decentralized support for adding new servers and excising dead or malicious servers. More importantly, Publius provides no accountability—there is no way to prevent publishers from entirely filling the system with garbage data.

Currently, Free Haven sacrifices efficiency and convenience to achieve its design requirements. Free Haven is designed more for anonymity and persistence of documents than for frequent querying. We expect that interesting material will be retrieved from the system and published in a more accessible fashion (such as in Freenet or normal web pages). Then the document in Free Haven will only need to be accessed if the other sources are shut down or the reader requires stronger anonymity. For more discussion of such "gatewaying" issues, refer to Chapter 19, *Interoperability Through Gateways*.

Privacy in data-sharing systems

Privacy is a term with positive connotations that every person can appreciate. One key way to achieve privacy, however—anonymity—is widely misunderstood both in daily life and in computer networking. The media and politicians stress socially disapproved activities (such as the exchange of unauthorized

Microsoft Exhibit 1019

music files or erotic pictures involving children) and ignore the important contributions that anonymity provides. Anonymity is used on an everyday basis in forums ranging from radio shows to Usenet newsgroups, by people who suffer from child abuse, drug dependency, or other social problems.

Anonymous publication and storage services allow individuals to speak freely without fear of persecution. Political dissidents must publish their views in order to reach enough people for their criticisms of a regime to be effective, yet they and their readers require anonymity at the same time. Less extreme examples involve cases in which a large and powerful private organization attempts to silence its critics by attacking either the critics themselves or those who make the criticism publicly available.

Developers and potential users of other peer-to-peer systems should be interested in the techniques we are developing to preserve anonymity in Free Haven, because they may prove useful in protecting the privacy of users in other systems as well. Many people would like to participate in communities and share information without revealing who they are. Their reasons may range from the trivial—such as avoiding spam—to deep social concerns. It is time to face these concerns directly so solutions can be designed fundamentally into peer-to-peer systems.

Peer-to-peer systems that attempt to address anonymity are just starting to be deployed, and the exact requirements and design choices are not yet clear. Recent events have highlighted some shortcomings of current systems. For instance, the limitations of Gnutella were dramatized by the Gnutella Wall of Shame, where someone lured readers to a web site by claiming to offer child pornography and then published each visitor's IP address. While Napster allowed people with MP3 files to find each other, it also made it easy for the band Metallica to find people who were offering unauthorized copies of Metallica songs and force them off the system.

These shortcomings cause people to look toward a new generation of anonymous publication services that address anonymity. In developing Free Haven, we hope to clarify some of the requirements for such systems and highlight the design choices.

Reliability with anonymity

In the physical world, people use safety deposit boxes to protect valuable items. Everything from passports to house titles to krugerrands—if it's important, it goes in the box, which is kept at the local bank. The bank has armed guards,

Microsoft Exhibit 1019

smiling tellers, and a history going back to the Knights Templar. Now suppose someone suggested to you that instead of going to the bank, it would be a better idea to hand your gold bars to the next guy on the street and ask him to "just hold these for a bit." You'd look at a person with such notions as though he had three heads... yet in some sense, this is exactly what distributed peer-to-peer file-sharing systems like Free Haven ask you to do.

The critical point is that for a safety deposit box, the only thing that really matters is reliability and availability: can you get your items when you want them? The rest is irrelevant. If the guy on the street could guarantee that you'll get your gold back and follow through, he would be "just as good" as the bank. In fact, if you're interested in protecting your privacy, the guy on the street may be better—he doesn't know or care who you really are. Of course, in the physical world, it's still a bad idea to give gold bars to random people on the street. Online, however, cryptography allows things to work out differently.

Many systems in addition to Free Haven need reliability, particularly peer-to-peer systems that ask people to share resources. When offering and retrieving resources, users want to preserve their privacy. When evaluating whether to transfer custody of their resources to another party on the system, users want to know whether that party can be trusted.

Initially these goals seem mutually exclusive, but the solution is to allow users to have *pseudonyms*, and to assign a *reputation* to each pseudonym. Free Haven differs from other systems in that the servers in the Free Haven system are known only by their pseudonyms, and we provide an automated system to track reputations (honesty and performance) for each server. A server's reputation influences how much data it can store in Free Haven and provides an incentive to act correctly. Reputation can be a complex matter—just think of all the reader reviews and "People also bought..." ratings on the Amazon.com retail site—so we'll leave its discussion to Chapter 16, *Accountability*, and Chapter 17, *Reputation*. Establishing trust through the use of pseudonyms is covered in Chapter 15, *Trust*.

What lets a malicious adversary find a person in real life? One way is to know his or her *true name*, a term first used in a short story by fiction author Vernor Vinge* and popularized by Tim May.† The true name is the legal identity of an

* Vernor Vinge (1987), *True Names... and Other Dangers*, Baen.

† Tim May, *Cyphernomicon*, *http://www-swiss.ai.mit.edu/6805/articles/crypto/cypherpunks/cyphernomicon*.

Microsoft Exhibit 1019

individual and can be used to find an address or other real-life connection. Obviously, a pseudonym should not be traceable to a true name.

As an author can use a pseudonym to protect his or her true name, in a computerized storage system a user can employ a pseudonym to protect another form of identity called *location*. This is an IP address or some other aspect of the person's physical connection to the computer system. In a successful system, a pseudonym always reflects the activities of one particular entity—but no one can learn the true name or location of the entity. The ability to link many different activities to a pseudonym is the key to supporting reputations.

Anonymity for anonymous storage

The word "anonymous" can mean many different things. Indeed, some systems claim "anonymity" without specifying a precise definition. This introduces a great deal of confusion when users are trying to evaluate and compare publishing systems to understand what protections they can expect from each system.

A publishing situation creates many types of anonymity—many requirements that a system has to meet in order to protect the privacy of both content providers and users. Here, we'll define the *author* of a document as whoever initially created it. The author may be the same as or different from the *publisher*, who places the document into Free Haven or another storage system. Documents may have *readers*, who retrieve the document from the system. And many systems, including Free Haven, have *servers*, who provide the resources for the system, such as disk space and bandwidth.

Free Haven tries to make sure that no one can trace a document back to any of these people—or trace any of them forward to a document. In addition, we want to prevent adversaries who are watching both a user and a document from learning anything that might convince them that the user is connected to that document. Learning some information that might imply a connection allows "linking" the user to that action or document. Thus, we define the following types of anonymity:

Author-anonymity
: A system is author-anonymous if an adversary cannot link an author to a document.

Publisher-anonymity
: A system is publisher-anonymous if it prevents an adversary from linking a publisher to a document.

Microsoft Exhibit 1019

Reader-anonymity

To say that a system has reader-anonymity means that a document cannot be linked with its readers. Reader-anonymity protects the privacy of a system's users.

Server-anonymity

Server-anonymity means no server can be linked to a document. Here, the adversary always picks the document first. That is, given a document's name or other identifier, an adversary is no closer to knowing which server or servers on the network currently possess this document.

Document-anonymity

Document-anonymity means that a server does not know which documents it is storing. Document-anonymity is crucial if mere possession of some file is cause for action against the server, because it provides protection to a server operator even after his or her machine has been seized by an adversary. This notion is sometimes also known as "plausible deniability," but see below under query-anonymity. There are two types of document-anonymity: isolated-server and connected-server.

Passive-server document-anonymity means that if the server is allowed to look only at the data that it is storing, it is unable to figure out the contents of the document. This can be achieved via some sort of secret sharing mechanism. That is, multiple servers split up either the document or an encryption key that recreates the document (or both). An alternative approach is to encrypt the document before publishing, using some key which is external to the server—Freenet takes this approach. Mojo Nation takes a different approach to get the same end: it uses a "two-layer" publishing system, in which documents are split up into shares, and then a separate "share map" is similarly split and distributed to participants called *content trackers*. In this way, servers holding shares of a document cannot easily locate the share map for that document, so they cannot determine which document it is.

Active-server document-anonymity refers to the situation in which the server is allowed to communicate and compare data with all other servers. Since an active server may act as a reader and do document requests itself, active-server document-anonymity seems difficult to achieve without some trusted party that can distinguish server requests from "ordinary" reader requests.

Query-anonymity

Query-anonymity means that the server cannot determine which document it is serving when satisfying a reader's request. A weaker form of query-anonymity is *server deniability*—the server knows the identity of the requested

Microsoft Exhibit 1019

document, but no third party can be sure of its identity. Query-anonymity can provide another aspect of plausible deniability.

Partial anonymity

Often an adversary can gain some partial information about the users of a system, such as the fact that they have high-bandwidth connections or all live in California. Preventing an adversary from obtaining *any* such information may be impossible. Instead of asking "Is the system anonymous?" the question shifts to "Is it anonymous enough?"

We might say that a system is *partially anonymous* if an adversary can only narrow down a search for a user to one of a "set of suspects." If the set is large enough, it is impractical for an adversary to act as if any single suspect were guilty. On the other hand, when the set of suspects is small, mere suspicion may cause an adversary to take action against all of them.

The design of Free Haven

Free Haven offers a community of servers called the *servnet*. Despite the name, all servers count the same, and within the servnet Free Haven is a peer-to-peer system. There are no "clients" in the old client/server sense; the closest approximation are users looking for files and potential publishers. Users query the entire servnet at once, not any single server in particular. Potential publishers do convince a single server to publish a document, but the actual publishing of a document is done by a server itself in a peer-to-peer fashion.

All of these entities—server, reader, and publisher—make up the Free Haven players. Thanks to pseudonymity, nobody knows where any server is located—including the one they use as their entry point to the system. Users query the system via broadcast.

Servers don't have to accept just any document that publishers upload to them. That would permit selfish or malicious people to fill up the available disk space. Instead, servers form contracts to store each other's material for a certain period of time.

Successfully fulfilling a contract increases a server's reputation and thus its ability to store some of its own data on other servers. This gives an incentive for each server to behave well, as long as cheating servers can be identified. We illustrate a technique for identifying cheating servers in the section "Accountability and the buddy system." In the section "Reputation system," we discuss the system that keeps track of trust in each server.

Microsoft Exhibit 1019

Some of these contracts are formed when a user inserts new data into the servnet through a server she operates. Most of them, however, are formed when two servers swap parts of documents (shares) by *trading*. Trading allows the servnet to be *dynamic* in the sense that servers can join and leave easily and without special treatment. To join, a server starts building up a reputation by storing shares for others—we provide a system where certain servers can act as *introducers* in order to smoothly add new servers. To leave, a server trades away all of its shares for short-lived shares, and then waits for them to expire. The benefits and mechanisms of trading are described in the later section "Trading."

The following sections explain how the design of Free Haven allows it to accomplish its goals. "Elements of the system" describes the design of the Free Haven system and the operations that it supports, including the insertion and retrieval of documents. We describe some potential attacks in the section "Attacks on Free Haven" and show how well the design does (or does not) resist each attack. We then compare our design to other systems aimed at anonymous storage and publication using the kinds of anonymity described in the section "An analysis of anonymity," allowing us to distinguish systems that at first glance look very similar. We conclude with a list of challenges for anonymous publication and storage systems, each of which reflects a limitation in the current Free Haven design.

Elements of the system

This chapter focuses on Free Haven's publication system, which is responsible for storing and serving documents. Free Haven also has a communications channel, which is responsible for providing confidential and anonymous communications between parties. Since this communications channel is implemented using preexisting systems that are fairly well known in the privacy community, we won't discuss it here. On the other hand, the currently available systems are largely insufficient for our accountability requirements; see Chapter 16.

The agents in our publication system are the author, publisher, server, and reader. As we stated in the section "Anonymity for anonymous storage," authors are agents that produce documents and wish to store them in the service, publishers place the documents in the storage system, servers are computers that store data for authors, and readers are people who retrieve documents from the service.

These agents know each other only by their pseudonyms and communicate only using the secure communications channel. Currently, the pseudonyms are pro-

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vided by the Cypherpunks remailer network,* and the communications channel consists of remailer reply blocks provided by that network. Each server has a public key and one or more reply blocks, which together can be used to provide secure, authenticated, pseudonymous communication with that server. Every machine in the servnet has a database that contains the public keys and reply blocks of other servers in the servnet.

As we said in the section "The design of Free Haven," documents are split into pieces and stored on different servers; each piece of a document is called a *share*. Unlike Publius or Freenet, servers in Free Haven give up something (disk space) and get other servers' disk space in return. In other words, you earn the right to store your data on the rest of the servnet after you offer to store data provided by the rest of the servnet.

The servnet is dynamic: shares move from one server to another every so often, based on each server's trust of the others. The only way to introduce a new file into the system is for a server to use (and thus provide) more space on its local system. This new file will migrate to other servers by the process of trading.

Publishers assign an expiration date to documents when they are published; servers make a promise to keep their shares of a given document until its expiration date is reached. To encourage honest behavior, some servers check whether other servers "drop" data early and decrease their trust of such servers. This trust is monitored and updated by use of a reputation system. Each server maintains a database containing its perceived reputation of the other servers.

Storage

When an author (call her Alice) wishes to store a new document in Free Haven, she must first identify a Free Haven server that's willing to store the document for her. Alice might do this by running a server herself. Alternatively, some servers might have public interfaces or publicly available reply blocks and be willing to publish data for others.

Publication

To introduce a file f into the servnet, the publishing server first splits it into shares. Like the Publius algorithm described in Chapter 11, *Publius*, we use an algorithm that creates a large number (n) of shares but allows the complete

* David Mazieres and M. Frans Kaashoek (1998), "The Design and Operation of an E-mail Pseudonym Server," *5th ACM Conference on Computer and Communications Security*.

Microsoft Exhibit 1019

document to be recreated using a smaller number (k) of those shares. We use Rabin's information dispersal algorithm (IDA)* to break the file into shares $f_1..f_n$. (For any integer i, the notation f_i indicates share i of document f.)

The server then generates a key pair (PK_{doc},SK_{doc}), constructs and signs a data segment for each share, and inserts these shares into its local server space. Attributes in each share include a timestamp, expiration information, hash(PK_{doc}) (a message digest or hash of the public key from the key pair†), information about share numbering, and the signature itself.

The robustness parameter k should be chosen based on some compromise between the importance of the file and the size and available space. A large value of k relative to n makes the file more brittle, because it will be unrecoverable after a few shares are lost. On the other hand, a smaller value of k implies a larger share size, since more data is stored in each share.

We maintain a content-neutral policy toward documents in the Free Haven system. That is, each server agrees to store data for the other servers without regard for the legal or moral issues for that data in any given jurisdiction. For more discussion of the significant moral and legal issues that anonymous systems raise, see the first author's master's degree thesis.‡

Retrieval

Documents in Free Haven are indexed by the public key PK_{doc} from the key pair that was used to sign the shares of the document. Readers must locate (or be running) a server that performs the document request. The reader generates a key pair (PK_{client},SK_{client}) for this transaction, as well as a one-time remailer reply block. The servnet server broadcasts a request containing a message digest or hash of the document's public key, hash(PK_{doc}), along with the client's public key, PK_{client}, and the reply block. This request goes to all the other servers that the initial server knows about. These broadcasts can be queued and then sent out in bulk to conserve bandwidth.

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* Michael O. Rabin (1989), "Efficient Dispersal of Information for Security, Load Balancing, and Fault Tolerance," *Journal of the ACM*, vol. 36, no. 2, pp. 335–348.

† Chapter 15 describes the purpose of message digests. Briefly, the digest of any data item can be used to prove that the data item has not been modified. However, no one can regenerate the data item from the digest, so the data item itself remains private to its owner.

‡ Roger Dingledine (2000), *The Free Haven Project*, MIT master's degree thesis, *http://freehaven.net/papers.html*.

Microsoft Exhibit 1019

Each server that receives the query checks to see if it has any shares with the requested hash of PK_{doc}. If it does, it encrypts each share using the public key PK_{client} enclosed in the request and then sends the encrypted share through the remailer to the enclosed address. These shares will magically arrive out of the ether at their destination; once enough shares arrive (k or more), the client recreates the file and is done.

Share expiration

Each share includes an expiration date chosen at share creation time. This is an absolute (as opposed to relative) timestamp indicating the time after which the hosting server may delete the share with no ill consequences. Expiration dates should be chosen based on how long the publisher wants the data to last; the publisher has to consider the file size and likelihood of finding a server willing to make the trade.

By allowing the publisher of the document to set its expiration time, Free Haven distinguishes itself from related works such as Freenet and Mojo Nation that favor frequently requested documents. We think this is the most useful approach to a persistent, anonymous data storage service. For example, Yugoslav phone books are currently being collected "to document residency for the close to one million people forced to evacuate Kosovo";* those phone books might not have survived a popularity contest. The Free Haven system is designed to provide privacy for its users. Rather than being a publication system aimed at convenience like Freenet, it is designed to be a private, low-profile storage system.

Document revocation

Some publishing systems, notably Publius, allow for documents to be "unpublished" or *revoked*. Revocation has some benefits. It allows the implementation of a read/write filesystem, and published documents can be updated as newer versions became available.

Revocation could be implemented by allowing the author to come up with a random private value x and then publishing a hash of it inside each share. To revoke the document, the author could broadcast his original value x to all servers as a signal to delete the document.

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* University of Michigan News and Information Services, "Yugoslav Phone Books: Perhaps the Last Record of a People," *http://www.umich.edu/~newsinfo/Releases/2000/Jan00/r012000e.html*.

Microsoft Exhibit 1019

On the other hand, revocation allows new attacks on the system. Firstly, it complicates accountability. Revocation requests may not reach all shares of a file, due either to a poor communication channel or to a malicious adversary who sends unpublishing requests only to some members of the servnet. Secondly, authors might use the same hash for new shares and thus "link" documents. Adversaries might do the same to make it appear that the same author published two unrelated documents. Thirdly, the presence of the hash in a share assigns "ownership" to a share that is not present otherwise. An author who remembers his x has evidence that he was associated with that share, thus leaving open the possibility that such evidence could be discovered and used against him later.

One of the most serious arguments against revocation was raised by Ross Anderson.* If the capability to revoke exists, an adversary has incentive to find who controls this capability and threaten or torture him until he revokes the document.

We could address this problem by making revocation optional: the share itself could make it clear whether that share can be unpublished. If no unpublishing tag is present, there would be no reason to track down the author. (This solution is used in Publius.) But this too is subject to attack: If an adversary wishes to create a pretext to hunt down the publisher of a document, he can republish the document *with* a revocation tag and use that as "reasonable cause" to target the suspected publisher.

Because the ability to revoke shares may put the original publisher in increased physical danger, as well as allow new attacks on the system, we chose to leave revocation out of the current design.

Trading

In the Free Haven design, servers periodically trade shares with each other. There are a number of reasons why servers trade:

To provide a cover for publishing
: If trades are common, there is no reason to assume that somebody offering a trade is the publisher of a share. Publisher-anonymity is enhanced.

To let servers join and leave
: Trading allows servers to exit the servnet gracefully by trading for short-lived shares and then waiting for them to expire. This support for a dynamic network is crucial, since many of the participants in Free Haven

* Ross Anderson, "The Eternity Service," *http://www.cl.cam.ac.uk/users/rja14/eternity/eternity.html*.

Microsoft Exhibit 1019

will be well-behaved but transient relative to the duration of the longer-lived shares.

To permit longer expiration dates

Long-lasting shares would be rare if trading them involved finding a server that promised to be available for the next several years.

To accommodate ethical concerns of server operators

Frequent trading makes it easy and unsuspicious for server operators to trade away a particular piece of data with which they do not wish to be associated. If the Catholic Church distributes a list of discouraged documents, server operators can use the hash of the public key in each share to determine if that document is in the list and then trade away the share without compromising either their reputation as a server or the availability of the document. In a non-dynamic environment, the server would suffer a reputation hit if it chose not to keep the document. While we do not currently offer this functionality, trading allows this flexibility if we need it down the road. In particular, the idea of servers getting an "ISP exemption" for documents they hold currently seems very dubious.

To provide a moving target

Encouraging shares to move from server to server through the servnet means that there is never any specific, static target to attack.

The frequency of trading should be a parameter set by the server operator. When server *Alice* wants to make a trade, it chooses another server, *Bob* from its list of known servers (based on reputation) and offers a share x and a request for size or duration of a return share. If *Bob* is interested, it responds with a share y of its own.

Trades are considered "fair" based on the two-dimensional currency of *size* × *duration*. That is, the bigger the size and the longer the document is to be held, the more expensive the trade becomes. The price is adjusted based on the preferences of the servers involved in the trade.

The negotiation is finalized by each server sending an acknowledgment of the trade (including a receipt, as described in the next section, "Receipts") to the other. In addition, each server sends a receipt to both the buddy of the share it is sending and the buddy of the share it is receiving; buddies and the accountability they provide are described later in the section "Accountability and the buddy system." Thus, the entire trading handshake takes four rounds: the first two to exchange the shares themselves, and the next two to exchange receipts while at the same time sending receipts to the buddies.

Microsoft Exhibit 1019

By providing the receipt on the third round of the trading handshake, *Alice* makes a commitment to store the share *y*. Similarly, the receipt that *Bob* generates on the fourth round represents a commitment to store the share *x*. *Bob* could cheat *Alice* by failing to continue the protocol after the third step; in this case, *Alice* has committed to keeping the share from *Bob*, but *Bob* has not committed to anything. At this point, *Alice's* only recourse is to broadcast a complaint against *Bob* and hope that the reputation system causes others to recognize that *Bob* has misbehaved. The alternative is to use a fair exchange protocol, which is unreasonably communications-intensive without a trusted third party.

When *Alice* trades a share to server *Bob*, *Alice* should keep a copy of the share around for a while, just in case *Bob* proves untrustworthy. This will increase the amount of overhead in the system by a factor of two or so (depending on duration), but provides greatly increased robustness. In this case, when a query is done for a share, the system responding should include a flag for whether it believes itself to be the "primary provider" of the data or just happens to have a copy still lying around. The optimum amount of time requires further study.

A diagram describing a trade is given in Figure 12-1. In this diagram, server *Alice* starts out in possession of share $Gita_w$—that is, share *w* of document *Gita*—and server *Bob* starts out in possession of document $Tune_y$. In this case, server *Charlie* has share *x* of document *Gita*, and server *David* has share *z* of document *Tune*. *w* and *x* are buddies, and *y* and *z* are buddies.

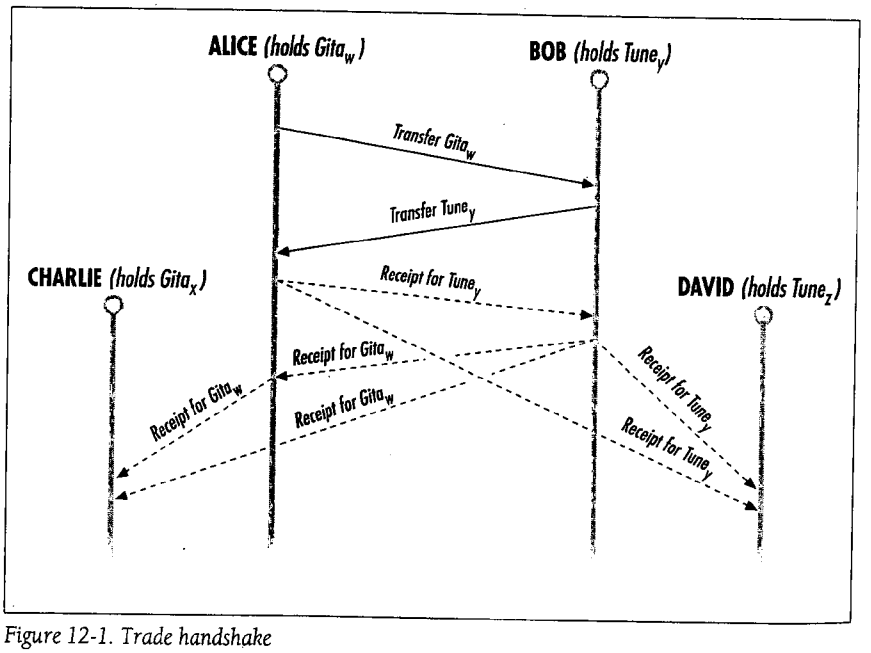

Figure 12-1. Trade handshake

Microsoft Exhibit 1019

Receipts

A receipt contains a hash of the public keys for the source server and the destination server, information about the share traded away, information about the share received, and a timestamp. For each share, it includes a hash of that document's key, which share number it was, its expiration date, and its size.

This entire set of information about the transaction is signed by server A. If B (or any other server) has to broadcast a complaint about the way A handled the transaction, furnishing this receipt along with the complaint will provide some rudimentary level of "proof" that B is not fabricating its complaint. Note that the expiration date of both shares is included within the receipt, and the signature makes this value immutable. Thus, other servers observing a receipt can easily tell whether the receipt is still "valid"—that is, they can check to see whether the share is still supposed to be kept on A. The size of each share is also included, so other servers can make an informed decision about how influential this transaction should be on their perceived reputation of the two servers involved in the trade.

We really aren't treating the receipt as proof of a transaction, but rather as proof of *half* of a transaction—an indication of a commitment to keep a given share safe. This is because the trading protocol is not bulletproof: The fact that Alice has a receipt from Bob could mean that they performed a transaction, or it could mean that they performed 3 out of the 4 steps of the transaction, and then Alice cheated Bob and never gave him a receipt. Thus, the most a given server can do when it detects a misbehaving server is broadcast a complaint and hope the reputation system handles it correctly.

Accountability and the buddy system

Malicious servers can accept document shares and then fail to store them. If enough shares are lost, the document is unrecoverable. Malicious servers can continue their malicious behavior unless there are mechanisms in place for identifying and excising them.

We've designed a *buddy system* that creates an association between two shares within a given document. Each share is responsible for maintaining information about the location of the other share, or *buddy*. When a share moves, it notifies its buddy,* as described in the earlier section "Trading."

* More precisely, it notifies both the server it's moving from and the server it's moving to.

Microsoft Exhibit 1019

Periodically, a server holding a given share should query for its buddy, to make sure its buddy is still alive. Should the server that is supposed to contain its buddy stop responding, the server with the share making the query is responsible for reporting an anomaly. This server announces which server had responsibility for the missing share when it disappeared. The results of this announcement are described later in this chapter in the section "Reputation system."

We considered allowing abandoned shares to optionally spawn a new share if their buddies disappear, but we discarded this notion. Buddy spawning would make the service much more robust, since lost shares could be regenerated. However, such spawning could cause an exponential population explosion of shares for the wrong reasons. If two servers are out of touch for a little while but are not misbehaving or dead, both shares will end up spawning new copies of themselves. This is a strong argument for not letting shares replicate.

When a share x moves to a new machine, there are two "buddy notifications" sent to its buddy x'. But since the communications channel we have chosen currently has significant latency, a notification to x' might arrive after x' has already been traded to a new server. The old server is then responsible for forwarding these buddy notifications to the new server that it believes currently holds x'. Since the old server keeps a receipt as a record of the transaction, it can use this information to remember the new location of x'. The receipt, and thus the forwarding address, is kept by the old server until the share's expiration date has passed.

When a buddy notification comes in, the forwarder is checked and the notification is forwarded if appropriate. This forwarding is *not* done in the case of a document request, since this document request has presumably been broadcast to all servers in the servnet.

We have attempted to distinguish between the design goals of robustness and accountability. The system is quite robust, because a document cannot be lost until a high threshold of its shares has been lost. Accountability, in turn, is provided by the buddy checking and notification system among shares, which protects against malicious or otherwise ill-behaving servers. Designers can choose the desired levels of robustness and accountability independently.

Communications channel

The Free Haven design requires a means of anonymously passing information between agents. One such means is the remailer network, including the Mixmaster remailers first designed by Lance Cottrell. This system is described in fairly nontechnical terminology in Chapter 7, *Mixmaster Remailers*.

Microsoft Exhibit 1019

Other examples of anonymous communication channels are Onion Routing* and Zero-Knowledge Systems' Freedom.† David Martin's doctoral thesis offers a comprehensive overview of anonymous channels in theory and practice.‡

The first implementation of the Free Haven design will use the Cypherpunks and Mixmaster remailers as its anonymous channel.

Reputation system

The reputation system in Free Haven is responsible for creating accountability. Accountability in a system so committed to anonymity is a difficult task. There are many opportunities to try to take advantage of other servers, such as neglecting to send a receipt after a trade or wrongly accusing another server of losing a share. Some of the attacks are quite insidious and complex. The history and issues to consider when developing a reputation system can be found in much more detail in Chapter 16.

Careful trust management should enable each server to keep track of which servers it trusts. Given the large number of shares into which documents are divided—and the relatively few shares required to reconstitute a document—no document should be irretrievably lost unless an astoundingly large number of the servers prove evil.

Each server needs to keep two values that describe each other server it knows about: reputation and credibility. Reputation signifies a belief that the server in question will obey the Free Haven Protocol. Credibility represents a belief that the utterances of that server are valuable information. For each of these two values, each server also needs to maintain a confidence rating. This represents the "stiffness" of the reputation and credibility values.

Servers should broadcast referrals in several circumstances, such as when they log the honest completion of a trade, when they suspect that a buddy of a share they hold has been lost, and when the reputation or credibility values for a server change substantially.

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* P.F. Syverson, D.M. Goldschlag, and M.G. Reed (1997), "Anonymous Connections and Onion Routing," *Proceedings of the 1997 IEEE Symposium on Security and Privacy*.

† Ian Goldberg and Adam Shostack (1999), *Freedom Network 1.0 Architecture*.

‡ David Michael Martin (2000), "Network Anonymity," Boston University Ph.D. thesis, *http://www.cs.du.edu/~dm/anon.html*.

Microsoft Exhibit 1019

Introducers

Document request operations are done via broadcast. Each server wants to store its documents on a lot of servers, and if it finds a misbehaving server it wants to complain to as many as possible. But how do Free Haven servers discover each other?

The reputation system provides an easy method of adding new servers and removing inactive ones. Servers that have already established a good reputation act as *introducers*. New servers can contact these introducers via the anonymous communication channel; the introducers will then broadcast referrals of this new server. This broadcast by itself does not imply an endorsement of the new server's honesty or performance; it is simply an indication that the new server is interested in performing some trades to increase its reputation. Likewise, a server may mark another as "dormant" given some threshold of unanswered requests. Dormant servers are not included in broadcasts or trade requests. If a dormant server starts initiating requests again, the other servers conclude it is not actually dormant and resume sending broadcasts and offering trades to this server.

Implementation status

The Free Haven Project is still in its design stages. Although we have a basic proof-of-concept implementation, we still wish to firm up our design, primarily in the areas of accountability and bandwidth overhead. Before deploying any implementation, we want to convince ourselves that the Free Haven system offers better anonymity than current systems. Still, the design is sufficiently simple and modular, allowing both a straightforward basic implementation and easy extensibility.

Attacks on Free Haven

Anonymous publishing and storage systems will have adversaries. The attacks and pressures that these adversaries employ might be technical, legal, political, or social in nature. The system's design and the nature of anonymity it provides also affect the success of nontechnical attacks.

We now consider possible attacks on the Free Haven system based on their respective targets: the availability of documents and servnet operation, the accountability offered by the reputation system, and the various aspects of anonymity relevant to anonymous storage and publication, as described in the ear-

Microsoft Exhibit 1019

lier section "Anonymity for anonymous storage." For a more in-depth consideration of attacks, we refer to Dingledine's thesis.*

This list of attacks is not complete. In particular, we do not have a systematic discussion of what *kinds* of adversaries we expect. Such a discussion would begin with the most powerful adversaries possible, asking questions like, "What if the adversary controls all but one of the servers in the servnet?" and scaling back from there. In analyzing systems like Free Haven, it is not enough to look at the everyday, plausible scenarios—every effort must be made to provide security against adversaries more powerful than the designers ever expect, because in real life, adversaries have a way of being more powerful than anyone ever expects.

Attacks on documents or the servnet

We've considered a wide variety of ways for adversaries to stop Free Haven or make it less effective, and some ways that we might prevent such attacks:

Physical attack

Destroy a server.

Prevention: Because we are breaking documents into shares, and only k of n shares are required to reconstruct the document, an adversary must find and destroy many servers before availability is compromised.

Legal action

Find a physical server and prosecute the owner based on its contents.

Prevention: Because of the passive-server document-anonymity property that the Free Haven design provides, the servnet operator may be able to plausibly deny knowledge of the data stored on his computer. This depends on the laws of the country in question.

Social pressure

Bring various forms of social pressure against server administrators. Claim that the design is patented or otherwise illegal. Sue the Free Haven Project and any known server administrators. Conspire to make a cause "unpopular," convincing administrators that they should manually prune their data. Allege that they "aid child pornographers" and other socially unacceptable activities.

Prevention: We rely on the notion of jurisdictional arbitrage. Information illegal in one place is frequently legal in others. Free Haven's content-neutral policies mean that there is no reason to expect that the server operator

* Dingledine, *op. cit.*

Microsoft Exhibit 1019

has looked at the data she holds, which might make it more difficult to prosecute. We further rely on having enough servers in enough different jurisdictions that organizations cannot conspire to bully a sufficient fraction of servers to make Free Haven unusable.

Denial of service

Attack the servnet by continued flooding of queries for data or requests to join the servnet. These queries may use up all available bandwidth and processing power for a server.

Prevention: We must assume that our communications channel has adequate protection and buffering against this attack, such as the use of client puzzles or other protections described in Chapter 16. Most communications channels we are likely to choose will not protect against this attack. This is a real problem.

Data flooding

Attempt to flood the servnet with shares, to use up available resources.

Prevention: The trading protocol implicitly protects against this type of denial of service attack against storage resources. The ability to insert shares, whether "false" or valid, is restricted to trading: that server must find another that trusts its ability to provide space for the share it would receive in return.

Similarly, the design provides protection against the corrupting of shares. Altering (or "spoofing") a share cannot be done, because the share contains a particular public key and is signed by the corresponding private key. Without knowledge of the original key that was used to create a set of shares, an adversary cannot forge new shares for a given document.

Share hoarding

Trade until a sufficient fraction of an objectionable document is controlled by a group of collaborating servers, and then destroy this document. Likewise, a sufficiently wealthy adversary could purchase a series of servers with very large drives and join the servnet, trading away garbage for "valuable data." He can trade away enough garbage to have a significant portion of all the data in the servnet on his drives, subject to deletion.

Prevention: We rely on the overall size of the servnet to make it unlikely or prohibitively expensive for any given server or group of collaborating servers to obtain a sufficient fraction of the shares of any given document. The failure of this assumption would leave us with no real defense.

Microsoft Exhibit 1019

Attacks on the reputation system

While attacks against the reputation system* are related to attacks directly against servers, their goal is not to directly affect document availability or servnet operation. Rather, these attacks seek to compromise the means by which we provide accountability for malicious or otherwise misbehaving servers.

Some of these attacks, such as temporary denials of service, have negative repercussions on the reputation of a server. These repercussions might be qualified as "unfair," but are best considered in the following light: if a server is vulnerable to these attacks, it may not be capable of meeting the specifications of the Free Haven Protocol. Such a server is not worthy of trust to meet those specifications. The reputation system does not judge intent, merely actions. Following are some possible attacks on the reputation system, and ways that we might prevent such attacks:

Simple betrayal

An adversary may become part of the servnet, act correctly long enough to gain a good reputation, and then betray this trust by deleting files before their expiration dates.

Prevention: The reputation economy is designed to make this unprofitable. In order to obtain enough "currency" to store data, a server must reliably store data for others. Because a corrupt server must store at least as much data for others as the amount of data it deletes, such an adversary at worst does no overall harm to the system and may even help.

A server that engages in this behavior should be caught by the buddy system when it deletes each share.

Buddy coopting

If a corrupt server (or group of colluding servers) can gain control of both a share and its buddy, it can delete both of them without repercussions.

Prevention: We assume a large quantity of shares in the servnet, making buddy capture more difficult. Servers also can modify reputation ratings if precise trading parameters, or constant trading, suggests an attempt to capture buddies. More concretely, a possible work-around involves separating the reply-block addresses for trading and for buddy checking, preventing corrupt servers from acquiring the buddies of the shares they already have. Such an approach adds complexity and possibly opens other avenues for attack.

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* Parts of this section were originally written by Brian T. Sniffen in "Trust Economies in the Free Haven Project," May 2000, *http://theory.lcs.mit.edu/~cis/cis-theses.html*.

Microsoft Exhibit 1019

False referrals

An adversary can broadcast false referrals, or even send them only to selected servers.

Prevention: The confidence rating of credibility can provide a guard against false referrals, combined with a single-reporting policy (i.e., at most one referral per target per source is used for reputation calculations).

Trading receipt games

While we believe that the signed timestamps attest to who did what and when, receipt-based accountability may be vulnerable to some attacks. Most likely, these will involve multiserver adversaries engaging in coordinated bait-and-switch games with target servers.

Entrapment

There are several ways in which an adversary can appear to violate the protocols. When another server points them out, the adversary can present receipts that show her wrong and can accuse her of sending false referrals. A more thorough system of attestations and protests is necessary to defend against and account for this type of attack.

Attacks on anonymity

There are a number of attacks that might be used to determine more information about the identity of some entity in the system:

Attacks on reader-anonymity

An adversary might develop and publish on Free Haven a customized virus that automatically contacts a given host upon execution. A special case of this attack would be to include mime-encoded URLs in a document to exploit reader software that automatically loads URLs. Another approach might be to become a server on both the servnet and the mix net and attempt an end-to-end attack, such as correlating message timing with document requests. Indeed, servers could claim to have a document and see who requests it, or simply monitor queries and record the source of each query. Sophisticated servers might attempt to correlate readers based on the material they download and then try to build statistical profiles and match them to people (outside Free Haven) based on activity and preferences. We prevent this attack by using each reply block for only one transaction.

Attacks on server-anonymity

Adversaries might create unusually large shares and try to reduce the set of known servers that might have the capacity to store such shares. This attacks the partial anonymity of these servers. An adversary could become a server and then collect routine status and participation information (such as

Microsoft Exhibit 1019

server lists) from other servers. This information might be combined with extensive knowledge of the bandwidth characteristics and limitations of the Internet to map servnet topology. By joining the mix net, an adversary might correlate message timing with trade requests or reputation broadcasts. An alternate approach is simply to spread a Trojan Horse or worm that looks for Free Haven servers and reports which shares they are currently storing.

Attacks on publisher-anonymity

An adversary could become a server and log publishing acts, and then attempt to correlate source or timing. Alternatively, he might look at servers that recently have published a document and try to determine who has been communicating with them recently.

There are also entirely social attacks that can be very successful, such as offering a large sum of money for information leading to the current location of a given document, server, reader, etc.

We avoid or reduce the threat of many of these attacks by using an anonymous channel that supports pseudonyms for our communications. This prevents most or all adversaries from being able to determine the source or destination of a given message or establish linkability between each endpoint of a set of messages. Even if server administrators are subpoenaed or otherwise pressured to release information about these entities, they can openly disclaim any knowledge.

An analysis of anonymity

We describe the protections offered for each of the broad categories of anonymity. In Table 12-1, we provide an overview of Free Haven and the different publishing systems that we examined. We consider the level of privacy provided—computational (C) and perfect-forward (P-F) anonymity—by the various systems.

Table 12-1. Anonymity properties of publishing systems

Project	Publisher		Reader		Server		Document	Query
	C	P-F	C	P-F	C	P-F	C	C
Gnutella								
Eternity Usenet	✓	✓	?				✓	
Freenet	✓	✓	?				✓	
Mojo Nation	?	?					✓	
Publius	✓	✓					✓	
Free Haven	✓	✓	✓	✓	✓		✓	

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Computational anonymity means that an adversary with "reasonable" computing resources and knowledge is unable to break the anonymity involved. The adversary may do anything it likes to try to break the system but is limited in how much power it has; for example, it may not be able to break the cryptography involved in building a system or be able to break into the computers of every single machine running the system.

Perfect-forward anonymity is analogous to perfect-forward secrecy: A system is perfect-forward anonymous if no information remains after a transaction is completed that could later identify the participants if one side or the other is compromised. This notion is a little bit trickier—think of it from the perspective of an adversary watching the system over a long period of time. Is there anything that the adversary can discover from watching several transactions that he can't discover from watching a single transaction?

Free Haven provides computational and perfect-forward author-anonymity, because authors communicate with publishers via an anonymous channel. Servers trade with other servers via pseudonyms, providing computational but not perfect-forward anonymity, as the pseudonyms can be broken later. Because trading is constant, however, Free Haven achieves publisher-anonymity for publishers trying to trade away all shares of the same document. The use of IDA to split documents provides passive-server document-anonymity, but the public key embedded in each share (which we require for authenticating buddy messages) makes it trivial for active servers to discover what they are storing. Because requests are broadcast via an anonymous channel, Free Haven provides computational reader-anonymity, and different reply blocks used and then destroyed after each request provide perfect-forward reader-anonymity.

Gnutella fails to provide publisher-anonymity, reader-anonymity, or server-anonymity because of the direct connections for actual file transfer. Because Gnutella servers start out knowing the intended contents of each document they are offering, they also fail to provide document-anonymity.

Eternity Usenet provides publisher-anonymity via the use of one-way anonymous remailers. Server-anonymity is not provided, because every Usenet server that carries the Eternity newsgroup is a server. Anonymity expert Adam Back, designer of the Eternity Usenet service, has pointed out that passive-server document-anonymity can be provided by encrypting files with a key derived from the URL; active servers might find the key and attempt to decrypt stored documents. Reader-anonymity is not provided by open public proxies unless the reader uses an anonymous channel, because the proxy can see what a user queries or downloads, and at what time. For local proxies, which connect to a sepa-

Microsoft Exhibit 1019

rate news server, however, the situation is better because the news server knows only what the user downloads. Even so, this is not quite satisfactory, because the user can be tied by the server to the contents of the Eternity newsgroup at a certain time.

Freenet achieves passive-server document-anonymity because servers are unable to reverse the hash of the document name to determine the key with which to decrypt the document. For active-server document-anonymity, the servers can check whether they are carrying a particular key but cannot easily match a stored document to a key due to the hash function. Server-anonymity is not provided because, given a document key, it is very easy to locate a server that is carrying that document—querying any server at all will result in that server carrying the document! Because of the TTL and Hops fields for both reading and publishing, it is also not clear that Freenet achieves publisher- or reader-anonymity, although it is much better in these regards than Gnutella. We note that the most recent Freenet design introduces randomized TTL and Hops fields in each request, and plans are in the works to allow a Publish or Retrieve operation to traverse a mix net chain before entering the Freenet system. These protections will make attacks based on tracking queries much more difficult.

Mojo Nation achieves document-anonymity, as described earlier, because the server holding a share doesn't know how to reconstruct that document. The Mojo Nation design is amenable to integrating publisher-anonymity down the road—a publisher can increase her anonymity by paying more Mojo and chaining requests through participants that act as "relays." The specifics of prepaying the path through the relays are not currently being designed. It seems possible that this technique could be used to ensure reader-anonymity as well, but the payment issues are even more complex. Indeed, the supplied digital cash model is not even anonymous currently; users need to uncomment a few lines in the source, and this action breaks Chaum's patents.

Publius achieves document-anonymity because the key is split between n servers, and without sufficient shares of the key, a server is unable to decrypt the document that it stores. The secret sharing algorithm provides a stronger form of this anonymity (albeit in a storage-intensive manner), since a passive server really can learn nothing at all about the contents of a document that it is helping to store. Because documents are published to Publius through a one-way anonymous remailer, it provides publisher-anonymity. Publius provides no support for protecting readers by itself, however, and the servers containing a given file are clearly marked in the URL used for retrieving that file. Readers can use a system such as ZKS Freedom or Onion Routing to protect themselves, but servers may still be liable for storing "bad" data.

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We see that systems can often provide publisher-anonymity via one-way communication channels, effectively removing any linkability; removing the need for a reply pseudonym on the anonymous channel means that there is "nothing to crack." The idea of employing a common mix net as a communication channel for each of these publication systems is very appealing. We could leave most of the anonymity concerns to the communication channel itself and provide a simple backend filesystem or equivalent service to transfer documents between agents. Thus the design of the backend system could be based primarily on addressing other issues such as availability of documents, protections against flooding and denial of service attacks, and accountability in the face of this anonymity.

Future work

Our experience designing Free Haven revealed several problems that have no simple solutions; further research is surely required. We state some of these problems here and refer to Dingledine's thesis* for in-depth consideration:

Deployed free low-latency pseudonymous channel

Free Haven requires pseudonyms in order to create server reputations. The only current widely deployed channels that support pseudonyms seem to be the Cypherpunk remailer network and ZKS Freedom mail. The Cypherpunk and ZKS Version 1 networks run over SMTP and consequently have high latency. This high latency complicates protocol design. The recently announced Version 2 of ZKS Freedom mail runs over POP and may offer more opportunity for the kind of channel we desire.

Modelling and metrics

When designing Free Haven, we made some choices, such as the choice to include trading, based on only our intuition of what would make a robust, anonymous system. A mathematical model of anonymous storage would allow us to test this intuition and run simulations. We also need *metrics*: Specific quantities that can be measured and compared to determine which designs are better. For example, we might ask, "How many servers must be compromised by an adversary for how long before any document's availability is compromised? Before a specific targeted document's availability is compromised?" or, "How many servers must be compromised by an adversary for how long before the adversary can link a document and a publisher?" This modelling could draw from a wide variety of previous work.

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* Dingledine, *op. cit.*

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Formal definition of anonymity

Closely related to the last point is the need to formalize the kinds of anonymity presented in the section "Anonymity for anonymous storage." By formally defining anonymity, we can move closer to providing meaningful *proofs* that a particular system provides the anonymity we desire. We might leverage our experience with cryptographic definitions of semantic security and nonmalleability to produce similar definitions and proofs.* A first step in this direction might be to carefully explore the connection remarked upon by Simon and Rackoff between secure multiparty computation and anonymous protocols.†

Usability requirements and interface

We stated in the introduction that we began the Free Haven Project out of concern for the rights of political dissidents. Unfortunately, at this stage of the project, we have contacted few political dissidents and, as a consequence, do not have a clear idea of the usability and interface requirements for an anonymous storage system. Our concern is heightened by a recent paper which points out serious deficiencies in PGP's user interface.‡

Efficiency

It seems like nearly everyone is doing a peer-to-peer system or WWW replacement these days. Which one will win? Adam Back pointed out that in many cases, the efficiency and perceived benefit of the system is more important to an end user than its anonymity properties. This is a major problem with the current Free Haven design: we emphasize a quality relatively few potential users care about at the expense of something nearly everyone cares about. Is there a way to create an anonymous system with a tolerable loss of perceived efficiency compared to its non-anonymous counterpart? And what does "tolerable" mean, exactly?

We consider the above to be "challenge problems" for anonymous publication and storage systems.

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* Oded Goldreich (1999). *Modern Cryptography, Probabilistic Proofs, and Pseudo-Randomness.* Springer-Verlag.

† Simon and Rackoff (1993), "Cryptographic Defense Against Traffic Analysis," *STOC 1993*, pp. 672–681.

‡ Alma Whitten and J.D. Tygar (1999), "Why Johnny Can't Encrypt," *USENIX Security 1999*, *http://www.usenix.org/publications/library/proceedings/sec99/whitten.html.*

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Conclusion

Free Haven aims to solve a problem that no other system currently addresses—creating a decentralized storage service that at the same time protects the anonymity of publishers, readers, and servers, provides a dynamic network, and ensures the availability of each document for a publisher-specified lifetime. We have made progress in identifying the requirements for each of these goals and designing solutions that meet them.

The current Free Haven design is unfortunately unsuitable for wide deployment, because of several remaining problems. The primary problem is efficiency—unless we can provide a sufficiently friendly and efficient interface to the documents stored in the system, we will find ourselves with very few servers. Indeed, since we need systems that are relatively reliable, we can't make as good use of typical end-user machines as a system like Freenet can. This small number of servers will in turn decrease the amount of robustness that our system offers.

Free Haven uses inefficient broadcasts for communication. A large step to address this problem is coupling Free Haven with a widely deployed efficient file-sharing service such as Freenet or Mojo Nation. Popular files will be highly accessible from within the faster service; Free Haven will answer queries for less popular documents that have expired in this service.

Free Haven sets out to accomplish several goals not considered *en masse* by other peer-to-peer publishing/storage systems: Flexibility, anonymity for all parties, content-neutral persistence of data, and accountability. These ambitious goals are the root cause of existing design difficulties. Without the requirement of long-term persistent storage, strong accountability measures are not as necessary. Without these measures, computational overhead can be greatly lowered, making unnecessary many communications that are used to manage reputation metrics. And without the requirement for such anonymity and the resulting latency from the communications channel, readers could enjoy much faster document retrieval. Yet, the study and emphasis of these ambitious goals are Free Haven's contribution and importance in a rapidly evolving peer-to-peer digital world.

Acknowledgments

Professor Ronald Rivest provided invaluable assistance reviewing Roger's master's thesis and as Michael's bachelor's thesis advisor and caused us to think hard about our design decisions. Professor Michael Mitzenmacher made possible

Microsoft Exhibit 1019

David's involvement in this project and provided insightful comments on information dispersal and trading. Beyond many suggestions for overall design details, Brian Sniffen provided the background for the reputation system (which we followed up with the discussion of reputation systems in Chapter 16), and Joseph Sokol-Margolis was helpful in considering attacks on the system. Andy Oram, our editor, was instrumental in turning this from an academic paper into something that is actually readable. Adam Back and Theodore Hong commented on our assessment of their systems and made our related work section much better. Furthermore, we thank Susan Born, Nathan Mahn, Jean-François Raymond, Anna Lysyanskaya, Adam Smith, and Brett Woolsridge for further insight and feedback.

Microsoft Exhibit 1019

part three

Technical Topics

In this part, project leaders choose various key topics in order to focus on the problems, purposes, and promises of peer-to-peer technologies.

Microsoft Exhibit 1019

chapter thirteen

Metadata

Rael Dornfest, O'Reilly Network, and
Dan Brickley, ILRT and RDFWeb

Today's Web is a great, big, glorious mess. Spiders, robots, screen-scraping, and plaintext searches are standard practices that indicate a desperate attempt to draw arbitrary distinctions between needles and hay. And they go only so far as the data we've taken the trouble to make available online.

Now peer-to-peer promises to turn your desktop, laptop, palmtop, and fridge into peers, chattering away with one another and making swaths of their data stores available online. Of course, if every single device on the network exposes even a small percentage of the resources it manages, it will exacerbate the problem by piling on more hay and needles in heaps. How will we cope with the sudden logarithmic influx of disparate data sources?

The new protocols being developed at breakneck speed for peer-to-peer applications also add to the mess by disconnecting data from the fairly bounded arena of the Web and the ubiquitous port 80. Loosening the hyperlinks that bind all these various resources together threatens to scatter hay and needles to the winds. Where previously we had application user interfaces for each and every information system, the Web gave us a single user interface—the browser—along with an organizing principle—the hyperlink—that allowed us to reach all the material, at least in theory. Peer-to-peer might undo all this good and throw us back into the dark ages of one application for each application type or application service. We already have Napster for MP3s and work has begun on Docster for documents—can JPEGster and Palmster be very far off?

And how shall we search these disparate, transitory clumps of data, winking in and out of existence as our devices go on and offline, to say nothing of finding the clumps in the first place? Napster is held up as a reassurance that everything can work out on its own. The inherent ubiquity of any one MP3 track gets around the problem of resource transience. However, isn't this abundance

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simply the direct result of its rather constrained problem space? MP3 files are popular, and MP3 rippers make it easy for huge numbers of people to create decent-quality files. As industry attention turns to peer-to-peer technologies, and as the content within these systems becomes more heterogeneous, the technology will have to accommodate content that is harder to accumulate and less popular; the critical mass of replicated files will not be attained. Familiar problems associated with finding a particular item may reemerge, this time in a decentralized environment rather than around the familiar Web hub.

Whether or not peer-to-peer fares any better than the Web, it certainly presents a new challenge for people concerned with describing and classifying information resources. Peer-to-peer provides a rich environment and a promising early stage for putting in place all we've learned about metadata over the past decade.

So, before we go much further, what exactly *is* metadata?

Data about data

Metadata is the stuff of card catalogues, television guides, Rolodexes, taxonomies, tables of contents—to borrow a Zen concept, the finger pointing at the moon. It is labels like "title," "author," "type," "height," and "language" used to describe a book, person, television program, species, etc. Metadata is, quite simply, data about data.

There are communities of specialists who have spent years working on—and indeed solving some of—the hard problems of categorizing, cataloguing, and making it possible to find things. Even in the early days of the Web, developers enlisted the help of these information scientists and architects, realizing that otherwise we'd be in for quite a mess. The Dublin Core Metadata Initiative (DMCI)* is just such an effort. An interdisciplinary, international group founded in 1994, the DCMI's charter is to use a minimal set of metadata constructs to make it easier to find things on the Web. We'll take a closer look at Dublin Core in a moment.

Yet, while well-understood systems exist for cataloguing and classifying some classic types of information, such as books (e.g., MARC records and the Dewey Decimal System), equivalent facilities were late to arrive on the Web—some

* Dublin Core Metadata Initiative, *http://www.dublincore.org*; "Metadata With a Mission: Dublin Core", *http://www.xml.com/pub/2000/10/25*; Dublin Core Metadata Element Set, Version 1.1, *http://purl.org/dc/elements/1.1*.

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would say far too late. They are emerging, however, just in time for peer-to-peer.

Metadata lessons from the Web

Peer-to-peer's power lies in its willingness to rethink old assumptions and reinvent the way we do things. This can be quite constructive, even revolutionary, but it also risks being hugely destructive in that we can throw out lessons previously learned from the web experience. In particular, we know that the Web suffered because metadata infrastructure was added relatively late (1997+), an add-on situation that had an impact on various levels.

The Web burst onto the scene before we managed to agree on common descriptive practices—ways of describing "stuff." Consequently, the vast majority of web-related tools lack any common infrastructure for specifying or using the properties of web content. WYSIWYG HTML editors don't go out of their way to make their metadata support (if any) visible, nor do they request metadata for a document when authors press the "Save" button. Search engines provide little room for registering metadata along with their associated sites. Robots and spiders often discard any metadata in the form of HTML `<meta>` tags they might find. This has resulted in an enormous hodgepodge of a data set with little rhyme or reason. The Web is hardly the intricately organized masterpiece represented by its namesake in nature.

Early peer-to-peer applications come from relatively limited spheres (MP3 file-sharing, messaging, Weblogs, groupware, etc.) with pretty well understood semantics and implicit metadata—we know it's an MP3 because it's in Napster. These communities have the opportunity, before heterogeneity and ubiquity muddy the waters, to describe and codify their semantics to allow for better organization, extraction, and search functionality down the road. Yet even at this early stage, we're already seeing the same mistakes creeping in.

Resource description

Until recently, the means available to content providers for describing the resources they make available on the Web have been inconsistent at best. About the only consistent metadata in an HTML document is the `<title>` element, which provides only a hint at best as to the content of the page. HTML's `<meta>` element is supposed to provide a method for embedding arbitrary metadata—but that creates more of a problem than a solution, because applications, books, articles, tutorials, and standards bodies alike express little guidance as to what good metadata should look like and how best to express it.

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The work of the aforementioned Dublin Core offers a wonderful start. The Dublin Core Metadata Element Set is a set of 15 elements (title, description, creator, date, publisher, etc.) that are useful in describing almost any web resource. Rather than attempt to define semantics for specific instances and situations, the DCMI focused on the commonalities found in resources of various shapes and flavors. The Dublin Core may just as easily be used to describe "a journal article in PDF format," "an MPEG encoding of an episode of *Buffy the Vampire Slayer* recorded on a hacked TiVO," or "a healthcare speech given by the U.S. President on March 2, 2000."

Example 13-1 shows a typical appearance of Dublin Core metadata in a fragment of HTML. Each `<meta>` tag contains an element of metadata defined by Dublin Core.

Example 13-1. Dublin Core metadata in an HTML document

```
<html>
  <head>
    <title>Distributed Metadata</title>
    <meta name="description" content="This article addresses...">
    <meta name="subject" content="metadata, rdf, peer-to-peer">
    <meta name="creator" content="Dan Brickley and Rael Dornfest">
    <meta name="publisher" content="O'Reilly & Associates">
    <meta name="date" content="2000-10-29T00:34:00+00:00">
    <meta name="type" content="article">
    <meta name="language" content="en-us">
    <meta name="rights" content="Copyright 2000, O'Reilly & Associates,
Inc.">
    ...
  </head>
  ...
```

While useful up to a point, the original HTML mechanism for embedding metadata has proven limited. There is no built-in convention to control the names given to the various embedded metadata fields. As a consequence, HTML `<meta>` tags can be ambiguous: we don't know which sense of "title" or "date" is being used.

XML represents another evolution in web architecture, and along with XML come namespaces. Example 13-2 illustrates some namespaces in use. Like peer-to-peer, namespaces exemplify decentralization. We can now mix descriptive elements defined by independent communities, without fear of naming clashes, since each piece of data is tied a URI that provides a context and definition for it.

Microsoft Exhibit 1019

Example 13-2. Dublin Core metadata in an XML document

```
<?xml version="1.0" encoding="iso-8859-1"?>

<rdf:RDF
  xmlns:rdf="http://www.w3.org/1999/02/22-rdf-syntax-ns#"
  xmlns:dc="http://purl.org/dc/elements/1.1/"
  xmlns="http://purl.org/rss/1.0/"
>
...
  <item rdf:about="http://www.oreillynet.com/.../metadata.html">
    <title>Distributed Metadata</title>
    <link>http://www.oreillynet.com/.../metadata.html </link>
    <dc:description>This article addresses...</dc:description>
    <dc:subject>metadata, rdf, peer-to-peer </dc:subject>
    <dc:creator>Dan Brickley and Rael Dornfest </dc:creator>
    <dc:publisher>O'Reilly & Associates</dc:publisher>
    <dc:date>2000-10-29T00:34:00+00:00</dc:date>
    <dc:type>article</dc:type>
    <dc:language>en-us</dc:language>
    <dc:format>text/html</dc:format>
    <dc:rights>Copyright 2000, O'Reilly & Associates, Inc.</dc:rights>
    ...
  </item>
  ...
```

In the example above, Dublin Core elements are prepended by the namespace name "dc:". The name is associated with the URI *http://purl.org/dc/elements/1.1* by the "xmlns:dc" construct at the beginning of the document. "dc:subject" is therefore understood to mean "the subject element in the dc namespace as defined at *http://purl.org/dc/elements/1.1*."

Namespaces let each author weave additional semantics required by particular types of resources or appropriate to a specific realm with the more general resource description such as that provided by the Dublin Core. In the book world, an additional definition might be the ISBN or Library of Congress number, while in the music world, it might be some form of compact disc identifier.

Now, we're not insisting that each and every document be described using all 15 Dublin Core elements and along various other lines as well. Something to keep in mind, however, is that every bit of metadata provides a logarithmic increase in available semantics, making resources less ambiguous and easier to find. Peer-to-peer application developers may then use the descriptions provided by a resource rather than having to resort to guesswork or such extremes as sequestering resources of a certain type to their own network.

Microsoft Exhibit 1019

Searching

Searching is the bane of the Web's existence, despite the plethora of search tools—Yahoo currently lists 193 registered web search engines.* Search engines typically suffer from a lack of semantics on both the gathering and querying ends. On the gathering side, search engines typically utilize one of two methods:

1. Internet directories typically ask content providers to register their web sites through an online form. Unfortunately, such forms don't provide slots for metadata such as publisher, author, subject keywords, etc.
2. Search engines scour the Web with armies of agents/spiders, scraping pages and following links for hints at semantics. Sadly, even if a site does embed metadata (such as HTML's `<meta>` tags) in its documents, this information is often ignored.

On the querying end, while some sites do make an attempt to narrow the context for particular word searches (using such categories as "all the words," "any of the words," or "in the title"), successful searching still comes down to keywords and best guess. It's virtually impossible to disambiguate between concepts like "by" and "about"—"find me all articles written *by* Andy Oram" versus "find me anything *about* Andy Oram." Queries like "find me anything on Perl written by the person whose email address is *larry@wall.org*" are out of the question.

While the needs of users clearly call for semantically rich queries, some peer-to-peer applications and systems are doing little to provide even the simplest of keyword searches. The categories "artist" and "title," which may be enough within Napster, will fold up and collapse in more heterogeneous peer-to-peer environments populated by MP3s, documents, images, and the various other data types found on the Web today. While Freenet does provide the boon of an optional accompanying metadata file to accompany any resource added to the cloud, this is currently of minimal use for a couple of reasons: a) No guidance exists on what this metadata file should contain, and b) There is currently no search functionality. Gnutella's InfraSearch allows for a wonderfully diverse interpretation and subsequent processing of search terms: While a dictionary node sees "country" as a term to be looked up, an MP3 node may see it as a music genre. Unfortunately, however, the InfraSearch user interface still provides only a simple text entry field and little chance for the user to be an active participant in defining the parameters of his or her search.

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* Yahoo!'s "Search Engines" category, *http://dir.yahoo.com/computers_and_internet/internet/world_wide_web/searching_the_web/search_engines*.

Microsoft Exhibit 1019

Hopefully we'll see peer-to-peer applications emerging that empower both the content provider and end user by providing semantically rich environments for the description and subsequent retrieval of content. This should be reflected both in the user interface and in the engine itself.

Resources and relationships: A historical overview

So where does this all leave us? How do we infuse our peer-to-peer applications with the metadata lessons learned from the Web?

The core of the World Wide Web Consortium's (W3C) metadata vision is a concept known as the *Semantic Web*. This is not a separate Web from the one we currently weave and wander, but a layer of metadata providing richer relationships between the ostensibly disparate resources we visit with our mouse clicks. While HTML's hyperlinks are simple linear paths lacking any obvious meaning, such semantics do exist and need only a means of expression.

Enter the *Resource Description Framework* (RDF),* a data model and XML serialization syntax for describing resources both on and off the Web. RDF turns those flat hyperlinks into arcs, allowing us to label not only the endpoints, but the arc itself—in other words, ascribe meaning to the relationship between the two resources at hand. A simple link between Andy Oram's home page and an article on the O'Reilly Network provides little insight into the relationship between the two. RDF disambiguates the relationship: "Andy wrote this particular article" versus "this is an article about Andy" versus "Andy found this article rather interesting."

RDF's history itself shows how emerging peer-to-peer applications can benefit from a generalized and consistent metadata framework. RDF has roots in an earlier effort, the *Platform for Internet Content Selection*, or PICS. One of the original goals for PICS was to facilitate a wide range of rating and filtering services, particularly in the areas of child protection and filtering of pornographic content. It defined a simple metadata "label" format that could encode a variety of classification and rating vocabularies (e.g., RSACi, MedPICS†). It included the goal of allowing diverse communities to create their own content rating languages and networked metadata services for distributing these descriptive labels. While

* Resource Description Framework, *http://www.w3.org/RDF*.

† Links to PICS vocabularies and W3C specifications, *http://www.w3.org/PICS*; "Metadata, PICS and Quality" (1997), *http://www.ariadne.ac.uk/issue9/pics*.

Microsoft Exhibit 1019

originally it defined a pretty comprehensive set of tools for rating and filtering systems, PICS as initially defined did not play well with other metadata applications. The protocols, data formats, and accompanying infrastructure were too tightly coupled to one narrow application—it wasn't general enough to be useful for everyone.

One critical piece PICS lacked was a namespaces mechanism that would allow a single PICS label to draw upon multiple, independently managed vocabularies. The designers of PICS eventually realized that all the work they had put into a well-designed query protocol, a digital signatures system, vocabularies, and so forth risked being reinvented for various other, non-PICS-specific metadata applications.

The threat of such duplication led to the invention of RDF. Unlike PICS, RDF has a highly general information model designed from the ground up to allow diverse applications to create data that can be easily intermingled. However diverse, RDF applications all share a common strategy: they talk about unambiguously named properties of unambiguously named resources. To eliminate ambiguous interpretations of properties such as "type" or "format," RDF rests on unique identifiers.

Foundations of resource description: Unique identifiers

Unique identification is the critical empowering technology for metadata. We benefit from having unique identifiers for both the things we describe (*resources*), and the ways we describe them (*properties*). In RDF, we call the things we're describing resources regardless of whether they're people, places, documents, movies, images, databases, etc. All RDF applications adopt a common convention for identifying these things (regardless of what else they disagree about!).

We identify the things we're describing with Uniform Resource Identifiers, or URIs.* You're most probably familiar with one subset of URIs, the Uniform Resource Locator, or URL. While URLs are concerned with the location and retrieval of resources, URIs more generally are unique identifiers for things that may not necessarily be retrievable.

We also need clarity concerning properties, which are how we describe our resources. To say that something is of a particular type, or has a certain relationship to another resource, or has some specified attribute, we need to uniquely

* URI defines a simple text syntax for URLs, URNs and similar controlled names for use on the Internet, *http://www.w3.org/Addressing*.

Microsoft Exhibit 1019

identify our descriptive concepts. RDF uses URIs for these too. Different communities can invent new descriptive properties (such as person, employee, price, and classification) and assign URIs to these properties.

Since the assignment of URIs is decentralized, we can be sure that uniquely named descriptive properties don't get mixed up when we integrate metadata from multiple sources. An auto-maker's concept of "type" is different from that of a cheese-maker's. The use of URIs such as *http://webuildcars.org/descriptions/types* and *http://weagecheese.org/descriptions/type* serves to uniquely identify the particular "type" we're using to describe a resource.

One critical lesson we can take away from the PICS story is that, when it comes to metadata, it is very hard to partition the problem space. The things we want to describe, the things we want to say about them, and the things we want to do with this data are all deeply entangled. RDF is an attempt to provide a generalized framework for all types of metadata. By providing a consistent abstraction layer that goes below surface differences, we gain an elegant core architecture on which to build. There is no limit to the material or applications RDF supports: through different URIs and namespaces, different groups can extend the common RDF model to describe the needs of the peer-to-peer application at hand. No standards committee or centralized initiative gets to decide how we describe things. Applications can draw upon multiple descriptive vocabularies in a consistent, principled manner. The combination of these two attributes—consistent framework and decentralized descriptive concepts—is a powerful architecture for the peer-to-peer applications being built today.

When it comes to metadata, the network becomes a poorer information resource whenever we create artificial boundaries between metadata applications. The Web's own metadata system, RDF, was built in acknowledgment of this. There is little reason to suppose peer-to-peer content is different in this regard since we're talking about pretty much the same kind of content, albeit in a radically new environment.

A contrasting evolution: MP3 and the metadata marketplace

The alternatives to erecting a rigorous metadata architecture like RDF can be illustrated by the most popular decentralized activity on the Internet today: MP3 file exchange.

How do people find out the names of songs on the CDs they're playing on their networked PCs? One immediate problem is that there is nothing resembling a URI scheme for naming CDs; this makes it difficult to agree on a protocol for

Microsoft Exhibit 1019

querying metadata servers about the properties of those CDs. While one might imagine taking one of the various CDDB-like algorithms and proposing a URI scheme for universal adoption (for instance, cd:8941207208781920 91), in practice this would be time-consuming and somewhat politicized. Meanwhile, peer-to-peer developers just want to build killer apps; they don't want to spend 18 months on a standards committee specifying the identifiers for compact discs (or people or films...). Most of us can't afford the time to create metadata tags, and if we could, we'd doubtless think of more interesting ways of using that time.

What to do? Having just stressed the importance of unique names when describing content, can we get by without them? Actually, it appears so.

Every day thousands of MP3 users work around the unique identification problem without realizing it. Their CD rippers inspect the CD, compute one of several identifying properties for the CD they're digitizing, and use this uniquely identifying property to consult a networked metadata service. This is metadata in action on a massive scale. But it also smacks of the PICS problem. MP3 listeners have settled on an application-specific piece of infrastructure rather than a more useful, generalized approach.

These metadata services exist and operate very successfully today, despite the lack of any canonical "standard" identifier syntax for compact discs. The technique they use to work around the standards bottleneck is simple, being much the same as saying things like "the person whose personal mailbox is..." or "the company whose corporate homepage is...". Being simple, it can (and should) be applied in other contexts where peer-to-peer and web applications want to query networked services for metadata. There's no reason to use a different protocol when asking for a CD track list and when asking for metadata describing any other kind of thing.

The basic protocol being used in CD metadata query is both simple and general: "tell me what you know about the resource whose CD checksum is some-huge-number"—a protocol reminiscent of the PICS label bureau protocol. The MP3 community could build enormously useful services on top of this, even without adopting a more general framework such as that provided by RDF, but they have stopped short of the next step.

On the contrary, while MP3 CD rippers currently embed lots of descriptive information (track listings) right into the encoding, they omit the most crucial piece of data from a fan's point of view: the CD and track identifiers. The simple unique identifier for a song on a CD, while only a tiny fragment of data, could

Microsoft Exhibit 1019

allow both peer-to-peer and web applications to hook into a marketplace of descriptive services. How could MP3 services use this information?

One application is to update the metadata inside MP3 files, either to correct errors or to add additional information. If we don't know which CD an MP3 file was derived from, it becomes hard to know which MP3 files to update when we learn more about that CD. MP3s of collected works (i.e., compilations) typically have very poor embedded metadata. Artist names often appear inside the track name, for example. This makes for difficulties in finding information: If I want to generate a browsable listing organized alphabetically by artist, I don't want half the songs filed away under "Various Artists," nor do I want to find dozens of artist names in the "By Track Title" listings. Embedding unique identifiers in MP3s would allow this mess to be fixed at a later date.

Another example can be found in the practice of sharing playlists: Given some convention for identifying songs and tracks, we can describe virtual, personalized compilation albums that another listener can recreate on his personal system by asking a peer-to-peer network for files representing those tracks. Unique identification strategies would provide the architectural glue that would allow us to reconnect fragmented information resources. Were someone to put a unique identification service in place, we could soon expect all kinds of new applications built on top:

- Collaborative filtering ("Who likes songs that I like?")
- E-commerce ("Where can I can I buy this T-shirt, CD, or book?" or "Is there a compilation album containing these tracks?")
- Discovery ("What are the words to this song?" or "Where can I find other offerings by this artist?")

The lesson for peer-to-peer metadata architecture is simple. Unique identifiers create markets. If you want to build interesting peer-to-peer applications that hook into a wide range of additional services, adopt the same strategy for uniquely identifying things that others are using.

Conclusion

Metadata applied at a fundamental level, early in the game, will provide rich semantics upon which innovators can build peer-to-peer applications that will amaze us with their flexibility. While the symmetry of peer-to-peer brings about a host of new and interesting ways of interacting, there's no substitute for taking the opportunity to rethink our assumptions and learn from the mistakes made

Microsoft Exhibit 1019

on the Web. Let's not continue the screen-scraping modus operandi; rather, let's replace extrapolation with forethought and rich assertions.

To summarize with a call to action for peer-to-peer architects, project leaders, developers, and end users:

- Use a single, coherent metadata framework such as that provided by RDF. When it comes to metadata, the network becomes a poorer information resource whenever we create artificial boundaries between metadata applications.
- Work on the commonalities between seemingly disparate data sources and formats. Work in your community to agree on some sort of common descriptive concepts. If such concepts already exist, borrow them.
- Describe your resources well, in a standard way, getting involved in this standardization process itself where necessary. Be sure to make as much of this description as possible available to peer applications and end users through clear semantics and simple APIs.
- Design ways of searching for (and finding) resources on the Net that take full advantage of any exposed metadata.

Microsoft Exhibit 1019

chapter fourteen

Performance

Theodore Hong, Imperial College of Science, Technology, and Medicine

We live in the era of speed. Practically as a matter of course, we expect each day to bring faster disks, faster networks, and above all, faster processors. Recently, a research group at the University of Arizona even published a tongue-in-cheek article arguing that large calculations could be done more quickly by slacking off for a few months first, then buying a faster computer:

> [B]y fine tuning your slacktitude you can actually accomplish more than either the lazy bum at the beach for two years or the hard working sucker who got started immediately. Indeed with a little bit of algebra we convince ourselves that there exists an optimal slack time $s\star$.*

In a world like this, one might well wonder whether performance is worth paying attention to anymore. For peer-to-peer file-sharing systems, the answer is a definite yes, for reasons I will explain in the next section.

Let me first emphasize that by performance, I don't mean abstract numerical benchmarks such as, "How many milliseconds will it take to render this many millions of polygons?" Rather, I want to know the answers to questions such as, "How long will it take to retrieve this file?" or "How much bandwidth will this query consume?" These answers will have a direct impact on the success and usability of a system.

Fault tolerance is another significant concern. Peer-to-peer operates in an inherently unreliable environment, since it depends on the personal resources of ordinary individual users. These resources may become unexpectedly unavailable at any time, for a variety of reasons ranging from users disconnecting from

* C. Gottbrath, J. Bailin, C. Meakin, T. Thompson, and J.J. Charfman (1999), "The Effects of Moore's Law and Slacking on Large Computations," arXiv:astro-ph/9912202.

Microsoft Exhibit 1019

the network or powering off a machine to users simply deciding not to participate any longer. In addition to these essentially random failures, personal machines tend to be more vulnerable than dedicated servers to directed hacking attacks or even legal action against their operators. Therefore, peer-to-peer systems need to anticipate failures as ordinary, rather than extraordinary, occurrences, and must be designed in a way that promotes redundancy and graceful degradation of performance.

Scaling is a third important consideration. The massive user bases of Napster and of the Web have clearly shown how huge the demand on a successful information-sharing system can potentially be. A designer of a new peer-to-peer system must think optimistically and plan for how it might scale under strains orders of magnitude larger in the future. If local indices of data are kept, will they overflow? If broadcasts are used, will they saturate the network? Scalability will also be influenced by performance: some design inefficiencies may pass unnoticed with ten thousand users, but what happens when the user base hits ten million or more? A recent report from Gnutella analysts Clip2, indicating that Gnutella may already be encountering a scaling barrier, should serve to sound a note of warning.

A note on terminology

We can classify peer-to-peer systems into three main categories, broadly speaking: centrally coordinated, hierarchical, and decentralized.

In a centrally coordinated system, coordination between peers is controlled and mediated by a central server, although peers may later act on information received from the central server to contact one another directly. Napster and SETI@home fall into this category.

A hierarchical peer-to-peer system devolves some or all of the coordination responsibility down from the center to a tree of coordinators. In this arrangement, peers are organized into hierarchies of groups, where communication between peers in the same group is mediated by a local coordinator, but communication between peers in different groups is passed upwards to a higher-level coordinator. Some examples are the Domain Name System (DNS) and the Squid web proxy cache.

Finally, completely decentralized peer-to-peer systems have no notion of global coordination at all. Communication is handled entirely by peers operating at a local level. This usually implies some type of forwarding mechanism in which

Microsoft Exhibit 1019

peers forward messages on behalf of other peers. Freenet and Gnutella are examples in this last category.

In this chapter, when I refer to peer-to-peer systems, I will be talking only about decentralized peer-to-peer. Since the performance issues of centralized systems have been discussed so much, it will be interesting to look at the issues of a fully decentralized system.

Why performance matters

Several factors combine to make decentralized peer-to-peer systems more sensitive to performance issues than other types of software. First, the essential characteristic of such systems is communication—a characteristic that makes them fundamentally dependent on the network. In network communication, as every dial-up user knows, connection speed dominates processor and I/O speed as the bottleneck. Since this situation will most likely persist into the foreseeable future, Moore's Law (so helpful elsewhere) provides little comfort. The problem is compounded by the highly parallel nature of peer-to-peer: A connection fast enough to talk to one remote peer quickly becomes much less so for ten trying to connect simultaneously. Thus, traffic minimization and load balancing become important considerations.

Second, decentralized systems like Freenet and Gnutella need to use messages that are forwarded over many hops from one peer to the next. Since there is no central server to maintain a master index, it necessarily takes more effort to search through the system to find out where data is. Each hop not only adds to the total bandwidth load but also increases the time needed to perform a query, since it takes a nontrivial amount of time to set up a connection to the next peer or to discover that it is down. As mentioned previously, the latter occurrence can be extremely common in peer-to-peer environments. If a peer is unreachable, TCP/IP can take up to several minutes to time out the connection. Multiply that by several times for retries to other peers and add the time needed to actually send the message over a possibly slow dial-up connection, and the elapsed time per hop can get quite high. It is therefore important to cut down on the number of hops that messages travel.

Third, the balance between resource providers and consumers must be considered. Like their counterparts in the real world, peer-to-peer communities depend on the presence of a sufficient base of communal participation and cooperation in order to function successfully.

Microsoft Exhibit 1019

However, there will always be those who consume resources without giving any back. Recent analysis by Eytan Adar and Bernardo Huberman at Xerox PARC* indicates that as many as 70% of current Gnutella users may be sharing no files at all.

If a high enough proportion of users are free riders, performance degrades for those who do contribute. A substantial decline in performance may impel some contributors to pull out of the system altogether. Their withdrawal worsens the situation further for the remainder, who will have even less incentive to stay, leading to a downward spiral (the well-known "tragedy of the commons").

To avoid this outcome, system designers must take into account the impact of free riding on performance and devise strategies to encourage higher rates of community participation. Some such strategies are discussed in Chapter 16, *Accountability*.

Bandwidth barriers

There has been some progress on the network speed front, of course. Today's 56-Kbps dial-up lines are a huge improvement on the 300-baud modems of yore. Still, true broadband has been slow to arrive.

Clip2's analysis of Gnutella is instructive in showing how bandwidth limitations can affect system capabilities. Based on a series of measurements over a period of a month, Clip2 noted an apparent scalability barrier of substantial performance degradation when query rates went above 10 queries per second. To explain this, they proposed the following model. A typical Gnutella query message is about 560 bits long, including TCP/IP headers. Clip2 observed that queries made up approximately a quarter of all message traffic, with another half being pings and the remainder miscellaneous messages. At any given time, Gnutella peers were seen to have an average of three remote peers actively connected simultaneously. Taking these numbers together, we get the following average burden on a user's link:

```
    10 queries per second
×  560 bits per query
×    4 to account for the other three-quarters of message traffic
×    3 simultaneous connections
-------------------------------------------------------------
67,200 bits per second
```

* E. Adar and B.A. Huberman (2000), "Free Riding on Gnutella," *First Monday* 5(10), *http://firstmonday.org/issues/issue5_10/adar/index.html*.

Microsoft Exhibit 1019

That's more than enough to saturate a 56-Kbps link. This calculation suggests that 10 queries per second is the maximum rate the system can handle in the presence of a significant population of dial-up users.

Even when broadband finally becomes widespread, it is unlikely to eliminate the importance of conserving bandwidth and usher in a new era of plenty. Just as building more highways failed to decrease traffic congestion because people drove more, adding more bandwidth just causes people to send larger files. Today's kilobit audio swapping becomes tomorrow's megabit video swapping. Hence, bandwidth conservation is likely to remain important for quite some time in the foreseeable future.

It's a small, small world

In 1967, Harvard professor Stanley Milgram mailed 160 letters to a set of randomly chosen people living in Omaha, Nebraska. He asked them to participate in an unusual social experiment in which they were to try to pass these letters to a given target person, a stockbroker working in Boston, Massachusetts, using only intermediaries known to one another on a first-name basis. That is, each person would pass her letter to a friend whom she thought might bring the letter closest to the target; the friend would then pass it on to another friend, and so on until the letter reached someone who knew the target personally and could give it to him. For example, an engineer in Omaha, on receiving the letter, passed it to a transplanted New Englander living in Bellevue, Nebraska, who passed it to a math teacher in Littleton, Massachusetts, who passed it to a school principal in a Boston suburb, who passed it to a local storekeeper, who gave it to a surprised stockbroker.

In all, 42 letters made it through, via a median number of just 5.5 intermediaries. Such a surprisingly low number, compared to the then-U.S. population of 200 million, demonstrated concretely for the first time what has become popularly known as the *small-world effect*. This phenomenon is familiar to anyone who has exclaimed "Small world, isn't it!" upon discovering a mutual acquaintance shared with a stranger.

Milgram's experiment was designed to explore the properties of *social networks*: the interconnecting bonds of friendship among individuals in a society. One way we can think about social networks is to use the mathematical discipline of *graph theory*. Formally, a *graph* is defined as a collection of points (called *vertices*)

Microsoft Exhibit 1019

that are connected in pairs by lines (called *edges*).* Figure 14-1 shows an example of a graph.

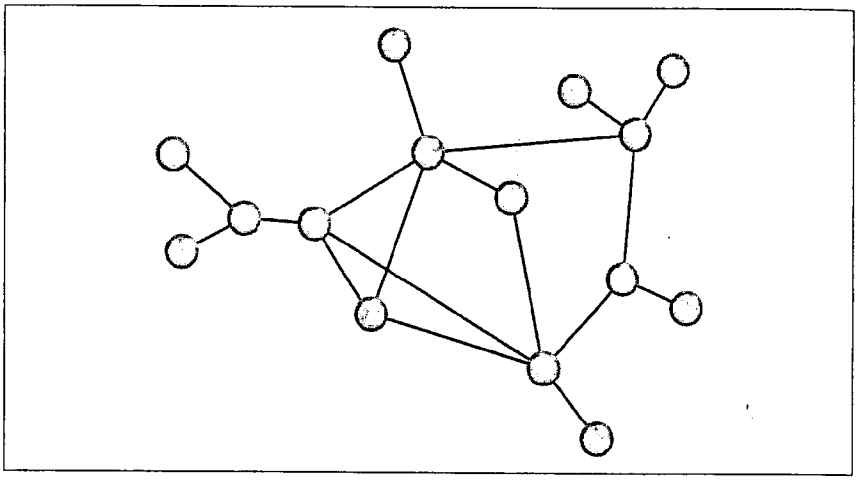

Figure 14-1. An example of a graph

How are graphs related to social networks? We can represent a social network as a graph by creating a vertex for each individual in the group and adding an edge between two vertices whenever the corresponding individuals know one another. Each vertex will have a different number of edges connected to it going to different places, depending on how wide that person's circle of acquaintances is. The resulting structure is likely to be extremely complex; for example, a graph for the United States would contain over 280 million vertices connected by a finely tangled web of edges.

Computer networks bear a strong resemblance to social networks and can be represented by graphs in a similar way. In fact, you've probably seen such a graph already if you've ever looked at a connectivity map for a LAN or WAN, although you might not have thought of it that way. In these maps, points representing individual computers or routers are equivalent to graph vertices, and lines representing physical links between machines are edges.

Another electronic analogue to a social network is the World Wide Web. The Web can be viewed as a graph in which web pages are vertices and hyperlinks are edges. Just as friendship links in a social network tend to connect members

* By the way, these graphs have nothing to do with the familiar graphs of equations used in algebra.

Microsoft Exhibit 1019

of the same social circle, hyperlinks frequently connect web pages that share a common theme or topic.

There is a slight complication because (unlike friendships) hyperlinks are one-way; that is, you can follow a hyperlink from a source page to a target page but not the reverse. For Web links, properly speaking, we need to use a *directed graph*, which is a graph in which edges point from a source vertex to a target vertex, rather than connecting vertices symmetrically. Directed graphs are usually represented by drawing their edges as arrows rather than lines, as shown in Figure 14-2.

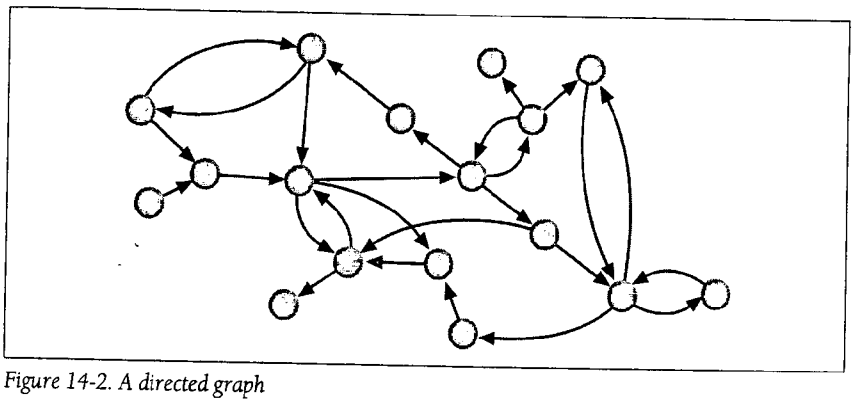

Figure 14-2. A directed graph

Most importantly for our purposes, peer-to-peer networks can be regarded as graphs as well. We can create a Freenet graph, for example, by creating a vertex for each computer running a Freenet node and linking each node by a directed edge to every node referenced in its data store. Similarly, a Gnutella graph would have a vertex for each computer running a Gnutella "servent" and edges linking servents that are connected to each other. These graphs form a useful abstract representation of the underlying networks. By analyzing them mathematically, we ought to be able to gain some insight into the functioning of the corresponding systems.

An excursion into graph theory

There are a number of interesting questions you can ask about graphs. One immediate question to ask is whether or not it is *connected*. That is, is it always possible to get from any vertex (or individual) to any other via some chain of intermediaries? Or are there some groups which are completely isolated from one another, and never the twain shall meet?

Microsoft Exhibit 1019

An important property to note in connection with this question is that paths in a graph are *transitive*. This means that if there is a path from point A to point B, and also a path from point B to point C, then there must be a path from A to C. This fact might seem too obvious to need stating, but it has broader consequences. Suppose there are two separate groups of vertices forming two subgraphs, each connected within itself but disconnected from the other. Then adding just one edge from any vertex V in one group to any vertex W in the other, as in Figure 14-3, will make the graph as a whole connected. This follows from transitivity: by assumption there is a path from every vertex in the first group to V, and a path from W to every vertex in the second group, so adding an edge between V and W will complete a path from every vertex in the first group to every vertex in the second (and vice versa). Conversely, deleting one critical edge may cause a graph to become disconnected, a topic we will return to later in the context of network robustness.

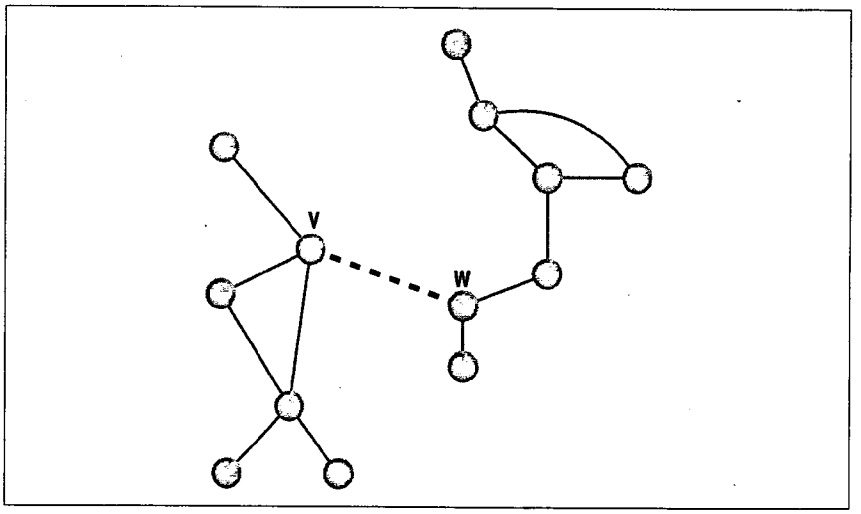

Figure 14-3. Adding an edge between V and W connects the two subgraphs

If it is possible to get from any vertex to any other by some path, a natural follow-up question to ask is how long these paths are. One useful measure to consider is the following: for each pair of vertices in the graph, find the length of the shortest path between them; then, take the average over all pairs. This number, which we'll call the *characteristic pathlength* of the graph, gives a sense of how far apart points are in the network.

In the networking context, the relevance of these two questions is immediately apparent. For example, performing a *traceroute* from one machine to another is equivalent to finding a path between two vertices in the corresponding graph.

Microsoft Exhibit 1019

Finding out whether a route exists, and how many hops it takes, are basic questions in network analysis and troubleshooting.

For decentralized peer-to-peer networks, these two questions have a similar significance. The first tells us which peers can communicate with one another (via some message-forwarding route); the second, how much effort is involved in doing so. To see how we can get a handle on these questions, let's return to the letter-passing experiment in more depth. Then we'll see if we can apply any insights to the peer-to-peer situation.

The small-world model

The success of Milgram's volunteers in moving letters between the seemingly disparate worlds of rural heartland and urban metropolis suggests that the social network of the United States is indeed connected. Its characteristic pathlength corresponds to the median number of intermediaries needed to complete a chain, measured to be about six.

Intuitively, it seems that the pathlength of such a large network ought to be much higher. Most people's social circles are highly cliquish or clustered; that is, most of the people whom you know also know each other. Equivalently, many of the friends of your friends are people whom you know already. So taking additional hops may not increase the number of people within reach by much. It seems that a large number of hops would be necessary to break out of one social circle, travel across the country, and reach another, particularly given the size of the U.S. How then can we explain Milgram's measurement?

The key to understanding the result lies in the distribution of links within social networks. In any social grouping, some acquaintances will be relatively isolated and contribute few new contacts, whereas others will have more wide-ranging connections and be able to serve as bridges between far-flung social clusters. These bridging vertices play a critical role in bringing the network closer together. In the Milgram experiment, for example, a quarter of all the chains reaching the target person passed through a single person, a local storekeeper. Half the chains were mediated by just three people, who collectively acted as gateways between the target and the wider world.

It turns out that the presence of even a small number of bridges can dramatically reduce the lengths of paths in a graph, as shown by a recent paper by Duncan Watts and Steven Strogatz in the journal *Nature*.* They began by considering a

* D.J. Watts and S.H. Strogatz (1998), "Collective Dynamics of 'Small-World' Networks," *Nature* 393, p.440.

Microsoft Exhibit 1019

simple type of graph called a *regular graph*, which consists of a ring of n vertices, each of which is connected to its nearest k neighbors. For example, if k is 4, each vertex is connected to its nearest two neighbors on each side (four in total), giving a graph such as the one shown in Figure 14-4.

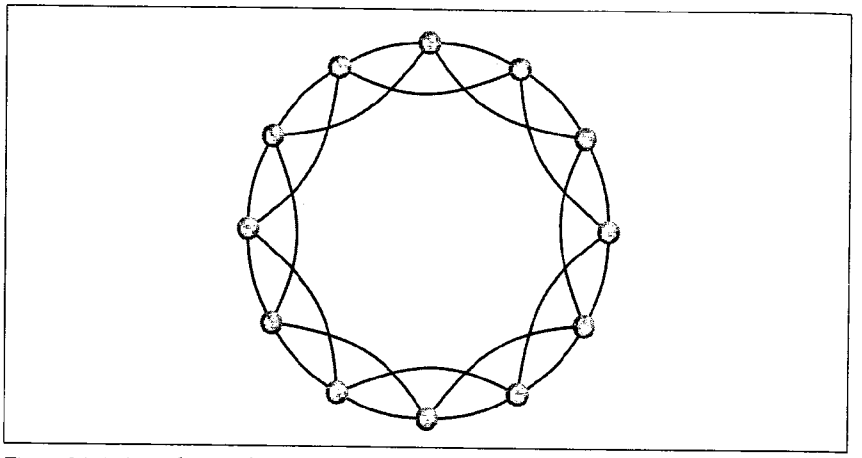

Figure 14-4. A regular graph

If we look at large regular graphs in which n is much larger than k, which in turn is much larger than 1, the pathlength can be shown to be approximately $n/2k$. For example, if n is 4,096 and k is 8, then $n/2k$ is 256—a very large number of hops to take to get where you're going! (Informally, we can justify the formula $n/2k$ by noticing that it equals half the number of hops it takes to get to the opposite side of the ring. We say only half because we are averaging over all pairs, some of which will be close neighbors and some of which will be on opposite sides.)

Another property of regular graphs is that they are highly clustered, since all of their links are contained within local neighborhoods. To make this notion more precise, we can define a measure of clustering as follows. For the k neighbors of a given vertex, the total number of possible connections among them is $k \times (k-1)/2$. Let's define the *clustering coefficient* of a vertex as the proportion (between 0 and 1) of these possible links that are actually present in the graph. For example, in the regular graph of Figure 14-4, each vertex has four neighbors. There are a total of $(4 \times 3)/2 = 6$ possible connections among the four neighbors (not counting the original vertex itself), of which 3 are present in the graph. Therefore the clustering coefficient of each vertex is $3/6 = 0.5$.

In social terms, this coefficient can be thought of as counting the number of connections among a person's friends—a measure of the cliquishness of a group.

Microsoft Exhibit 1019

If we do the math, it can be shown that as the number of vertices in the graph increases, the clustering coefficient approaches a constant value of 0.75 (very cliquish).

More generally, in a non-regular graph, different vertices will have different coefficients. So we define the clustering coefficient of a whole graph as the average of all the clustering coefficients of the individual vertices.

The opposite of the completely ordered regular graph is the *random graph*. This is just a graph whose vertices are connected to each other at random. Random graphs can be categorized by the number of vertices n and the average number of edges per vertex k. Notice that a random graph and a regular graph having the same values for n and k will be comparable in the sense that both will have the same total number of vertices and edges. For example, the random graph shown in Figure 14-5 has the same number of vertices (12) and edges (24) as the regular graph in Figure 14-4. It turns out that for large random graphs, the pathlength is approximately $\log n/\log k$, while the clustering coefficient is approximately k/n. So using our previous example, where n was 4,096 and k was 8, the pathlength would be $\log 4{,}096/\log 8 = 4$—much better than the 256 hops for the regular graph!

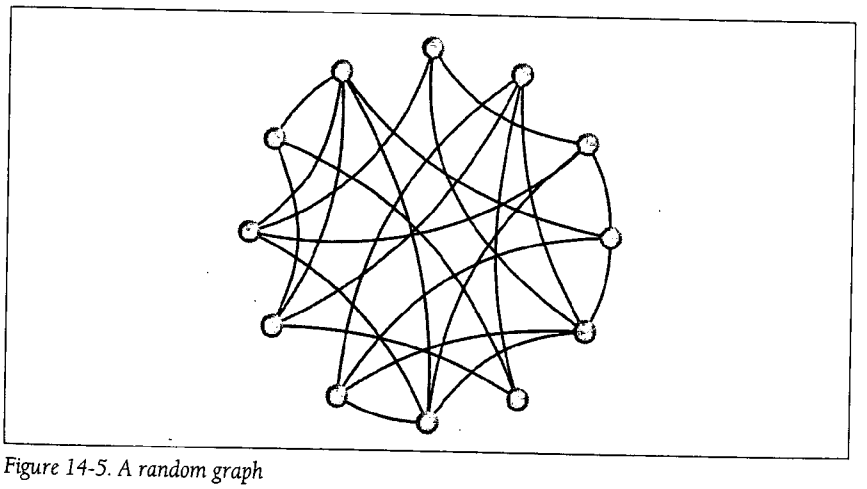

Figure 14-5. A random graph

On the other hand, the clustering coefficient would be $8/4{,}096 \approx 0.002$—much less than the regular graph's 0.75. In fact, as n gets larger, the clustering coefficient becomes practically 0.

If we compare these two extremes, we can see that the regular graph has high clustering and a high pathlength, whereas the random graph has very low clustering and a comparatively low pathlength. (To be more precise, the pathlength

Microsoft Exhibit 1019

of the regular graph grows linearly as n gets larger, but the pathlength of the random graph grows only logarithmically.)

What about intermediate cases? Most real-world networks, whether social networks or peer-to-peer networks, lie somewhere in between—neither completely regular nor completely random. How will they behave in terms of clustering and pathlength?

Watts and Strogatz used a clever trick to explore the in-between region. Starting with a 1000-node regular graph with k equal to 10, they "rewired" it by taking each edge in turn and, with probability p, moving it to connect to a different, randomly chosen vertex. When p is 0, the regular graph remains unchanged; when p is 1, a random graph results. The region we are interested in is the region where p is between 0 and 1. Figure 14-6 shows one possible rewiring of Figure 14-4 with p set to 0.5.

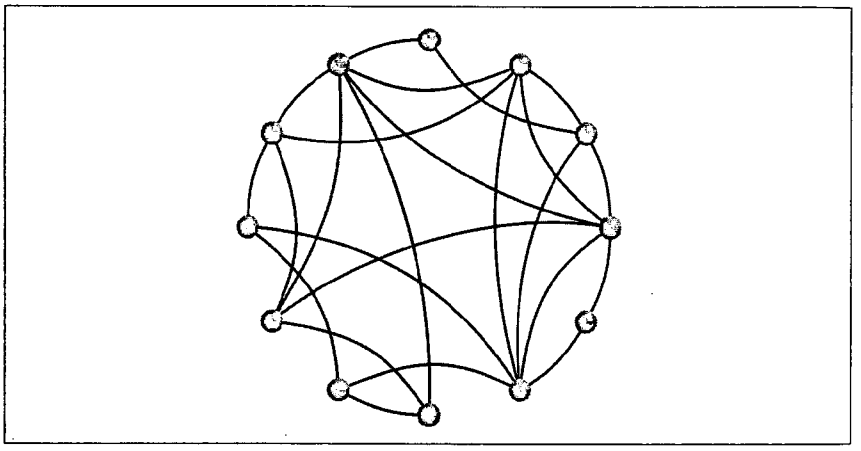

Figure 14-6. A rewiring of a regular graph

Surprisingly, what they found was that with larger p, clustering remains high but pathlength drops precipitously, as shown in Figure 14-7. Rewiring with p as low as 0.001 (that is, rewiring only about 0.1% of the edges) cuts the pathlength in half while leaving clustering virtually unchanged. At a p value of 0.01, the graph has taken on hybrid characteristics. Locally, its clustering coefficient still looks essentially like that of the regular graph. Globally, however, its pathlength has nearly dropped to the random-graph level. Watts and Strogatz dubbed graphs with this combination of high local clustering and short global pathlengths *small-world graphs*.

Two important implications can be seen. First, only a small amount of rewiring is needed to promote the small-world transition. Second, the transition is barely

Microsoft Exhibit 1019

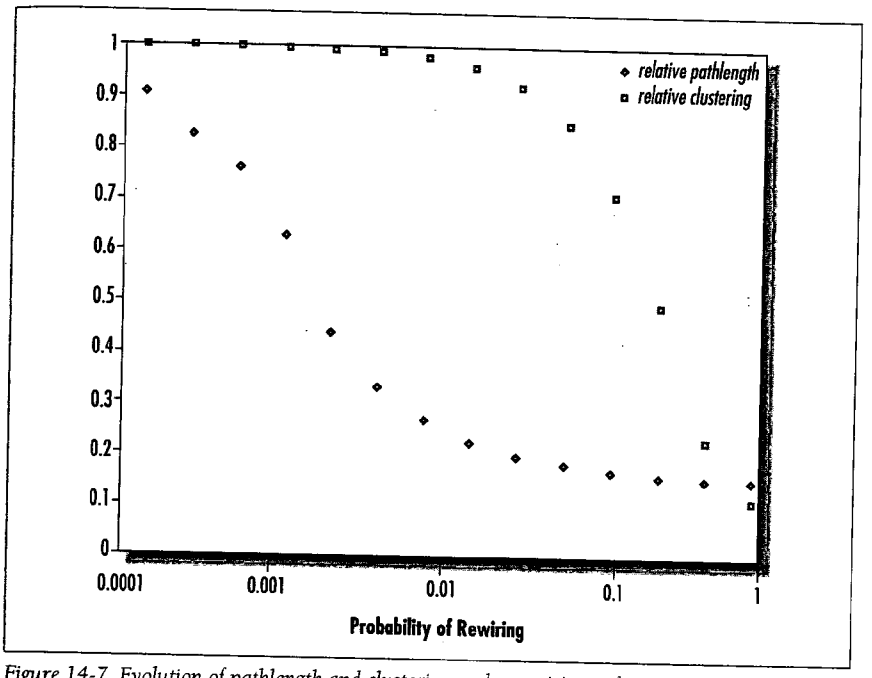

Figure 14-7. Evolution of pathlength and clustering under rewiring, relative to initial values

noticeable at the local level. Hence it is difficult to tell whether or not your world is a small world, although it won't take much effort to turn it into one if it isn't.

These results can explain the small-world characteristics of the U.S. social network. Even if local groups are highly clustered, as long as a small fraction (1% or even fewer) of individuals have long-range connections outside the group, pathlengths will be low. This happens because transitivity causes such individuals to act as shortcuts linking entire communities together. A shortcut doesn't benefit just a single individual, but also everyone linked to her, and everyone linked to those who are linked to her, and so on. All can take advantage of the shortcut, greatly shortening the characteristic pathlength. On the other hand, changing one local connection to a long-range one has only a small effect on the clustering coefficient.

Let's now look at how we can apply some of the concepts of the small-world model to peer-to-peer by considering a pair of case studies.

Case study 1: Freenet

The small-world effect is fundamental to Freenet's operation. As with Milgram's letters, Freenet queries are forwarded from one peer to the next according to

Microsoft Exhibit 1019

local decisions about which potential recipient might make the most progress towards the target. Unlike Milgram's letters, however, Freenet messages are not targeted to a specific named peer but toward any peer having a desired file in its data store.

To take a concrete example, suppose I were trying to obtain a copy of *Peer-to-Peer*. Using Milgram's method, I could do this by trying to get a letter to Tim O'Reilly asking for a copy of the book. I might begin by passing it to my friend Dan (who lives in Boston), who might pass it to his friend James (who works in computers), who might pass it to his friend Andy (who works for Tim), who could pass it to Tim himself. Using Freenet's algorithm, I don't try to contact a particular person. Instead, I might ask my friend Alison (who I know has other O'Reilly books) if she has a copy. If she didn't, she might similarly ask her friend Helena, and so on. Freenet's routing is based on evaluating peers' bookshelves rather than their contacts—any peer owning a copy can reply, not just Tim O'Reilly specifically.

For the Freenet algorithm to work, we need two properties to hold. First, the Freenet graph must be connected, so that it is possible for any request to eventually reach some peer where the data is stored. (This assumes, of course, that the data does exist on Freenet somewhere.) Second, despite the large size of the network, short routes must exist between any two arbitrary peers, making it possible to pass messages between them in a reasonable number of hops. In other words, we want Freenet to be a small world.

The first property is easy. Connectedness can be achieved by growing the network incrementally from some initial core. If each new node starts off by linking itself to one or more introductory nodes already known to be reachable from the core, transitivity will assure a single network rather than several disconnected ones. There is a potential problem, however: If the introductory node fails or drops out, the new node and later nodes connected to it might become stranded.

Freenet's request and insert mechanisms combat this problem by adding additional redundant links to the network over time. Even if a new node starts with only a single reference to an introductory node, each successful request will cause it to gain more references to other nodes. These references will provide more links into the network, alleviating the dependence on the introductory node. Conversely, performing inserts creates links in the opposite direction, as nodes deeper in the network gain references to the inserting node. Nonetheless, the effect of node failures needs to be examined more closely. We will return to this subject later.

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The second property presents more of a challenge. As we saw earlier, it is difficult to tell from local examination alone whether or not the global network is a small world, and Freenet's anonymity properties deliberately prevent us from measuring the global network directly. For example, it is impossible to even find out how many nodes there are. Nor do we know precisely which files are stored in the network or where, so it is hard to infer much from local request outcomes. We therefore turn to simulation.

Initial experiments

Fortunately, simulation indicates that Freenet networks do evolve small-world characteristics. Following Watts and Strogatz, we can initialize a simulated Freenet network with a regular topology and see how it behaves over time. Suppose we create a network of 1,000 identical nodes having initially empty data stores with a capacity of 50 data items and 200 additional references each. To minimally bootstrap the network's connectivity, let's number the nodes and give each node references to 2 nodes immediately before and after it numerically (modulo 1,000). For example, node 0 would be connected to nodes 998, 999, 1, and 2. We have to associate keys with these references, so for convenience we'll use a hash of the referenced node number as the key. Using a hash has the advantage of yielding a key that is both random and consistent across the network (that is, every node having a reference to node 0 will assign the same key to the reference, namely *hash*(0)). Figure 14-8 shows some of the resulting data stores. Topologically, this network is equivalent to a directed regular graph in which n is 1,000 and k is 4.

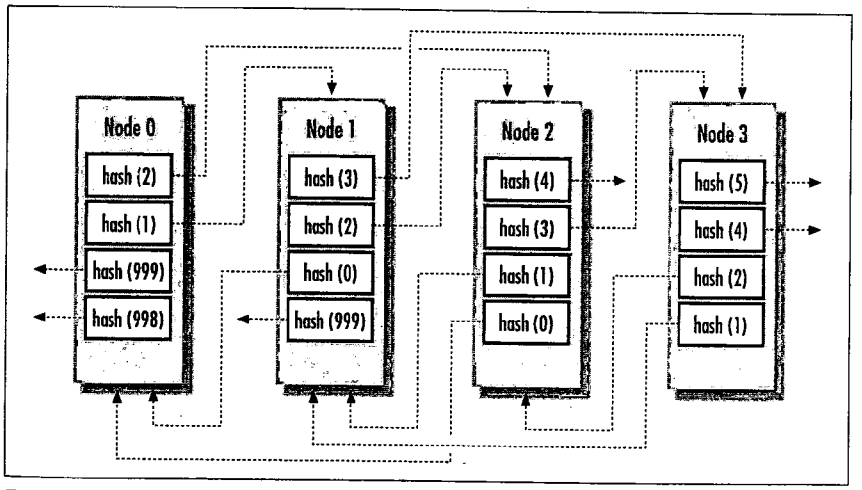

Figure 14-8. Initial data stores for a simulated network

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What are the initial characteristics of this network? Well, from the earlier discussion of regular graphs, we know that its pathlength is $n/2k$, or 1,000/8 = 125. Each node has four neighbors—for example, node 2 is connected to nodes 0, 1, 3, and 4. Of the 12 possible directed edges among these neighbors, 6 are present (from 0 to 1, 1 to 3, and 3 to 4, and from 1 to 0, 3 to 1, and 4 to 3), so the clustering coefficient is 6/12 = 0.5.

A comparable random graph, on the other hand, would have a pathlength of log 1,000/log 4 ≈ 5 and a clustering coefficient of 4/1,000 = 0.004.

Now let's simulate a simple network usage model. At each time step, pick a node at random and flip a coin to decide whether to perform a request or an insert from that node. If requesting, randomly choose a key to request from those known to be present in the network; if inserting, randomly choose a key to insert from the set of all possible keys. Somewhat arbitrarily, let's set the hops-to-live to 20 on both insert and request.

Every 100 time steps, measure the state of the network. We can directly calculate its clustering coefficient and characteristic pathlength by examining the data stores of each node to determine which other nodes it is connected to and then performing a breadth-first search on the resulting graph.

Figure 14-9 shows the results of simulating this model. Ten trials were taken, each lasting 5,000 time steps, and the results were averaged over all trials.

As we can see, the pathlength rapidly decreases by a factor of 20 within the first 500 time steps or so before leveling off. On the other hand, the clustering coefficient decreases only slowly over the entire simulation period. The final pathlength hovers slightly above 2, while the final clustering is about 0.22. If we compare these figures to the values calculated earlier for the corresponding regular graph (125 pathlength and 0.5 clustering) and random graph (5 pathlength and 0.004 clustering), we can see the small-world effect: Freenet's pathlength approximates the random graph's pathlength while its clustering coefficient is of the same order of magnitude as the regular graph.

Does the small-world effect translate into real performance, however? To answer this question, let's look at the request performance of the network over time. Every 100 time steps, we probe the network by simulating 300 requests from randomly chosen nodes in the network. During this probe period, the network is frozen so that no data is cached and no links are altered. The keys requested are chosen randomly from those known to be stored in the network and the hops-to-live is set to 500. By looking at the number of hops actually taken, we can measure the distance that a request needs to travel before finding data. For our purposes, a request that fails will be treated as taking 500 hops. At each

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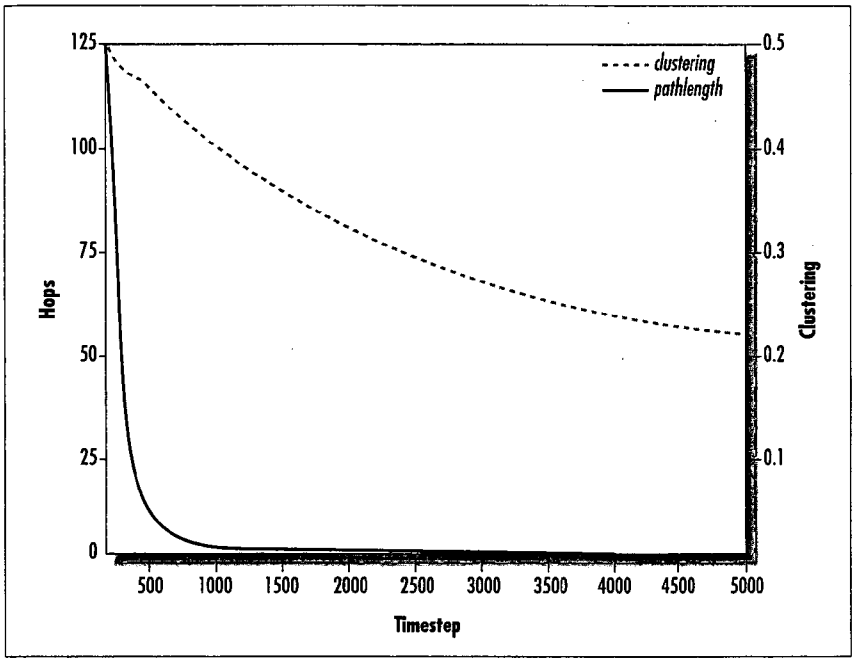

Figure 14-9. Evolution of pathlength and clustering over time in a Freenet network

snapshot, we'll plot the median pathlength of all requests (that is, the top 50% fastest requests).

These measurements are plotted in Figures 14-10 and 14-11. Reassuringly, the results indicate that Freenet does actually work. The median pathlength for requests drops from 500 at the outset to about 6 as the network converges to a stable state. That is, half of all requests in the mature network succeed within six hops. A quarter of requests succeed within just three hops or fewer.

Note that the median request pathlength of 6 is somewhat higher than the characteristic pathlength of 2. This occurs because the characteristic pathlength measures the distance along the *optimal* path between any pair of nodes. Freenet's local routing cannot always choose the globally optimal route, of course, but it manages to get close most of the time.

On the other hand, if we look at the complete distribution of final pathlengths, as shown in Figure 14-12, there are some requests that take a disproportionately long time. That is, Freenet has good average performance but poor worst-case performance, because a few bad routing choices can throw a request completely off the track.

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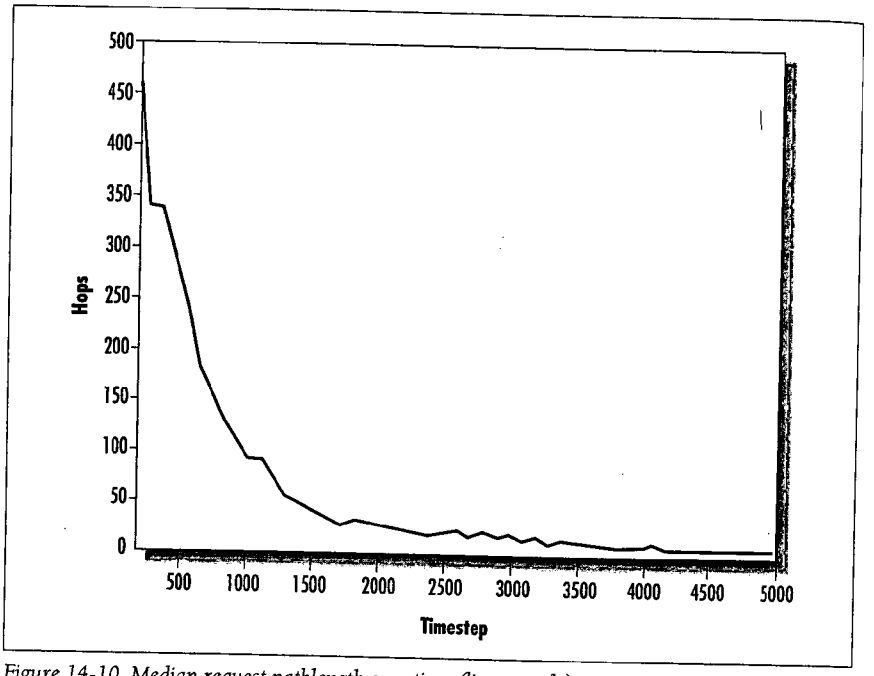

Figure 14-10. Median request pathlength over time (linear scale)

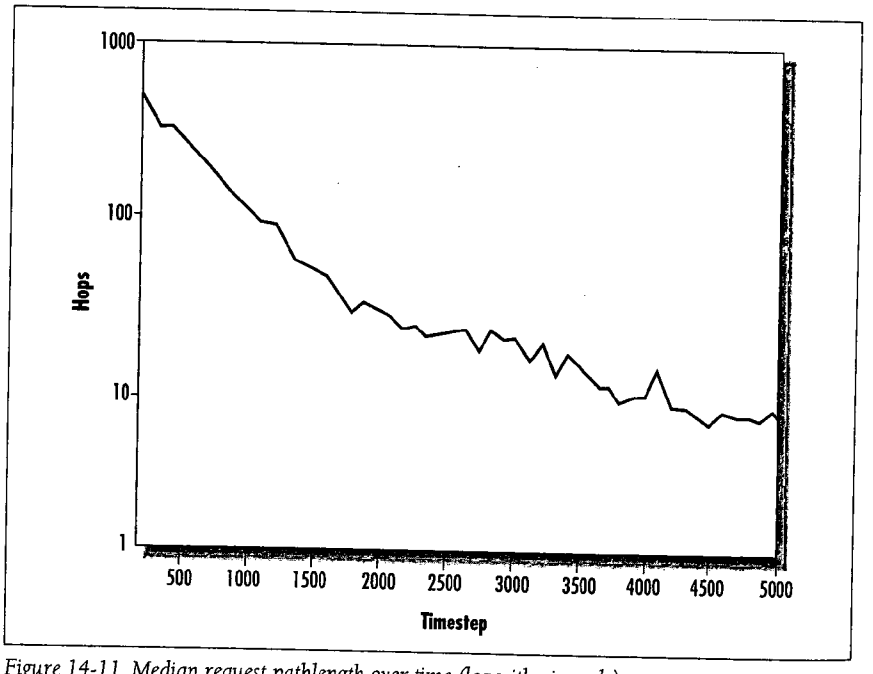

Figure 14-11. Median request pathlength over time (logarithmic scale)

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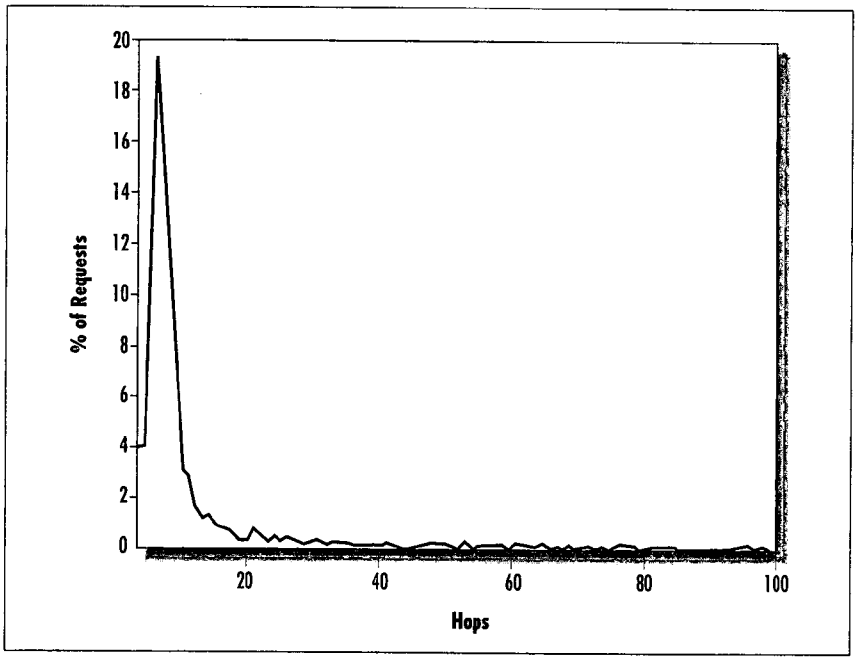

Figure 14-12. Distribution of all request pathlengths at the end of the simulation

Indeed, local routing decisions are extremely important. Although the small-world effect tells us that short routes exist between any pair of vertices in a small-world network, the tricky part is actually finding these short routes.

To illustrate this point, consider a Freenet-like system in which nodes forward query messages to some peer randomly chosen from the data store, rather than the peer associated with the closest key to the query. Performing the same simulation on this system gives the measurements shown in Figure 14-13.

We see that the median request pathlength required now is nearly 50, although analysis of the network shows the characteristic pathlength to still be about 2. This request pathlength is too high to be of much use, as 50 hops would take forever to complete. So although short paths exist in this network, we are unable to make effective use of them.

These observations make sense if we think about our intuitive experience with another small-world domain, the Web. The process of navigating on the Web from some starting point to a desired destination by following hyperlinks is quite similar to the process of forwarding a request in Freenet. A recent paper in

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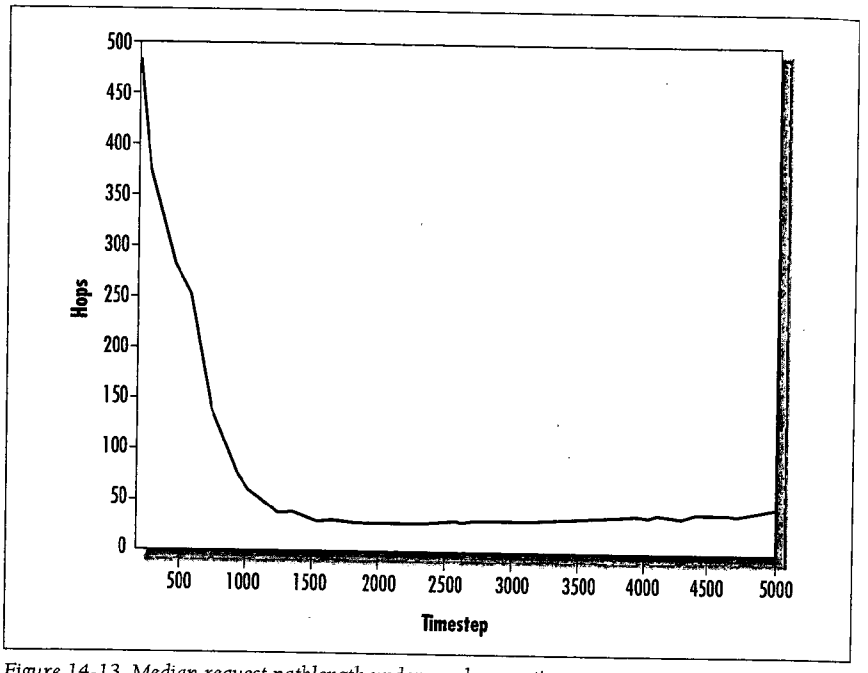

Figure 14-13. Median request pathlength under random routing

Nature by Réka Albert, Hawoong Jeong, and Albert-László Barabási* reported that the Web is a small-world network with a characteristic pathlength of 19. That is, from any given web page, it is possible to surf to any other one of the nearly 800 million reachable pages in existence with an average of 19 clicks.

However, such a path can be constructed only by an intelligent agent able to make accurate decisions about which link to follow next. Even humans often fail in this task, getting "lost in the Web." An unintelligent robot choosing links at random would clearly get nowhere. The only hope for such a robot is to apply brute-force indexing, and the force required is brute indeed: Albert *et al.* estimated that a robot attempting to locate a web page at a distance of 19 hops would need to index at least a full 10% of the Web, or some 80 million pages.

Simulating growth

Having taken a preliminary look at the evolution of a fixed Freenet network, let's now look at what happens in a network that grows over time. When a new node wants to join Freenet, it must first find (through out-of-band means) an initial

* R. Albert, H. Jeong, and A. Barabási (1999), "Diameter of the World-Wide Web," *Nature* 401, p.130.

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introductory node that is already in the network. The new node then sends an announcement message to the introductory node, which forwards it into Freenet. Each node contacted adds a reference to the new node to its data store and sends back a reply containing its own address, before forwarding the announcement on to another node chosen randomly from its data store. In turn, the new node adds all of these replies to its data store. The net result is that a set of two-way links are established between the new node and some number of existing nodes, as shown in Figure 14-14.

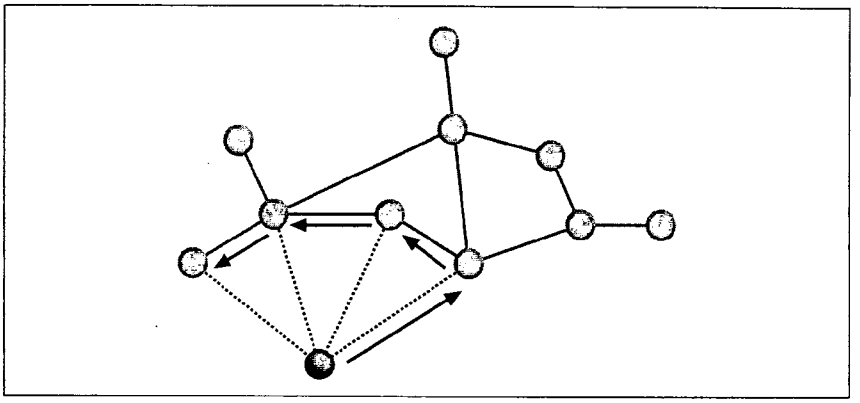

Figure 14-14. Adding a new node to Freenet (arrows show the path of the announcement message; dotted lines show the new links established)

We can simulate this evolution by the following procedure. Initialize the network with 20 nodes connected in a regular topology as before, so that we can continue to use a hops-to-live of 20 from the outset. Add a new node every 5 time steps until the network reaches a size of 1,000. When adding a new node, choose an introductory node at random and send an announcement message with hops-to-live 10. Meanwhile, inserts and requests continue on every time step as before, and probes every 100 time steps.

It might seem at first that this simulation won't realistically model the rate of growth of the network, since nodes are simply added linearly every five steps. However, simulation time need not correspond directly to real time. The effect of the model is essentially to interpose five requests between node additions, regardless of the rate of addition. In real time, we can expect that the number of requests per unit time will be proportional to the size of the network. If we assume that the rate at which new nodes join is also proportional to the size of the network, the linear ratio between request rate and joining rate is justified.

Figure 14-15 shows the results of simulating this model. As before, 10 trials were run and the results averaged over all trials.

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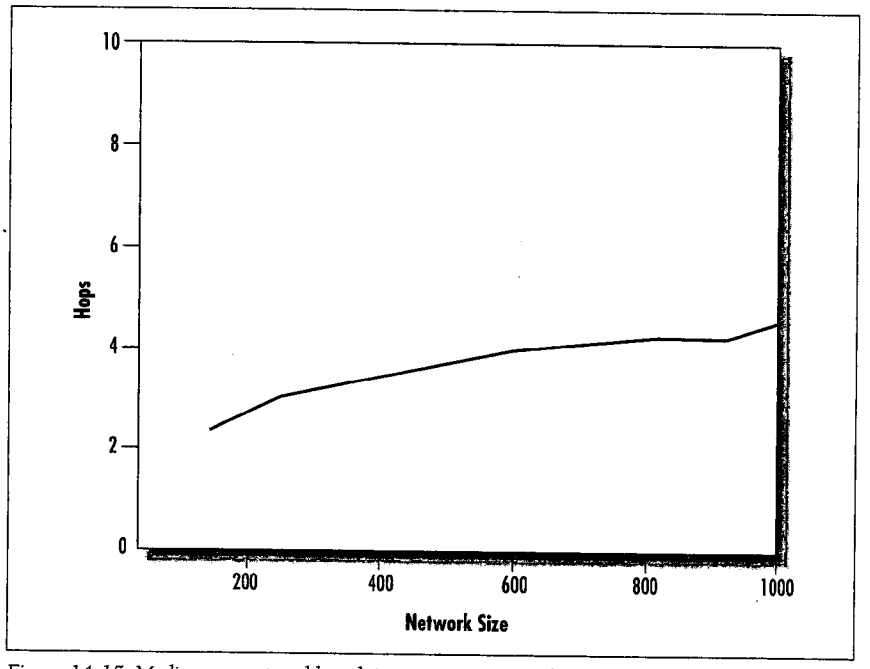

Figure 14-15. Median request pathlength in a growing network

The results are extremely promising. The request pathlength starts off low, unsurprisingly, since the network is so small initially that even random routing should find the data quickly. However, as the network grows, the request pathlength remains low.

By the end of the simulation, the network is performing even better than the fixed-size simulation having the same number of nodes. Now 50% of all requests succeed within just 5 hops or fewer, while 84% succeed within 20. Meanwhile, the characteristic pathlength and the clustering coefficient are not appreciably different from the fixed case—about 2.2 for the pathlength and about 0.25 for the clustering coefficient.

Simulating fault tolerance

Let's turn to some aspects of robustness. As mentioned earlier, an important challenge in designing a peer-to-peer system is coping with the unreliability of peers. Since peers tend to be personal machines rather than dedicated servers, they are often turned off or disconnected from the network at random. Another consideration for systems that may host content disapproved of by some group is the possibility of a deliberate attempt to bring the network down through technical or legal attacks.

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Taking as a starting point the network grown in the second simulation, we can examine the effects of two node failure scenarios. One scenario is random failure, in which nodes are simply removed at random from the network. The other scenario is targeted attack, in which the most important nodes are targeted for removal. Here we follow the approach of another paper by Albert, Jeong, and Barabási on the fault tolerance of the Internet.*

We can model the random failure scenario by progressively removing more and more nodes selected at random from the network and watching how the system's performance holds up. Figure 14-16 shows the request pathlength plotted against the percentage of nodes failing. The network remains surprisingly usable, with the median request pathlength remaining below 20 even when up to 30% of nodes fail.

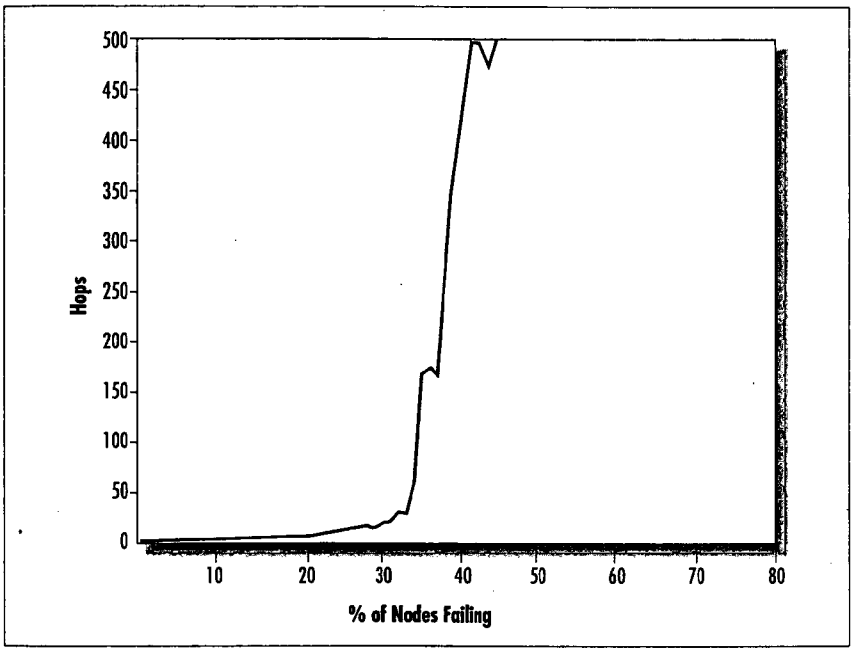

Figure 14-16. Change in request pathlength under random failure

An explanation can be offered by looking at the distribution of links within the network. If we draw a histogram of the proportion of nodes having different numbers of links, as shown in Figure 14-17, we can see that the distribution is highly skewed. Most nodes have only a few outgoing links, but a small number

* R. Albert, H. Jeong, and A. Barabási (2000), "Error and Attack Tolerance of Complex Networks," *Nature* 406, p.378.

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of nodes toward the right side of the graph are very well-connected. (The unusually large column at 250 links is an artifact of the limited data store size of 250—when larger data stores are used, this column spreads out farther to the right.)

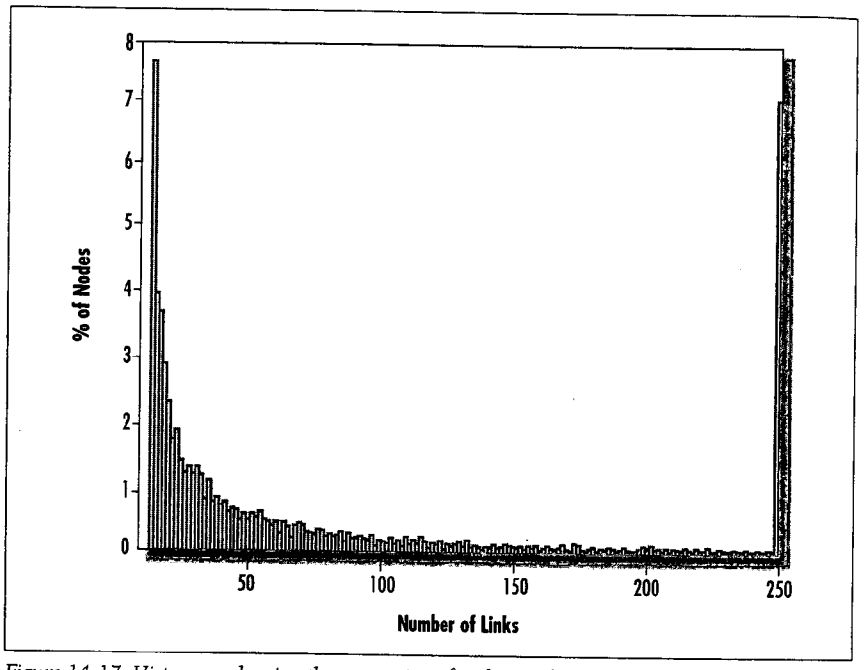

Figure 14-17. Histogram showing the proportion of nodes vs. the number of links

When nodes are randomly removed from the network, most of them will probably be nodes with few links, and thus their loss will not hurt the routing in the network much. The highly connected nodes in the right-hand tail will be able to keep the network connected. These nodes correspond to the shortcuts needed to make the small-world effect happen.

The attack scenario, on the other hand, is more dangerous. In this scenario, the most-connected nodes are preferentially removed first. Figure 14-18 shows the trend in the request pathlength as nodes are attacked. Now the network becomes unusable much more quickly, with the median request pathlength passing 20 at the 18% failure level. This demonstrates just how important those nodes in the tail are. When they are removed, the network starts to fall apart into disconnected fragments.

Figure 14-19 shows the contrast between the two failure modes in more detail, using a semi-log scale.

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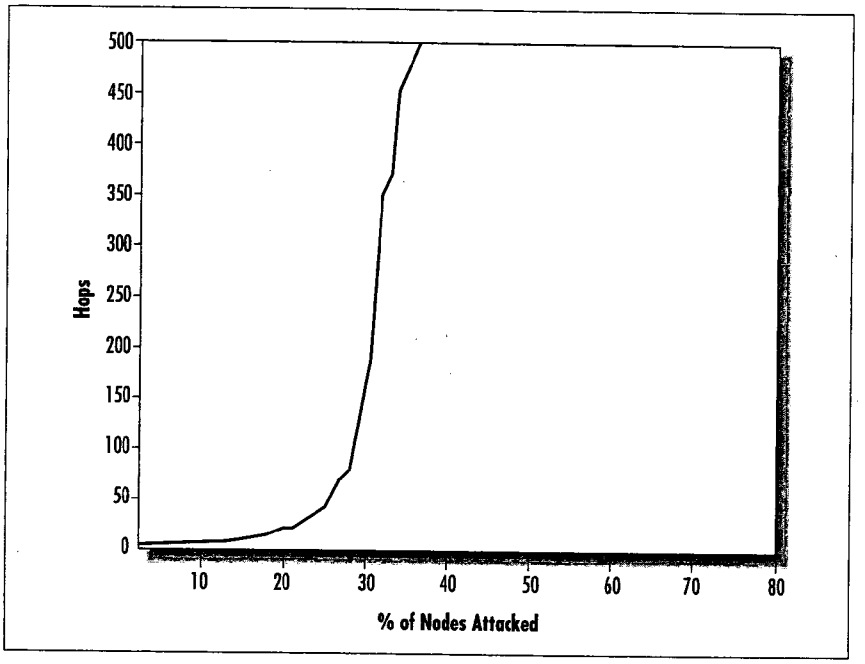

Figure 14-18. Change in request pathlength under targeted attack

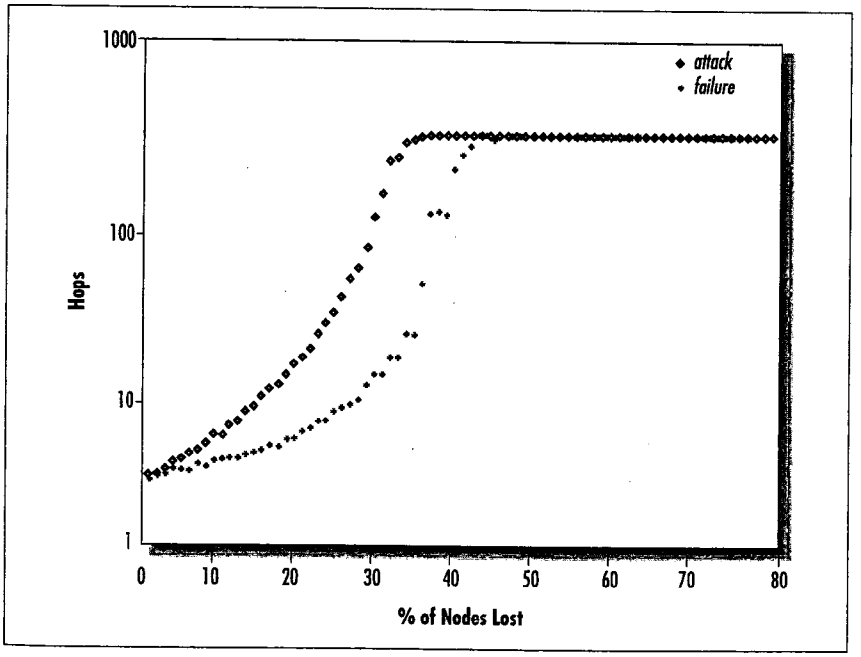

Figure 14-19. Comparison of the effects of attack and failure on median request pathlength

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Link distribution in Freenet

Where do the highly connected nodes come from? We can get some hints by trying to fit a function to the observed distribution of links. If we redraw the histogram as a log-log plot, as shown in Figure 14-20, we can see that the distribution of link numbers roughly follows a straight line (except for the anomalous point at 250). Since the equation for a downward-sloping line is:

$$y = -kx + b$$

where k and b are constants, this means that the proportion of nodes p having a given number of links L satisfies the equation:

$$\log p = -k \log L + b$$

By exponentiating both sides, we can express this relationship in a more normal-looking way as:

$$p = A \times L^{-k}$$

This is called a *scale-free* relationship, since the total number of nodes doesn't appear in the equation. Therefore it holds regardless of the size of the network, big or small. In fact, scale-free link distributions are another characteristic often used to identify small-world networks.

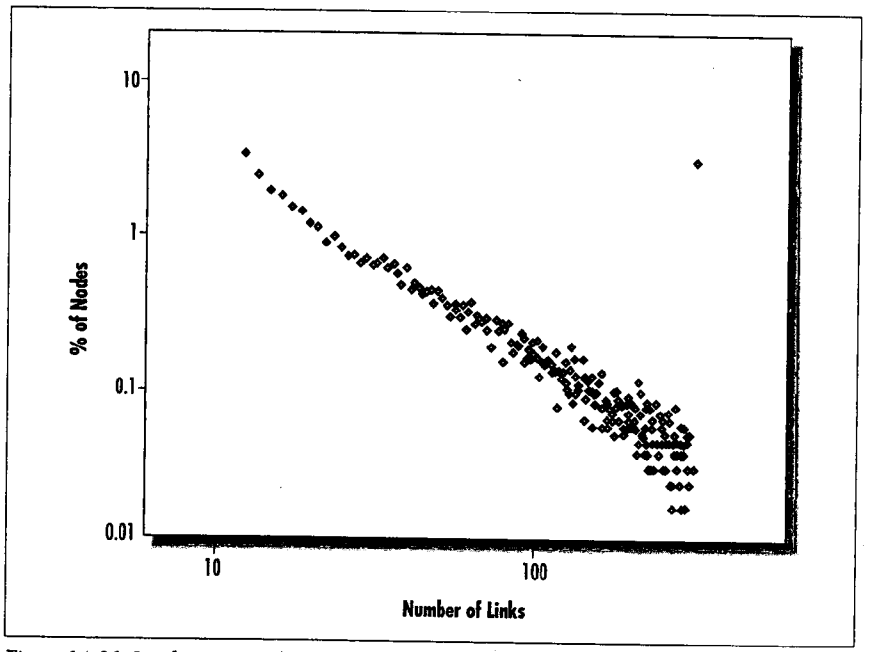

Figure 14-20. Log-log scatter plot of the proportion of nodes vs. the number of links

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It turns out that this type of relationship arises naturally from the interaction of two processes: Growth and preferential attachment. Growth just means that new nodes are added over time. Preferential attachment means that new nodes tend to add links to nodes that have a lot of links already. This makes sense because nodes that are well known (i.e., have lots of links) will tend to see more requests and hence will tend to become even better connected.

The impact of free riding

In addition to being robust against node failures, peer-to-peer systems must be able to cope with free riders. Just as in any other social system, there are always those who take from the system without contributing anything back. In the peer-to-peer context, this might mean downloading files but not sharing any for upload, or initiating queries without forwarding or answering queries from others. At best, such behavior just means increased load for everyone else; at worst, it can significantly harm the functioning of the system.

Freenet deals with free riders by simply ignoring them. If a node never provides any files, no other nodes will gain references to it. To the rest of the network, it might as well not exist, so it won't have any effect on the pathlengths of others' requests. However, its own requests will contribute to the total bandwidth load on the network while providing no additional capacity. Similarly, if a node refuses to accept incoming connections, other nodes will treat it as though it were down and try elsewhere. Only if a node drops messages without responding will untoward things start to happen, although in that case it is behaving more like a malicious node than a free riding one.

Scalability

Finally, let's consider Freenet's scaling properties. In small-world graphs, the characteristic pathlength scales logarithmically with the size of the network, since it follows the random-graph pathlength of $\log n / \log k$. That is, a geometric increase in the number of vertices results in only a linear increase in the characteristic pathlength. This means that for example, if k is 3, increasing the size of the network by 10 times would increase the pathlength by just 2. If Freenet's routing continues to work in large networks, the request pathlength should scale similarly. (Remember that the correlation between the request pathlength and the characteristic pathlength depends on the accuracy of the routing.)

Figure 14-21 shows the results of extending our earlier growth simulation up to 200,000 nodes. As hoped, the request pathlength does appear to scale logarithmically.

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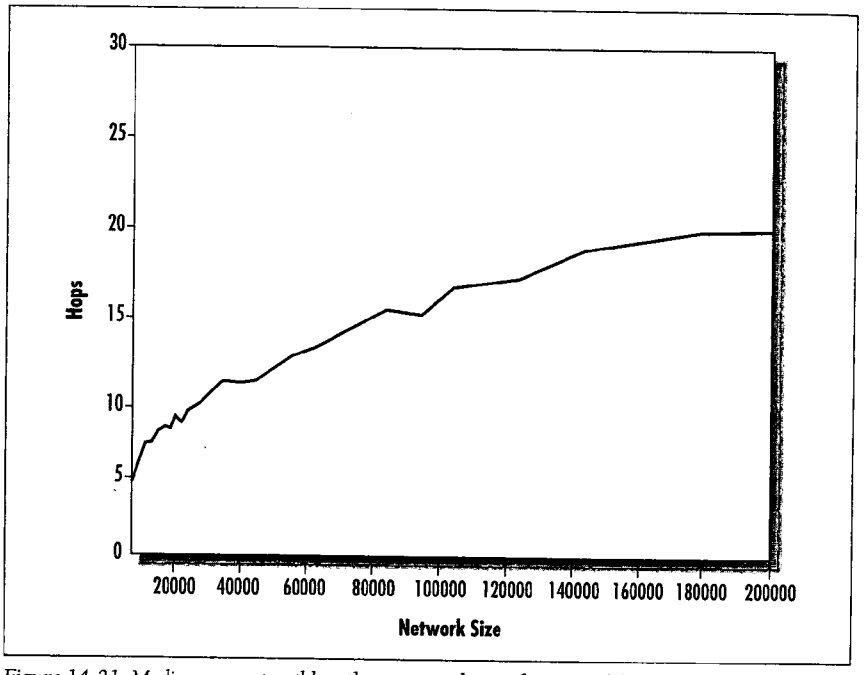

Figure 14-21. Median request pathlength vs. network size (linear scale)

We can see the scaling more clearly on the semi-log plot shown in Figure 14-22. On this plot, the data follow approximately straight lines, showing that pathlength is indeed roughly proportional to log(*size*). The median line has a "knee" where it changes slope at 50,000 nodes. This probably results from data stores becoming filled and could be corrected by creating larger data stores. Note that our data stores were limited to 250 links by the memory requirements of the simulation, whereas real Freenet nodes can easily hold thousands of references. In fact, if we recall the connectivity distribution shown in Figure 14-17, only a small number of high-capacity nodes should be necessary. Even with small data stores, the trend shows that Freenet scales very well: Doubling the network size brings a pathlength increase of only 4 hops.

The number of messages that must be transmitted per request is proportional to the request pathlength, since the latter indicates the number of times a request is forwarded. In turn, the bandwidth used is proportional to the number of messages sent. Thus, the bandwidth requirements of requests should also scale logarithmically in relation to the size of the network. Considering that, in general, the effort required to search for an item in a list grows logarithmically in relation to the size of the list, this is probably the best scaling that can be expected from a decentralized peer-to-peer system.

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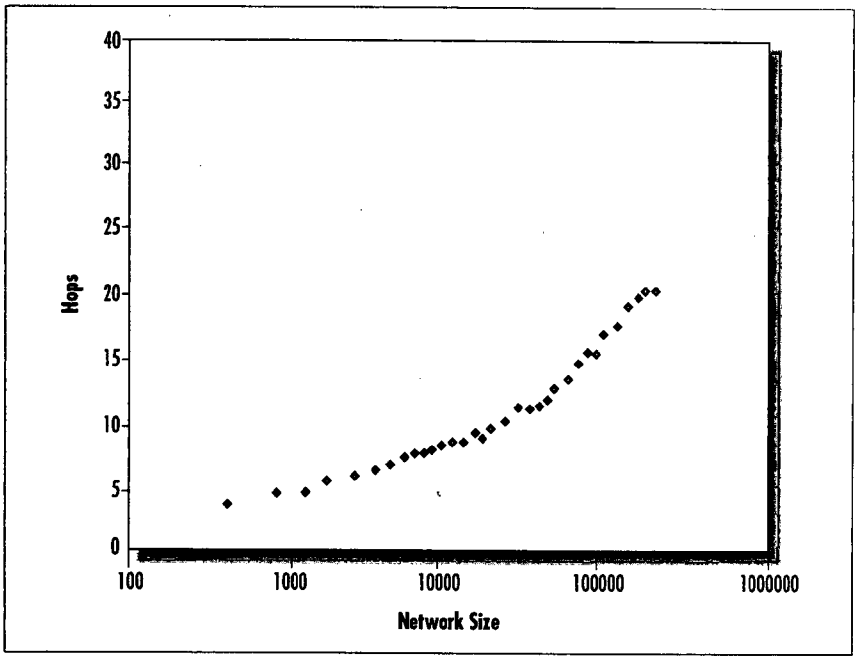

Figure 14-22. Median request pathlength vs. network size (logarithmic scale)

Case study 2: Gnutella

Gnutella uses a simple broadcast model to conduct queries, which does not invoke the small-world effect. Nonetheless, many of the concepts presented in this chapter can be taken as a useful framework for thinking about Gnutella's performance, which has been in the trade press so much recently.

In Gnutella, each peer tries to maintain a small minimum number (typically around three) of active simultaneous connections to other peers. These peers are selected from a locally maintained *host catcher* list containing the addresses of the other peers that this peer knows about. Peers can be discovered through a wide variety of mechanisms, such as watching for PING and PONG messages, noting the addresses of peers initiating queries, receiving incoming connections from previously unknown peers, or using out-of-band channels such as IRC and the Web. However, not all peers so discovered may accept new connections, since they may already have enough connections or be picky about the peers they will talk to. Establishing a good set of connections can in general be a somewhat haphazard process. Further, peers leaving the network will cause additional shuffling as the remaining peers try to replace lost connections.

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It therefore seems reasonable to model a Gnutella network by a random graph with a k of 3. Note that such a graph does not necessarily have exactly three edges per vertex. Rather, there will be some distribution in which the probability of finding a vertex having a given number of edges peaks around 3 and decreases exponentially with increasing numbers of edges. We will have more to say about this later.

Gnutella queries propagate through the network as follows. Upon receiving a new query, a peer broadcasts it to every peer that it is currently connected to, each of which in turn will broadcast the query to the peers *it* is connected to, and so on, in the manner of a chain letter. If a peer has a file that matches the query, it sends an answer back to the originating peer, but still forwards the query anyway. This process continues up to a maximum depth (or "search horizon") specified by the time-to-live field in the query. Essentially, Gnutella queries perform breadth-first searches on the network graph, in which searches broaden out and progressively cover the vertices closest to the starting point first. (By contrast, Freenet's style is closer to depth-first search, in which searches are directed deeper into the graph first.)

As before, it is necessary for the network graph to be connected, so that it is possible for any query to eventually reach some peer having the desired data. Achieving complete connectivity is somewhat more difficult than in Freenet because of the random nature of Gnutella connectivity. We can imagine that a random assignment of connections might leave some subset of peers cut off from the rest. However, in practice connectedness appears to hold.

Second, there must again be short routes between arbitrary peers, so that queries will be able to reach their targets before exceeding their depth limits. We turn to simulation to explore these properties.

Initial experiments

Suppose we create a network of 1,000 identical nodes initially sharing no files. To model its connectivity, let's add 1,500 edges by picking random nodes to be connected, two at a time, and creating edges between them. Topologically, the resulting network will be equivalent to a random graph in which n is 1,000 and k is 3.

Now let's add data to fill the network, since Gnutella does not have an explicit "insert" or "publish" mechanism. To make this simulation broadly comparable to the Freenet simulation, we'll randomly generate data items to be stored on 20 nodes each (the equivalent of a Freenet insert with hops-to-live 20). This can be imagined as 20 users independently choosing to share the same file, perhaps a

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particular MP3. We set the number of different data items added to be the same as the number inserted over the course of the Freenet simulation—that is, about 2,500.

As before, we simulate a simple network usage model. Since a Gnutella network does not evolve organically over time the way a Freenet network does, a single set of probe measurements should suffice. Following our previous method, we perform 300 queries from randomly chosen nodes in the network. The keys requested are chosen randomly from those known to be stored in the network, and the time-to-live is set to infinity, so these queries will always succeed eventually. To gauge the distance a query must travel before finding data, we stop the query as soon as a hit is found and note the number of hops taken to that point. (In the real Gnutella, queries proceed in parallel on a large number of nodes, so it is not practicable to halt them after finding a match on one node.) Figure 14-23 shows the resulting distribution of query pathlengths.

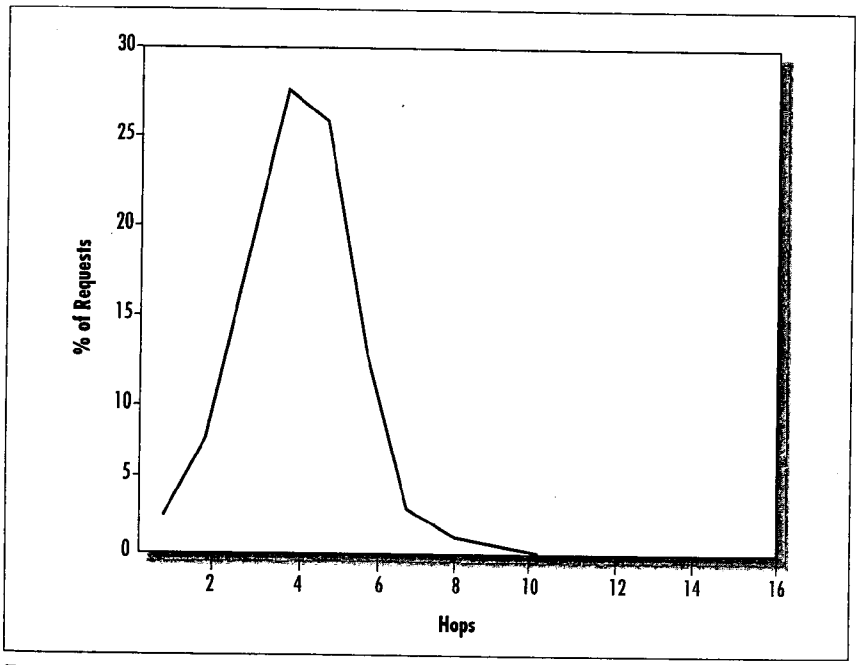

Figure 14-23. Distribution of query pathlengths in Gnutella

We see that Gnutella queries are satisfied extremely quickly, under both average-case and worst-case conditions. Indeed, the breadth-first search guarantees that the optimal shortest path to the data will always be found, making the query pathlength equal to the characteristic pathlength. However, this is not a true

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measure of the effort expended by the network as a whole, since queries are broadcast to so many nodes. A better measure is to consider the number of nodes contacted in the course of a query, as shown in Figure 14-24.

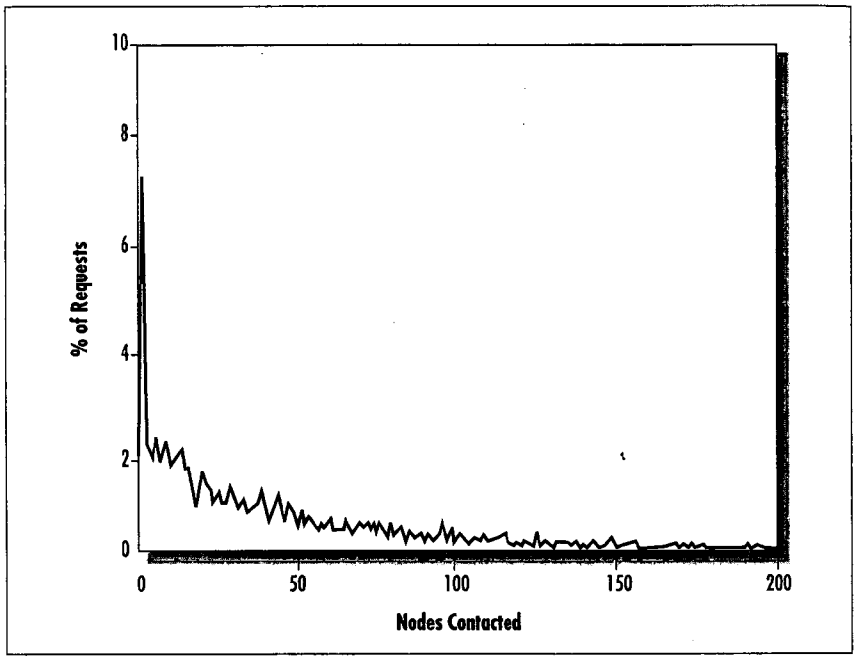

Figure 14-24. Distribution of the number of nodes contacted per query

A significant number of queries require the participation of 50 nodes, and many even call for 100 or more. It is apparent that the price paid for a quick result is a large expenditure of effort to exhaustively search a significant proportion of the network. Vis-à-vis Freenet, Gnutella makes a trade-off of much greater search effort in return for optimal paths and better worst-case performance.

Fault tolerance and link distribution in Gnutella

What are Gnutella's fault-tolerance characteristics? As before, we can consider its behavior under two node failure scenarios: random failure and targeted attack. The distribution of links in Freenet was an important factor in its robustness, so let's look at Gnutella's corresponding distribution, shown in Figure 14-25.

Mathematically, this is a "Poisson" distribution peaked around the average connectivity of 3. Its tail drops off exponentially, rather than according to a power law as Freenet's does. This can be seen more clearly in the log-log plot of Figure 14-26.

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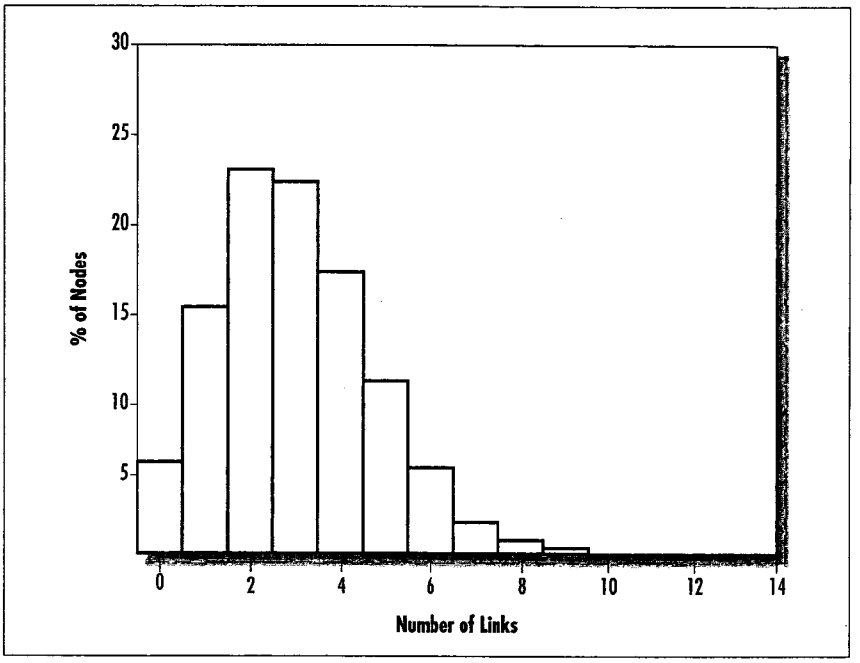

Figure 14-25. Histogram showing the distribution of links in Gnutella

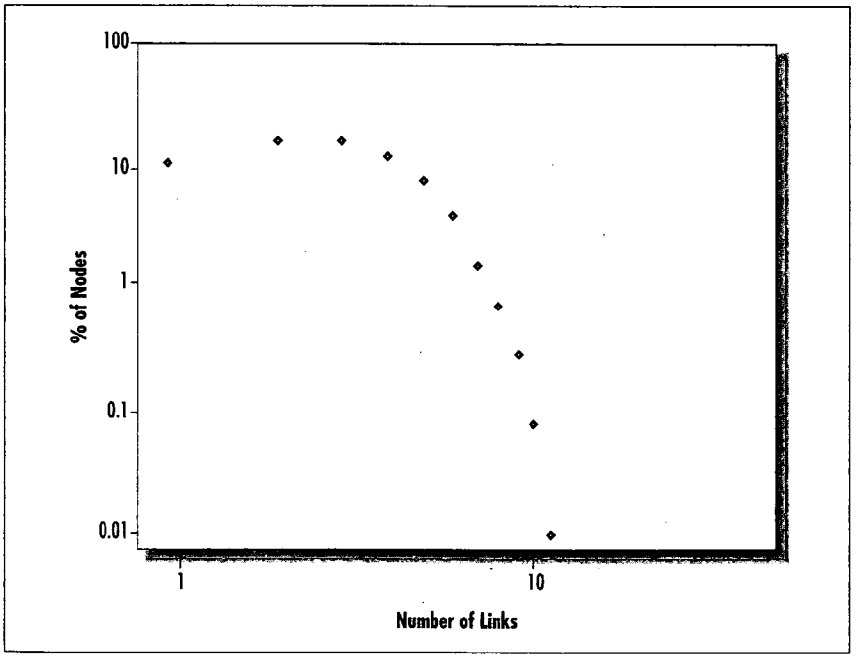

Figure 14-26. Log-log scatter plot of the distribution of links in Gnutella

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Comparing this plot to Figure 14-20, we can see that Figure 14-26 drops off much more sharply at high link numbers. As a result, highly connected nodes are much less of a factor in Gnutella than they are in Freenet.

Let's see how Gnutella behaves under the targeted attack scenario, in which the most-connected nodes are removed first. Figure 14-27 shows the number of nodes contacted per query (as a percentage of the surviving nodes) versus the percentage of nodes attacked. (A request that fails is treated as a value of 100%.) If we compare this plot to Figure 14-18, we can see that Gnutella resists targeted attack better than Freenet does, since the highly connected nodes play less of a role.

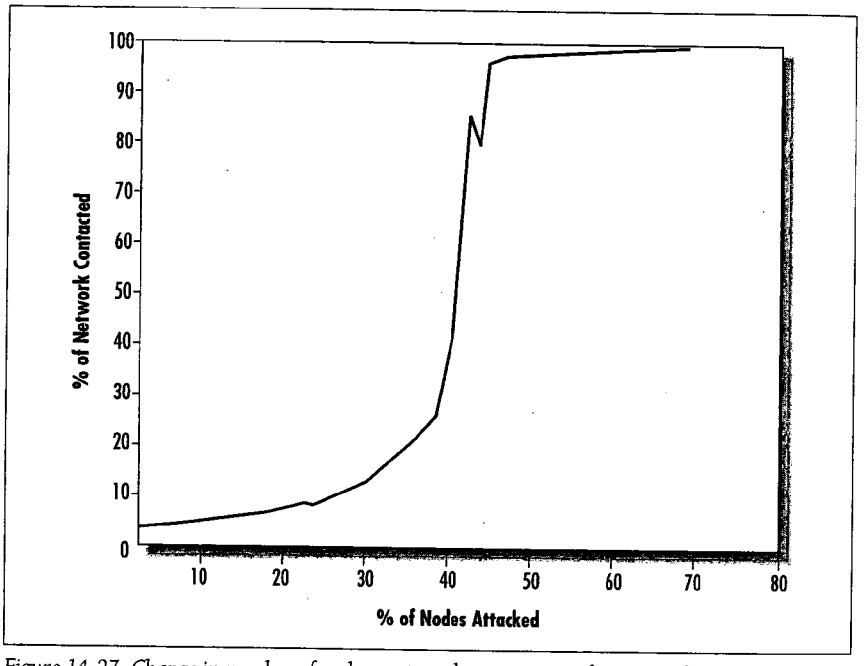

Figure 14-27. Change in number of nodes contacted per query, under targeted attack

On the other hand, the random failure scenario is the opposite. Figure 14-28 shows the number of nodes contacted versus the percentage of nodes failing. If we compare this to Figure 14-16, Freenet does better.

In fact, this occurs because Gnutella performs about the same under both random failure and targeted attack, as can be seen more clearly in Figure 14-29. Here again is a trade-off: Gnutella responds equally to failure and attack, since all of its nodes are roughly equivalent. Freenet's highly connected nodes enable it to better cope with random failure, but these then become points of vulnerability for targeted attack.

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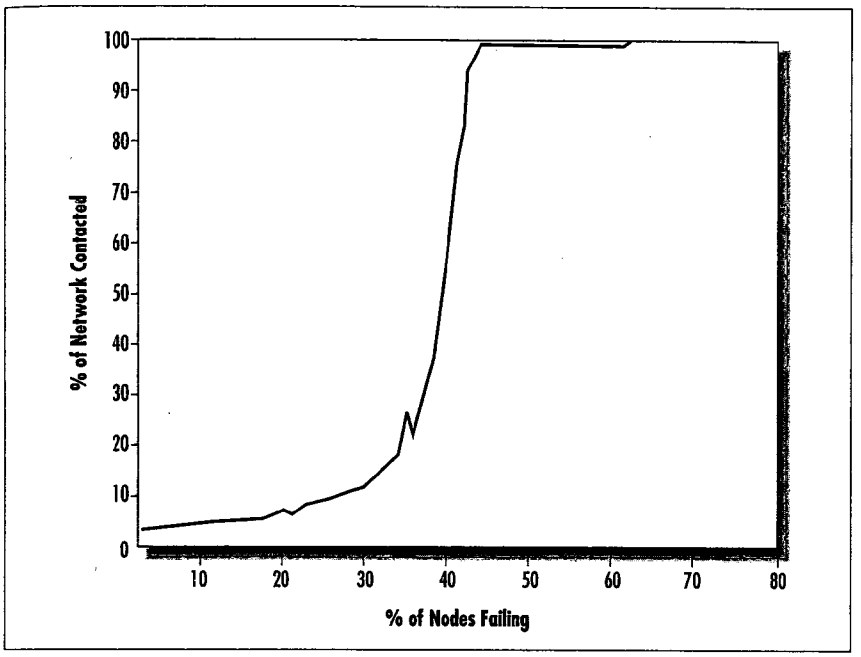

Figure 14-28. Change in number of nodes contacted per query, under random failure

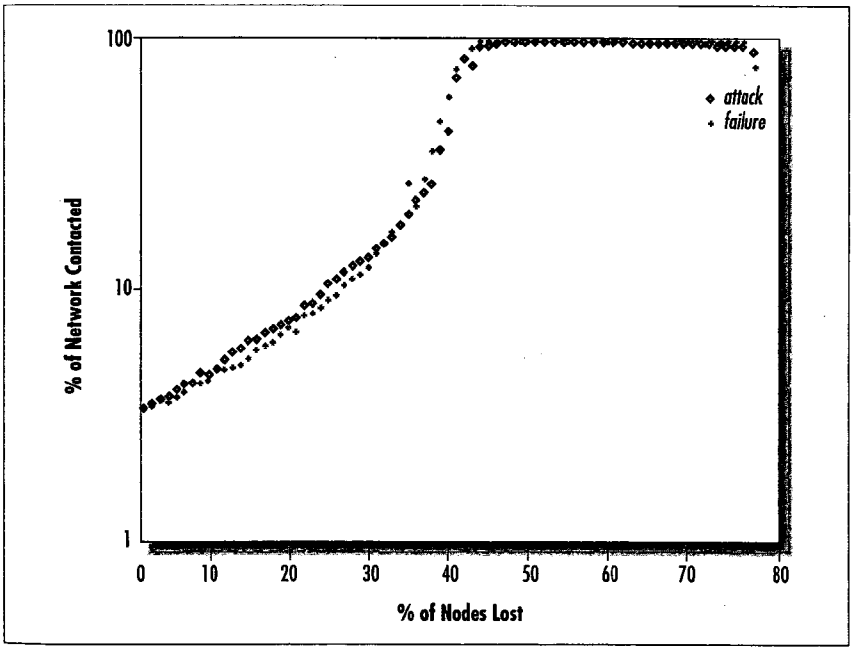

Figure 14-29. Comparison of the effects of attack and failure

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This is brought out in more detail by Figure 14-30, which plots the four scenarios together using an arbitrary scale. We can see that the Freenet failure curve grows much more slowly than the Gnutella curves, while the Freenet attack curve shoots up sooner.

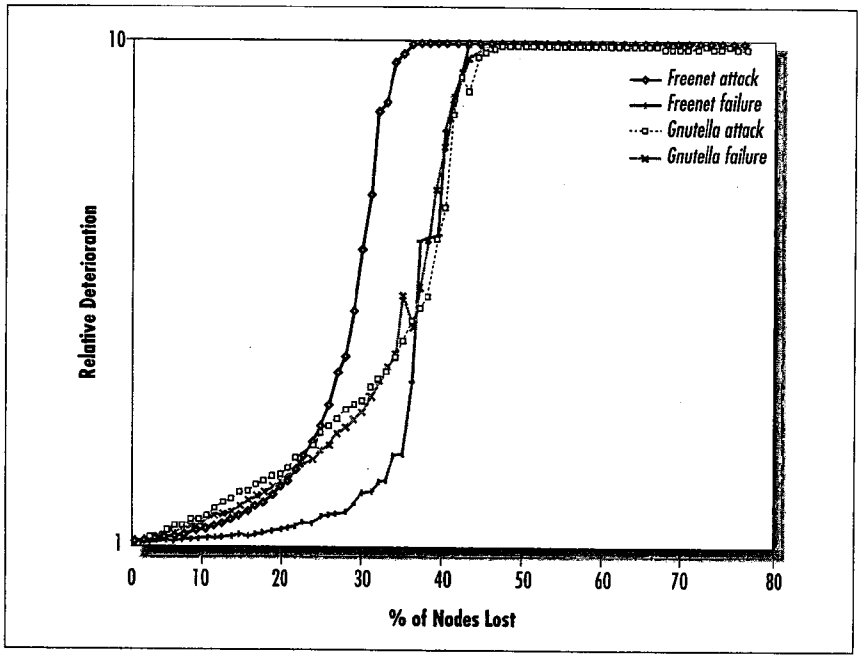

Figure 14-30. Comparison of attack and failure nodes in Freenet and Gnutella

The impact of free riding

Free riding in Gnutella is of more than merely theoretical interest, as indicated by the Xerox PARC paper mentioned earlier. Gnutella is vulnerable to free riders because its peers do not maintain any state information about other peers, so they cannot distinguish free riding from non–free riding peers. In particular, free riding peers will still have queries sent to them even if they never answer any. The presence of free riders will thus "dilute" the network, making queries travel farther before finding data. This can cause queries to fail if the desired data is pushed beyond the search horizon.

Ironically, it may be better for the network if free riding peers drop queries altogether instead of forwarding them, since queries will then simply flow around the free riders (unless portions of the network are completely cut off, of course). This is the opposite of the Freenet situation: Freenet free riders that drop que-

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ries are harmful since they kill off those queries, but those that forward queries unanswered actually help the network to route around them later on by propagating information about downstream peers.

Scalability

Finally, let's consider Gnutella's scalability. As a random graph, its characteristic pathlength scales logarithmically with the size of the network. Since its breadth-first search finds optimal paths, the request pathlength always equals the characteristic pathlength and also scales logarithmically. We have already seen that these pathlengths are quite low, so the amount of time taken by queries should be manageable up to very large network sizes. This does not accurately reflect their bandwidth usage, however.

The bandwidth used by a query is proportional to the number of messages sent, which in turn is proportional to the number of nodes that must be contacted before finding data. Actually, this is an underestimate, since many nodes will be sent the same query more than once and queries continue after finding data. Figure 14-31 shows the median number of nodes contacted per query versus network size, up to 200,000 nodes.

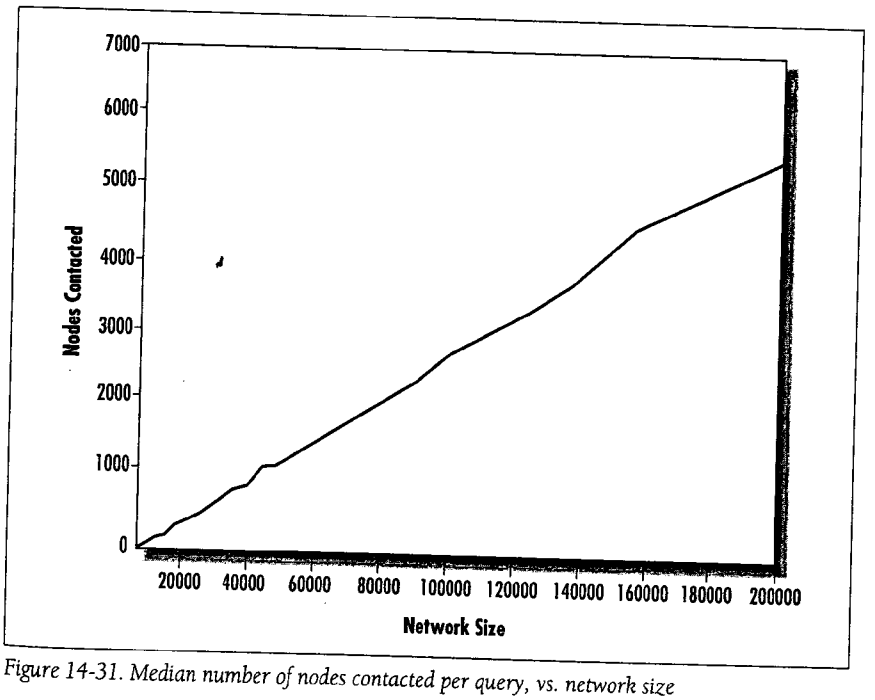

Figure 14-31. Median number of nodes contacted per query, vs. network size

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We can see that the number of nodes that must be contacted scales essentially linearly, meaning that every doubling of network size will also double the bandwidth needed per query. An alternate way of looking at this is to see that if bandwidth usage is kept lower by limiting search depths, success rates will drop since queries will not be able to reach the data. This may pose a serious scalability problem.

One solution already being explored is to modify Gnutella from a pure decentralized peer-to-peer model to a partly hierarchical model by using *super peers*. These are special peers that act as aggregators for other peers located "behind" them in the manner of a firewall. Super peers maintain indices of all the files their subordinate peers are sharing, and appear to the rest of the network as though they were sharing those files themselves. When queried for a file, a super peer can route the query directly to the relevant peer without a broadcast. In addition, if one of its subordinates requests a file held by another subordinate, it can satisfy the request immediately without involving the wider network. Super peers thus reduce the effective size of the network by replacing a group of ordinary peers with a single super peer.

From there, it is a short step to imagine "super-super peers" that aggregate queries for super peers, "super-super-super peers," and so on. Taken to the extreme, this could yield a completely hierarchical search tree like DNS. Such an arrangement would place each peer in successively larger aggregate groups, ultimately ending in a root peer managing the entire network. Searches in such a tree would scale logarithmically; however, it implies a considerable loss of the autonomy promised by peer-to-peer.

Conclusions

Performance is likely to remain an important issue in peer-to-peer systems design well into the foreseeable future. Within the peer-to-peer model, a number of trade-offs can be used to tailor different sets of performance outcomes. Freenet, for example, emphasizes high scalability and efficient searches under average conditions while sacrificing worse-case performance. At the other end of the spectrum, Gnutella sacrifices efficiency for faster searches and better worst-case guarantees. Ideas drawn from graph theory and the small-world model can help to quantify these trade-offs and to analyze systems in concrete terms.

Fault tolerance and free riding are additional challenges to deal with, and here again we can see different approaches. Systems like Freenet that develop specialized nodes can improve their robustness under random failure, but more uniform systems like Gnutella can better cope with targeted attacks. Free riding, a

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different type of failure mode, needs to be addressed in terms of routing around or otherwise neutralizing uncooperative nodes.

Last but not least, scalability is a crucial concern for systems that hope to make the leap from conceptual demonstration to world-wide usage. For systems that do not inherently scale well, a further set of trade-offs can allow better scalability through a move toward a hierarchical peer-to-peer model, though at the expense of local autonomy.

The peer-to-peer model encompasses a diverse set of approaches. By recognizing the wide range of possibilities available, inventing new ideas and new combinations, and using analytical methods to evaluate their behaviors, system designers will be well-equipped to exploit the power of peer-to-peer.

Acknowledgments

I would like to thank Catherine Reive and the Marshall Aid Commemoration Commission for their help and support. Many thanks also to Ian Clarke for creating Freenet, to *Need to Know* for bringing it to my attention, to David Molnar for prompting a submission to the ICSI Workshop on Design Issues in Anonymity and Unobservability, and to Sammy Klein, Matt Bruce, and others for reviewing early drafts. Jon Kleinberg's influence in serendipitously publishing an article on navigation in small-world networks the very week I started writing this was invaluable. Thanks to my thesis advisor, Professor Keith Clark, for putting up with my extracurricular activities. Finally, big thanks go to Andy Oram for putting the whole thing together.

Microsoft Exhibit 1019

chapter fifteen

Trust

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Trust plays a central role in many aspects of computing, especially those related to network use. Whether downloading and installing software, buying a product from a web site, or just surfing the Web, an individual is faced with trust issues. Does this piece of software really do what it says it does? Will the company I make purchases from sell my private information to other companies? Is my ISP logging all of my network traffic? All of these questions are central to the trust issue. In this chapter we discuss the areas of trust related to distributed systems—computers that communicate over a network and share information.

Trust in peer-to-peer, collaborative, or distributed systems presents its own challenges. Some systems, like Publius, deliberately disguise the source of data; all of the systems use computations or files provided by far-flung individuals who would be difficult to contact if something goes wrong—much less to hold responsible for any damage done.

Trust in real life, and its lessons for computer networks

In the physical world, when we talk about how much we trust someone, we often consider that person's reputation. We usually are willing to put great trust in someone whom we have personally observed to be highly capable and have a high level of integrity. In the absence of personal observation, the recommendation of a trusted friend can lead one to trust someone. When looking for someone to provide a service for us in which trust is an issue—whether a doctor, baby-sitter, or barber—we often ask our friends for recommendations or check with a trusted authority such as a childcare referral service. In real life, *trust is often increased by establishing positive reputations and networks for conveying these*

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reputations. This is true for computer networks as well. More will be said about this in Chapter 16, *Accountability*, and Chapter 17, *Reputation*. Once reputations are established, digital certificates and networks like the PGP web of trust (described later in this chapter) can be used to convey reputation information in a trustworthy way.

Information conveyed by a trusted person is itself seen to be trustworthy, especially if it is based on a personal observation by that person. Information conveyed via a long chain of people, trustworthy or not, is generally viewed to be less trustworthy—even if the entire chain is trustworthy, the fact that it was conveyed via a long chain introduces the possibility that somewhere along the way the facts may have been confused. So we may trust all of the people in the chain to be honest, but we may not trust all of them to accurately remember and convey every detail of a story told by someone else. If it is important to us to have confidence in the veracity of a piece of information, we may try to follow the chain back to its source. *Essentially, we are able to increase trust by reducing the number of people that must be trusted.* This applies to real life situations as well as computer networks.

We can also look at trust from a risk assessment perspective. We tend to be more willing to place trust in people when the risk of adverse consequences should our trust be misplaced is small. Likewise, even if there is high risk, if the potential consequences are not that bad, we may still be willing to trust. *Thus, we can also increase trust by reducing risk.* Just as in real life we may reduce risk by removing valuable items from a car before leaving it with a parking lot attendant, in a networked environment we may reduce risk by creating protected "sandboxes" where we can execute untrusted code without exposing critical systems to danger.

Sometimes we interact with people whose reputations are unknown to us, but they somehow seem worthy of our trust. We may talk about people who seem to have an honest face or a trustworthy demeanor. In the online world, a web site that looks very professional may also appear to be trustworthy, even if we do not know anything else about it. Clearly there are ways of changing perceptions about trust when our later experiences conflict with our first impressions. Indeed, many companies spend a lot of effort honing a marketing message or corporate image in an effort to convey more of an image of trustworthiness to consumers.

In this chapter, we are concerned less about perceptions of trustworthiness than we are about designing systems that rely as little as possible on trust. Ultimately, we would like to design systems that do not require anyone to trust any

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aspect of the system, because there is no uncertainty about how the system will behave. In this context, the ideal trusted system is one that everyone has confidence in because they do not have to trust it. Where that is impossible, we use techniques such as reputation building and risk reduction to build trust.

We first examine the issue of downloaded software. For the most part, the software described in this book can be downloaded from the Web. This simple act of downloading software and running it on your computer involves many trust-related issues. We then examine anonymous publishing systems. This discussion uses Publius as an example, but many of these issues apply to other systems as well. The last part of the chapter examines the trust issues involved in file sharing and search engines.

Trusting downloaded software

Trust issues exist even before an individual connects her computer to any network. Installing software supplied with your computer or that perhaps was bought in a retail store implies a level of trust—you trust that the software will work in the manner described and that it won't do anything malicious. By purchasing it from a "reputable" company, you believe that you know who wrote the software and what kind of reputation is associated with their software products. In addition, you may be able to take legal action if something goes horribly wrong.

The advent of the Internet changes this model. Now software can be downloaded directly onto your computer. You may not know who the author is and whether the software has been maliciously modified or really does what it claims. We have all heard stories of an individual receiving an attachment via email that, when executed, deleted files on the victim's hard drive.

Ideally, when downloading software from the Internet, we would like to have the same assurances that we have when we purchase the software directly from a store. One might think that simply downloading software from companies that one is already familiar with raises no trust issues. However, you can see from the sampling of potential problems in Table 15-1 this is really not true. The software you are downloading may have been modified by a malicious party before you even begin downloading it. Even if it begins its journey unmodified, it has to travel to you over an untrusted network—the Internet. The software, while it is traveling on the network, can be intercepted, modified, and then forwarded to you—all without your knowledge. Even if this doesn't happen, your Internet service provider (ISP) or another party could be logging the fact that you are downloading a particular piece of software or visiting a particular web site. This

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information can, for example, be used to target specific advertising at you. At the very least, this logging is an invasion of privacy. As we shall see there are ways of overcoming each of these problems.*

Table 15-1. Trust issues when downloading software

Risk	Solution	Trust principle
Software doesn't behave as advertised, and may even damage your computer system.	Only download software from companies or individuals who have established a good reputation, or those you know where to find should a problem occur.	Look for positive reputations.
Software is modified (on server or in transit).	Check for digital signature on message digest and verify signature against author's certificate.	Use tools that accurately convey reputations.
Your downloads (and other online behavior) are logged by your ISP or other parties.	Use an anonymity tool so other parties do not get access to information that might link you to a particular download.	Reduce risk.

Message digest functions

Almost all of the software described in this book is given away for free. The only way to acquire it is to download it—you can't walk into your local computer store and purchase it. We would like some way to verify that the downloaded files have not been tampered with in any way. This can be accomplished through the use of a *message digest function*, which is also known as a cryptographically secure hash function. A message digest function takes a variable-length input message and produces a fixed-length output. The same message will always produce the same output. If the input message is changed in any way, the digest function produces a different output value. This feature makes digest functions ideal for detecting file tampering.

Now that we have message digest functions, it looks like all of our tamper problems are solved—the author of a piece of software just places the value of the file's hash on the same web page that contains the file download link. After the user downloads the file, a separate program finds the digest of the file. This digest is then compared with the one on the web page. If the digests don't match the file has been tampered with; otherwise it is unchanged. Unfortunately things are not that simple. How do we know that the digest given on the web page is correct? Perhaps the server administrator or some malicious hacker changed the

* For a more comprehensive discussion of this topic, see Bruce Schneier (1999), *Secrets and Lies: Digital Security in a Networked World*, John Wiley & Sons.

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software and placed the digest of the modified file on the web page. If someone downloaded the altered file and checked the replaced digest everything would look fine. The problem is that we do not have a mechanism to guarantee that the author of the file was the one who generated the particular digest. What we need is some way for the author to state the digest value so that someone else cannot change it.

Digital signatures

Public key cryptography and digital signatures can be used to help identify the author of a file. Although the mathematics behind public key cryptography are beyond the scope of this book, suffice it to say that a pair of keys can be generated in such a way that if one key is used to sign some piece of data, the other key can be used to verify the signed data. Keys are essentially large numbers that are needed for the signature and verification operations. One of these keys is kept secret and is therefore called the private key. The other key is made available to everyone and is called the public key. Someone can send you an authenticated message simply by signing the message with his private key. You can then use his public key to verify the signature on the message.

So it looks like our problem is almost solved. The author of the software generates a public and private key. The author then computes the digest of the software package. This digest is then signed using a private key. A file containing the signed digest is placed on the same web page as the file to be downloaded (the software package). After downloading the software an individual finds its digest. The signed digest file is downloaded from the web site and verified using the author's public key.

Digital certificates

The problem with the scheme is that we have no way of verifying the author's public key. How do we know that someone didn't just generate a public/private key pair, modify the file, and sign its digest with the private key just generated? The public key on the web site cannot necessarily be trusted. We need a way to certify that a particular public key does indeed belong to the author of the software. Digital certificates are meant to provide this binding of public keys to individuals or organizations.

Digital certificates are issued by companies called certifying authorities (CAs). These are organizations that mint digital certificates for a fee; they are often called *trusted third parties* because both you and your correspondent trust them. An individual or corporation requesting a certificate must supply the CA with

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the proper credentials. Once these credentials have been verified, the CA mints a new certificate in the name of the individual or corporation. The CA signs the certificate with its private key and this signature becomes part of the certificate. The CA signature guarantees its authenticity.

The certificate creation process just described is a simplification of the actual process. Different classes of certificates exist corresponding to the type of credentials presented when applying for the certificate. The more convincing the credentials, the more verification work is created for the CA, and therefore it assesses a higher annual fee on the individual or corporation applying for the certificate. Therefore, certain types of certificates are more trustworthy than others.

Signature verification

Now all the pieces are in place. The author of some software applies to a CA for a certificate. This certificate binds her to a public key—only she knows the associated private key. She signs her software using the method described above. The signed digest and a link to the software are placed on the author's web page. In addition, a link to the author's certificate is added to the web page. At some later time, an individual downloads the software and author's certificate. The digest function is performed on the file. The author's certificate is verified using the CA's public key, which is available on the CA's web page. Once verified, the author's public key is used to verify the signature on the digest. This digest is compared to the one just performed on the file. If the digests match, the file has not been tampered with. See Figure 15-1 for an illustration of the process.

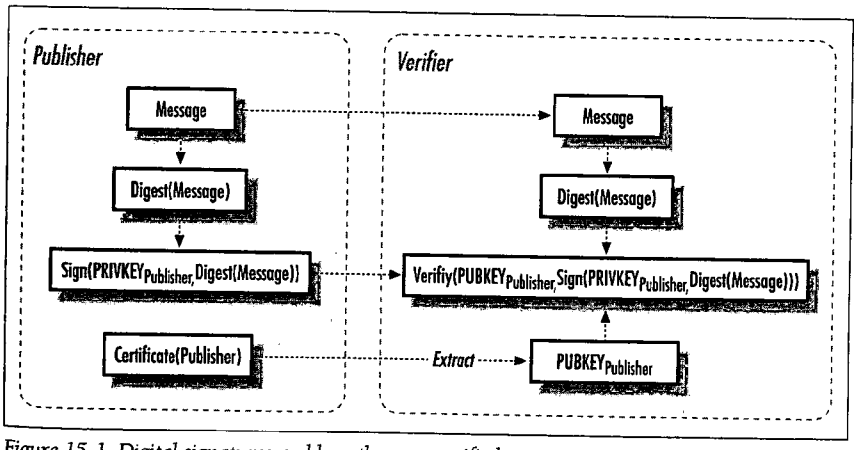

Figure 15-1. Digital signatures and how they are verified

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This verification process provides assurance that the downloaded software is signed by someone who has a private key that was issued to a software author with a particular name. Of course, there is no guarantee that the software author did not let someone else use her key, or that the key was not stolen without her knowledge. Furthermore, if we don't know anything about the reputation of this particular software author, knowing her name may not give us any confidence in her software (although if we have confidence in the CA, we may at least believe that it might be possible to track her down later should her software prove destructive).

The previously described verification process is not performed by hand. A number of software products are available that automate the task.

Pretty Good Privacy (PGP) is a well-known tool for encrypting files and email. It also allows individuals to sign and verify files. Rather than having to trust a third party, the CA, PGP allows individuals to create their own certificates. These certificates by themselves are not very helpful when trying to verify someone's identity; however, other people can sign the certificates. People that know you can sign your certificate, and you in turn can sign their certificates. If you receive a certificate from someone you don't trust, you can check the signatures on the certificate and see if you trust any of them. Based on this information you can decide if you wish to trust the certificate. This is a trust system based on intermediaries, and it forms what is called the "web of trust." The web of trust can be thought of as a peer-to-peer certification system. No centralized certifying authority is needed. A free version of PGP is available for download at *http://web.mit.edu/network/pgp.html*.

Unfortunately, digital certificates and signatures don't solve all of our problems. Not all software packages are signed. An author's private key can become compromised, allowing others to sign any piece of software with the compromised key. Just because software is signed doesn't mean that it doesn't have malicious intent. So one must still be vigilant when it comes to downloaded software.

Open source software

Much of the software available for download is available as source code, which needs to be compiled or interpreted in order to run on a specific computer. This means that one can examine the source code for any malicious intent. However, this is really practical only for rather small programs. This naturally leads to the question of whether it is possible to write a program that examines the source code of another program to determine if that program really does what it claims to do. Unfortunately, computer scientists have shown that, in general, it is

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impossible to determine if a program does what it claims to do.* However, we can build programs to monitor or constrain the behavior of other programs.

Sandboxing and wrappers

Programs that place limits on the behavior of other programs existed before the Internet. The most obvious example of this type of program is an operating system such as Unix or Windows NT. Such an operating system, for example, won't allow you to delete a file owned by someone else or read a file owned by another user unless that user has granted you permission. Today, programs exist that can constrain the behavior of programs you download while surfing the Web. When a web page contains a Java applet, that applet is downloaded and interpreted by another program running on your computer. This interpreter prevents the applet from performing operations that could possibly damage your computer, such as deleting files. The term used to describe the process of limiting the type of operations a program can perform is called sandboxing. The applet or other suspicious program is allowed to execute only in a small sandbox. Thus the risk of damage is reduced substantially. Programs called wrappers allow the behavior of CGI scripts to be constrained in a similar manner.

Trust in censorship-resistant publishing systems

We now examine trust issues specific to distributed file-sharing and publishing programs. We use Publius as an example; however, the problems and solutions discussed are applicable to many of the other programs discussed in this book.

Publius is a web-based publishing system that allows people to publish documents in such a way that they are resistant to censorship. For a full description of Publius, see Chapter 11.

Publius derives its censorship resistance in part from a collection of independently owned web servers. Each server donates a few hundred megabytes of disk space and runs a CGI script that allows it to store and retrieve Publius files. Since each server is independently owned, the server administrator has free rein over the server. This means the administrator can arbitrarily read, delete or modify any files on the server including the Publius files. Because the Publius

* Proving certain properties of programs can be reduced to proving the "Halting Problem." See, for example, Michael Sipser (1997). *Introduction to the Theory of Computation.* PWS Publishing Company.

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files are encrypted, reading a file does not reveal anything interesting about the file (unless the server administrator knows the special Publius URL).

Publius in a nutshell

Before describing the trust issues involved in Publius, we briefly review the Publius publication process. When an individual publishes a file, the Publius client software generates a key that is used to encrypt the file. This key is split into a number of pieces called shares. Only a small number of these shares are required to reconstruct the key. For example, the key can be split into 30 shares such that any 3 shares are needed to reconstruct the document.

A large number of Publius servers—let's say 20—then store the file, each server taking one share of the key along with a complete copy of the encrypted file. Each server stores a different share, and no server holds more than one share. A special Publius URL is generated that encodes the location of the encrypted file and shares on the 20 servers. In order to read the document, the client software parses the special URL, randomly picks 3 of these 20 servers, and downloads the share stored on each of them. In addition, the client software downloads one copy of the encrypted file from one of the servers.

The client now combines the shares to form the key and uses the key to decrypt the file. A tamper check is performed to see if the file was changed in any way. If the file was changed, a new set of three shares and a new encrypted document are retrieved and tested. This continues until a file passes the tamper check or the system runs out of different encrypted file and share combinations.

Risks involved in web server logging

Most web servers keep a log of all files that have been requested from the server. These logs usually include the date, time, and the name of the file that was requested. In addition, these logs usually hold the IP address of the computer that made the request. This IP address can be considered a form of identification. While it may be difficult to directly link an individual to a particular IP address, it is not impossible.

Even if your IP address doesn't directly identify you, it certainly gives some information about you. For example, an IP address owned by an ISP appearing in some web server log indicates that an individual who uses that ISP visited the web site on a certain date and time. The ISP itself may keep logs as to who was using a particular IP address during a particular date and time. So while it may not be possible to directly link an individual to a web site visit, an indirect route may exist.

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Web servers almost always log traffic for benign reasons. The company or individual who owns the server simply wishes to get an idea how many requests the web server is receiving. The logs may answer questions central to the company's business. However, as previously stated, these logs can also be used to identify someone. This is a problem faced by Publius and many of the other systems described in this book.

Why would someone want to be anonymous on the Internet? Well, suppose that you are working for a company that is polluting the environment by dumping toxic waste in a local river. You are outraged but know that if you say anything you will be fired from your job. Therefore you secretly create a web page documenting the abuses of the corporation. You then decide you want to publish this page with Publius. Publishing this page from your home computer could unwittingly identify you. Perhaps one or more of the Publius servers are run by friends of the very corporation that you are going to expose for its misdeeds. Those servers are logging IP addresses of all computers that store or read Publius documents. In order to avoid this possibility you can walk into a local cyber café or perhaps the local library and use their Internet connection to publish the web page with Publius. Now the IP address of the library or cyber café will be stored in the logs of the Publius servers. Therefore there is no longer a connection to your computer. This level of anonymity is still not as great as we would like. If you are one of a very few employees of the company living in a small town, the company may be able to figure out you leaked the information just by tracing the web page to a location in that town.

Going to a cyber café or library is one option to protect your privacy. Anonymizing software is another. Depending on your trust of the anonymity provided by the cyber café or library versus your trust of the anonymity provided by software, you may reach different conclusions about which technique provides a higher level of anonymity in your particular situation. Whether surfing the Web or publishing a document with Publius, anonymizing software can help you protect your privacy by making it difficult, if not impossible, to identify you on the Internet. Different types of anonymizing software offer varying degrees of anonymity and privacy protection. We now describe several anonymizing and privacy-protection systems.

Anonymizing proxies

The simplest type of anonymizing software is an anonymizing proxy. Several such anonymizing proxies are available today for individuals who wish to surf the Web with some degree of anonymity. Two such anonymizing proxies are

Microsoft Exhibit 1019

Anonymizer.com and Rewebber.de. These anonymizing proxies work by acting as the intermediary between you and the web site you wish to visit. For example, suppose you wish to anonymously view the web page with the URL *http://www.oreilly.com*. Instead of entering this address into the browser, you first visit the anonymizing proxy site (e.g., *http://www.anonymizer.com*). This site displays a form that asks you to enter the URL of the site you wish to visit. You enter *http://www.oreilly.com*, and the anonymizing proxy retrieves the web page corresponding to this URL and displays it in your browser. In addition, the anonymizing proxy rewrites all the hyperlinks on the retrieved page so that when you click on any of these hyperlinks the request is routed through the anonymizing proxy. Any logs being kept by the server *www.oreilly.com* will only record the anonymizing proxy's IP address, as this is the computer that actually made the request for the web page. The process is illustrated in Figure 15-2.

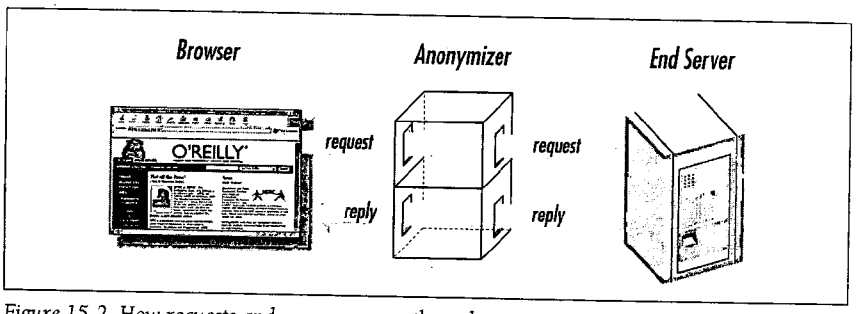

Figure 15-2. How requests and responses pass through an anonymizing proxy

The anonymizing proxy solves the problem of logging by the Publius servers but has introduced the problem of logging by the anonymizing proxy. In other words, if the people running the proxy are dishonest, they may try to use it to snare you.

In addition to concern over logging, one must also trust that the proxy properly transmits the request to the destination web server and that the correct document is being returned. For example, suppose you are using an anonymizing proxy and you decide to shop for a new computer. You enter the URL of your favorite computer company into the anonymizing proxy. The company running the anonymizing proxy examines the URL and notices that it is for a computer company. Instead of contacting the requested web site, the proxy contacts a competitor's web site and sends the content of the competitor's web page to your browser. If you are not very familiar with the company whose site you are visiting, you may not even realize this has happened. In general, if you use a proxy you must just resolve to trust it, so try to pick a proxy with a good reputation.

Microsoft Exhibit 1019

Censorship in Publius

Now that we have a possible solution to the logging problem, let's look at the censorship problem. Suppose that a Publius server administrator named Eve wishes to censor a particular Publius document. Eve happened to learn the Publius URL of the document and by coincidence her server is storing a copy of the encrypted document and a corresponding share. Eve can try a number of things to censor the document.

Upon inspecting the Publius URL for the document she wishes to censor, Eve learns that the encrypted document is stored on 20 servers and that 3 shares are needed to form the key that decrypts the document. After a bit of calculation Eve learns the names of the 19 other servers storing the encrypted document. Recall that Eve's server also holds a copy of the encrypted document and a corresponding share. If Eve simply deletes the encrypted document on her server she cannot censor the document, as it still exists on 19 other servers. Only one copy of the encrypted document and three shares are needed to read the document. If Eve can convince at least 17 other server administrators to delete the shares corresponding to the document then she can censor the document, as not enough shares will be available to form the key. (This possibility means that it is very difficult, but not impossible, to censor Publius documents. The small possibility of censorship can be viewed as a limitation of Publius. However, it can also be viewed as a "safety" feature that would allow a document to be censored if enough of the server operators agreed that it was objectionable.)

Using the Update mechanism to censor

Eve and her accomplices have not been able to censor the document by deleting it; however, they realize that they might have a chance to censor the document if they place an *update* file in the directory where the encrypted file and share once resided. The *update* file contains the Publius URL of a file published by Eve.

Using the Update file method described in Chapter 11, Eve and her accomplices have a chance, albeit a very slim one, of occasionally censoring the document. When the Publius client software is given a Publius URL it breaks up the URL to discover which servers are storing the encrypted document and shares. The client then randomly chooses three of these servers from which to retrieve the shares. The client also retrieves the encrypted document from one of these servers. If all three requests for the share return with the same *update* URL, instead of the share, the client follows the *update* URL and retrieves the corresponding document.

How successful can a spoofed update be? There are 1,140 ways to choose 3 servers from a set of 20. Only 1 of these 1,140 combinations leads to Eve's

Microsoft Exhibit 1019

document. Therefore Eve and her cohorts have only a 1 in 1,140 chance of censoring the document each time someone tries to retrieve it. Of course, Eve's probability of success grows as she enlists more Publius server administrators to participate in her scheme. Furthermore, if large numbers of people are trying to retrieve a document of some social significance, and they discover any discrepancies by comparing documents, Eve could succeed in casting doubt on the whole process of retrieval.

A publisher worried about this sort of update attack has the option of specifying that the file is not updateable. This option sets a flag in the Publius URL that tells the Publius client software to ignore *update* URLs sent from any Publius server. Any time the Publius client receives an *update* URL, it simply treats it as an invalid response from the server and attempts to acquire the needed information from another server. In addition to the "do not update" option, a "do not delete" option is available to the publisher of a Publius document. While this cannot stop Eve or any other server administrator from deleting files, it does protect the publisher from someone trying to repeatedly guess the correct password to the delete the file. This is accomplished by not storing a password with the encrypted file. Because no password is stored on the server, the Publius server software program refuses to perform the Delete command.

As previously stated, the Publius URL also encodes the number of shares required to form the key. This is the same as the number of *update* URLs that must match before the Publius client retrieves an *update* URL. Therefore, another way to make the update attack more difficult is to raise the number of shares needed to reconstruct the key. The default is three, but it can be set to any number during the Publish operation. However, raising this value increases the amount of time it takes to retrieve a Publius document because more shares need to be retrieved in order to form the key.

On the other hand, requiring a large number of shares to reconstruct the document can make it easier for an adversary to censor it. Previously we discussed the possibility of Eve censoring the document if she and two friends delete the encrypted document and its associated shares. We mentioned that such an attack would be unsuccessful because 17 other shares and encrypted documents exist. If the document was published in such a way that 18 shares were required to form the key, Eve would have succeeded in censoring the document because only 17 of the required 18 shares would be available. Therefore, some care must be taken when choosing the required number of shares.

Alternatively, even if we do not increase the number of shares necessary to reconstruct a Publius document, we could develop software for retrieving Publius documents that retrieves more than the minimum number of required

Microsoft Exhibit 1019

shares when an *update* file is discovered. While this slows down the process of retrieving updated documents, it can also provide additional assurance that a document has not been tampered with (or help the client find an unaltered version of a document that has been tampered with).

The attacks in this censorship section illustrate the problems that can occur when one blindly trusts a response from a server or peer. Responses can be carefully crafted to mislead the receiving party. In systems such as Publius, which lack any sort of trust or reputation mechanism, one of the few ways to try to overcome such problems is to utilize randomization and replication. By replication we mean that important information should be replicated widely so that the failure of one or a small number of components will not render the service inoperable (or, in the case of Publius, easy to censor). Randomization helps because it can make attacks on distributed systems more difficult. For example, if Publius always retrieved the first three shares from the first three servers in the Publius URL, then the previously described update attack would always succeed if Eve managed to add an *update* file to these three servers. By randomizing share retrieval the success of such an attack decreases from 100% to less than 1%.

Publius proxy volunteers

In order to perform any Publius operation one must use the Publius client software. The client software consists of an HTTP proxy that intercepts Publius commands and transparently handles non-Publius URLs as well. This HTTP proxy was designed so that many people could use it at once—just like a web server. This means that the proxy can be run on one computer on the Internet and others can connect to it. Individuals who run the proxy with the express purpose of allowing others to connect to it are called Publius proxy volunteers.

Why would someone elect to use a remote proxy rather than a local one? The current Publius proxy requires the computer language Perl and a cryptographic library package called Crypto++. Some individuals may have problems installing these software packages, and therefore the remote proxy provides an attractive alternative.

The problem with remote proxies is that the individual running the remote proxy must be trusted, as we stated in the "Anonymizing proxies" section earlier in this chapter. That individual has complete access to all data sent to the proxy. As a result, the remote proxy can log everything it is asked to publish, retrieve, update, or delete. Therefore, users may wish to use an anonymizing tool to access the Publius proxy.

The remote proxy, if altered by a malicious administrator, can also perform any sort of transformation on retrieved documents and can decide how to treat any

Microsoft Exhibit 1019

Publius commands it receives. The solutions to this problem are limited. Short of running your own proxy, probably the best thing you can do is use a second remote proxy to verify the actions of the first.

Third-party trust issues in Publius

Besides trusting the operators of the Publius servers and proxies, users of Publius may have to place trust in other parties. Fortunately some tools exist that reduce the amount of trust that must be placed in these parties.

Other anonymity tools

While not perfect, anonymizing proxies can hide your IP address from a Publius server or a particular web site. As previously stated, the anonymizing proxy itself could be keeping logs.

In addition, your Internet service provider (ISP) can monitor all messages you send over the Internet. An anonymizing proxy doesn't help us with this problem. Instead, we need some way of hiding all communication from the ISP. Cryptography helps us here. All traffic (messages) between you and another computer can be encrypted. Now the ISP sees only encrypted traffic, which looks like gibberish. The most popular method of encrypting web traffic is the Secure Sockets Layer (SSL) Protocol.

SSL

SSL allows two parties to create a private channel over the Internet. In our case this private channel can be between a Publius client and a server. All traffic to and from the Publius client and server can be encrypted. This hides everything from the ISP except the fact that you are talking to a Publius server. The ISP can see the encrypted channel setup messages between the Publius client and server. Is there a way to hide this piece of information too? It turns out there is.

Mix networks

Mix networks are systems for hiding both the content and destination of a particular message on the Internet.* One of the best-known mix networks is discussed in Chapter 7, *Mixmaster Remailers*.

* Mix networks were first introduced by David Chaum. See David Chaum (1981), "Untraceable Electronic Mail, Return Addresses, and Digital Pseudonyms," *Communications of the ACM*, vol. 24, no. 2, pp. 84–88.

Microsoft Exhibit 1019

A mix network consists of a collection of computers called routers that use a special layered encryption method to hide the content and true destination of a message. To send a message, the sender first decides on a path through a subset of the mixes. Each mix has an associated public and private key pair. All users of the mix network know all the public keys. The message is repeatedly encrypted using the public keys of the routers on the chosen path. First the message is encrypted with the public key of the last router in the chosen path. This encrypted message is then encrypted once again using the public key of the next-to-last router. This is repeated until the message is finally encrypted with the public key of the first router in the chosen path. As the encrypted message is received at each router, the outer layer of encryption is removed by decrypting it with the router's private key. This reveals only the next router in the mix network to receive the encrypted message. Each router can only decrypt the outer layer of encryption with its private key. Only the last router in the chosen path knows the ultimate destination of the message; however, it doesn't know where the message originated. The layers of encryption are represented in Figure 15-3.

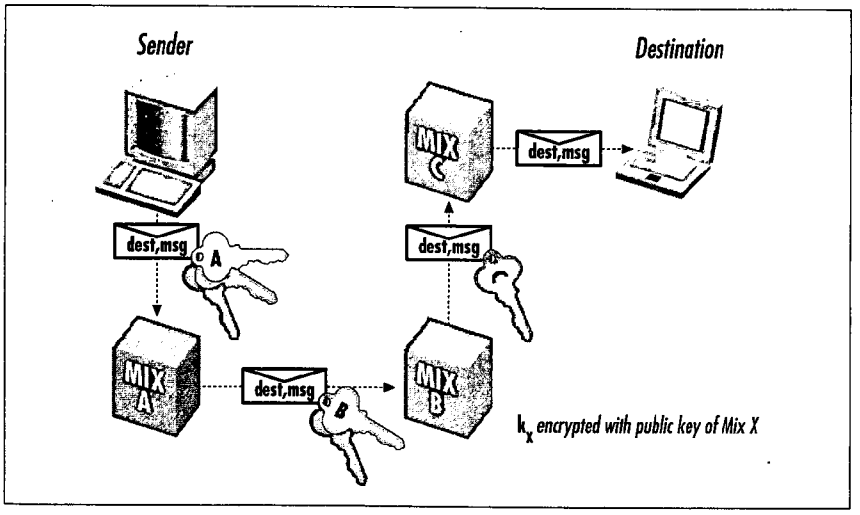

Figure 15-3. A mix network adds and strips off layers of encryption

Mix networks are also used to try to thwart traffic analysis. Traffic analysis is a method of correlating messages emanating from and arriving at various computers or routers. For instance, if a message leaves one node and is received by another shortly thereafter, and if the pattern is immediately repeated in the other direction, a monitor can guess that the two systems are engaged in a request and acknowledgment protocol. Even when a mix network is in use, this type of analysis is feasible if all or a large percentage of the mix network can be monitored by

Microsoft Exhibit 1019

an adversary (perhaps a large government). In an effort to combat this type of analysis, mix networks usually pad messages to a fixed length, buffer messages for later transmission, and generate fake traffic on the network, called covering traffic. All of these help to complicate or defeat traffic analysis.

Researchers at the U.S. Department of Defense developed an implementation of mix networks called Onion Routing (*http://www.onion-router.net*) and deployed a prototype network. The network was taken offline in January 2000. Zero-Knowledge Systems developed a commercial implementation of mix networks in a product called Freedom—see *http://www.freedom.net* for more information.

Crowds

Crowds is a system whose goals are similar to that of mix networks but whose implementation is quite different. Crowds is based on the idea that people can be anonymous when they blend into a crowd. As with mix networks, Crowds users need not trust a single third party in order to maintain their anonymity. A crowd consists of a group of web surfers all running the Crowds software. When one crowd member makes a URL request, the Crowds software on the corresponding computer randomly chooses between retrieving the requested document or forwarding the request to a randomly selected member of the crowd. The receiving crowd member can also retrieve the requested document or forward the request to a randomly selected member of the crowd, and so on. Eventually, the web document corresponding to the URL is retrieved by some member of the crowd and sent back to the crowd member that initiated the request.

Suppose that computers A, B, C, D, E, and F are all members of a crowd. Computer B wants to anonymously retrieve the web page at the URL *http://www.oreilly.com*. The Crowds software on computer B sends this URL to a random member of the crowd, say computer F. Computer F decides to send it to computer C. Computer C decides to retrieve the URL. Computer C sends the web page back to computer F. Computer F then sends the web page back to computer B. Notice that the document is sent back over the path of forwarding computers and not directly from C to B. All communication between crowd members is encrypted using symmetric ciphers. Only the actual request from computer C to *http://www.oreilly.com* remains unencrypted (because the software has to assume that *http://www.oreilly.com* is uninterested in going along with the crowd). The structure of the system is shown in Figure 15-4.

Notice that each computer in the crowd is equally likely to make the request for the specific web page. Even though computer C's IP address will appear in the log of the server *http://www.oreilly.com*, the individual using computer C can plausibly deny visiting the server. Computer C is a member of the crowd and

Microsoft Exhibit 1019

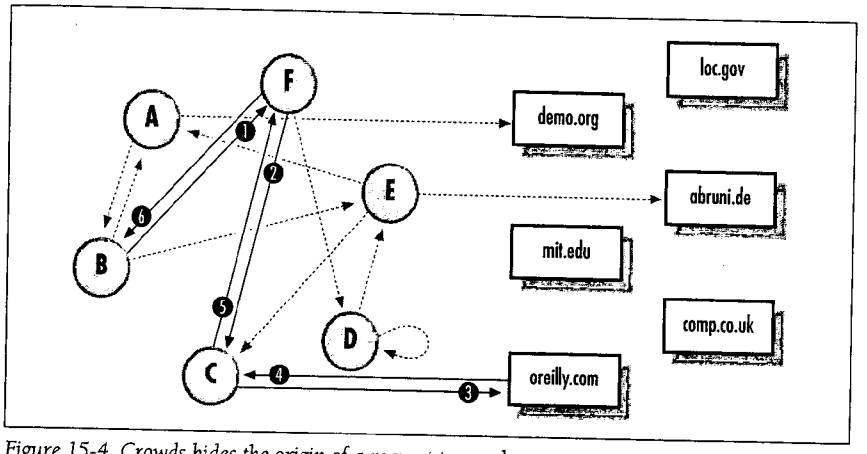

Figure 15-4. Crowds hides the origin of a request to a web server

therefore could have been retrieving the page for another member of the crowd. Notice that each crowd member cannot tell which other member of the crowd requested the particular URL. In the previous example, computer B sends the URL to computer F. Crowd member F cannot tell if the URL request originated with B or if B was simply an intermediary forwarding the request from another crowd member. This is the reason that the retrieved web page has to be passed back over the list of crowd members that forwarded the URL.

Crowds is itself an example of a peer-to-peer system.

Denial of service attacks

Publius relies on server volunteers to donate disk space so others can publish files in a censorship-resistant manner. Disk space, like all computer resources, is finite. Once all the disks on all the Publius servers are full, no more files can be published until others are deleted. Therefore an obvious attack on Publius is to fill up all the disks on the servers. Publius clients know the locations of all the servers, so identifying the servers to attack is a simple matter. Attacks with the intention of making resources unavailable are called denial of service attacks.

Systems that blindly trust users to conserve precious resources are extremely vulnerable to this kind of attack. Therefore, non–trust based mechanisms are needed to thwart such attacks.

Can systems be designed to prevent denial of service attacks? The initial version of Publius tried to do so by limiting the size of any file published with Publius to 100K. While this certainly won't prevent someone from trying to fill up the hard drives, it does make this kind of attack more time consuming. Other methods

Microsoft Exhibit 1019

such as CPU payment schemes, anonymous e-cash payment schemes, or quota systems based on IP address may be incorporated into future versions of Publius. While these methods can help deter denial of service attacks, they cannot prevent them completely.

Quota systems

Quota systems based on IP address could work as follows. Each Publius server keeps track of the IP address of each computer that makes a Publish request. If a Publius client has made more than ten Publish requests to a particular server in the last 24 hours, subsequent Publish requests will be denied by that server. Only after a 24-hour time period has elapsed will the server once again honor Publish requests from that Publius client's IP address.

The problem with this scheme is that it is not foolproof. An attacker can easily fake IP addresses. In addition, the 10-file limit may unfairly limit individuals whose IP addresses are dynamically assigned. For example, suppose someone with an IP address from AOL publishes ten files on some server. If later in the day someone else is assigned that same IP address, the individual will be unfairly excluded from publishing on that particular server.

CPU-based payment schemes

CPU-based payment schemes are used to help prevent denial of service attacks by making it expensive, in terms of time, to carry out such an attack. In Publius, for example, before the server agrees to publish a file, it could ask the publishing client to solve some sort of puzzle. The client spends some time solving the puzzle and then sends the answer to the server. The server agrees to publish the file only if the answer is correct. Each time the particular client asks to publish a file the server can make the puzzle a bit harder—requiring the client to expend more CPU time to find the puzzle answer.

While this scheme makes denial of service attacks more expensive, it clearly does not prevent them. A small Publius system created by civic-minded individuals could be overwhelmed by a large company or government willing to expend the computing resources to do the necessary calculations.

By design, Publius and many other publishing systems have no way of authenticating individuals who wish to publish documents. This commitment to anonymous publishing makes it almost impossible to stop denial of service attacks of this sort.

Microsoft Exhibit 1019

Anonymous e-cash payment schemes

Another way of preventing denial of service attacks is to require publishers to pay money in order to publish their documents with Publius. An anonymous e-cash system could allow publishers to pay while still remaining anonymous. Even if a well-funded attacker could afford to pay to fill up all available Publius servers, the fees collected from that attacker could be used to buy more disks. This could, of course, result in an arms race if the attacker had enough money to spend on defeating Publius. Chapter 16 discusses CPU- and anonymous e-cash–based payment schemes in more detail.

Legal and physical attacks

All of the methods of censorship described so far involve using a computer. However, another method of trying to censor a document is to use the legal system. Attackers may try to use intellectual property law, obscenity laws, hate speech laws, or other laws to try to force server operators to remove Publius documents from their servers or to shut their servers down completely. However, as mentioned previously, in order for this attack to work, a document would have to be removed from a sufficient number of servers. If the Publius servers in question are all located in the same legal jurisdiction, a single court order could effectively shut down all of the servers. By placing Publius servers in many different jurisdictions, such attacks can be prevented to some extent.

Another way to censor Publius documents is to learn the identity of the publishers and force them to remove their documents from the Publius servers. By making threats of physical harm or job loss, attackers may "convince" publishers to remove their documents. For this reason, it may be especially important for some publishers to take precautions to hide their identities when publishing Publius documents. Furthermore, publishers can indicate at the time of publication that their documents should never be deleted. In this case, no password exists that will allow the publishers to delete their documents—only the server operators can delete the documents.

Trust in other systems

We now examine issues of trust in some popular file-sharing and anonymous publishing systems.

Microsoft Exhibit 1019

Mojo Nation and Free Haven

Many of the publishing systems described in this book rely on a collection of independently owned servers that volunteer disk space. As disk space is a limited resource, it is important to protect it from abuse. CPU-based payment schemes and quotas, both of which we mentioned previously, are possible deterrents to denial of service attacks, but other methods exist.

Mojo Nation uses a digital currency system called Mojo that must be paid before one can publish a file on a server. In order to publish or retrieve files in the Mojo Nation network, one must pay a certain amount of Mojo.

Mojo is obtained by performing a useful function in the Mojo Nation network. For example, you can earn Mojo by volunteering to host Mojo content on your server. Another way of earning Mojo is to run a search engine on your server that allows others to search for files on the Mojo Nation network.

The Free Haven project utilizes a trust network. Servers agree to store a document based on the trust relationship that exists between the publisher and the particular server. Trust relationships are developed over time and violations of trust are broadcast to other servers in the Free Haven network. Free Haven is described in Chapter 12.

The Eternity Service

Publius, Free Haven, and Mojo Nation all rely on volunteer disk space to store documents. All of these systems have their roots in a theoretical publishing system called the Eternity Service.* In 1996, Ross Anderson of Cambridge University first proposed the Eternity Service as a server-based storage medium that is resistant to denial of service attacks.

An individual wishing to anonymously publish a document simply submits it to the Eternity Service with an appropriate fee. The Eternity Service then copies the document onto a random subset of servers participating in the service. Once submitted, a document cannot be removed from the service. Therefore, an author cannot be forced, even under threat, to delete a document published on the Eternity Service.

Anderson envisioned a system in which servers were spread all over the world, making the system resistant to legal attacks as well as natural disasters. The

* See Ross Anderson (1996), "The Eternity Service," *PragoCrypt'96*.

Microsoft Exhibit 1019

distributed nature of the Eternity Service would allow it to withstand the loss of a majority of the servers and still function properly.

Anderson outlined the design of this ambitious system, but did not provide the crucial details of how one would construct such a service. Over the years, a few individuals have described in detail and actually implemented scaled-down versions of the Eternity Service. Publius, Free Haven, and the other distributed publishing systems described in this book fit into this category.

Eternity Usenet

An early implementation of a scaled-down version of the Eternity Service was proposed and implemented by Adam Back. Unlike the previously described publishing systems, this system didn't rely on volunteers to donate disk space. Instead, the publishing system was built on top of the Usenet news system. For this reason the system was called Eternity Usenet.

The Usenet news system propagates messages to servers all over the world and therefore qualifies as a distributed storage medium. However, Usenet is far from an ideal long-term storage mechanism. Messages posted to a Usenet newsgroup can take days to propagate to all Usenet servers. Not all Usenet news servers subscribe to all Usenet newsgroups. In fact, any system administrator can locally censor documents by not subscribing to a particular newsgroup. Usenet news posts can also become the victims of cancel or supercede messages. They are relatively easy to fake and therefore attractive to individuals who wish to censor a particular Usenet post.

The great volume of Usenet traffic necessitates the removal of old Usenet articles in favor of newer ones. This means that something posted to Usenet today may not be available two weeks from now, or even a few days from now. There are a few servers that archive Usenet articles for many years, but because there are not many of these servers, they present an easy target for those who wish to censor an archived document.

Finally, there is no way to tell if a Usenet message has been modified. Eternity Usenet addresses this by allowing an individual to digitally sign the message.

File-sharing systems

Up until now we have been discussing only systems that allow an individual to publish material on servers owned by others. However, Napster, a program that allows individuals to share files residing on their own hard drives, has been said to have started the whole peer-to-peer revolution. Napster allows individuals to share MP3 files over the Internet. The big debate concerning Napster is whether

Microsoft Exhibit 1019

this file sharing is legal. Many of the shared MP3 files are actually copied from one computer to another without any sort of royalty being paid to the artist that created the file. We will not discuss this particular issue any further as it is beyond the scope of this chapter. We are interested in the file-sharing mechanism and the trust issues involved.

Napster

Let's say Alice has a collection of MP3 files on her computer's hard drive. Alice wishes to share these files with others. She downloads the Napster client software and installs it on her computer. She is now ready to share the MP3 files. The list of MP3 files and associated descriptions is sent to the Napster server by the client software. This server adds the list to its index of MP3 files. In addition to storing the name and description of the MP3 files, the server also stores Alice's IP address. Alice's IP address is necessary, as the Napster server does not actually store the MP3 files themselves, but rather just pointers to them.

Alice can also use the Napster client software to search for MP3 files. She submits a query to the Napster server and a list of matching MP3 files is returned. Using the information obtained from the Napster server, Alice's client can connect to any of the computers storing these MP3 files and initiate a file transfer. Once again the issue of logging becomes important. Not only does Alice have to worry about logging on the part of the Napster server, but she also has to worry about logging done by the computer that she is copying files from. It is this form of logging that allowed the band Metallica to identify individuals who downloaded their music.

The natural question to ask is whether one of our previously described anonymizing tools could be used to combat this form of logging. Unfortunately the current answer is no. The reason for this is that the Napster server and client software speak a protocol that is not recognized by any of our current anonymizing tools. A protocol is essentially a set of messages recognized by both programs involved in a conversation—in this case the Napster client and server. This does not mean that such an anonymizing tool is impossible to build, only that current tools won't fit the bill.

Gnutella

Gnutella, described in Chapter 8, is a pure peer-to-peer file-sharing system. Computers running the Gnutella software connect to some preexisting network and become part of this network. We call computers running the Gnutella software Gnutella clients. Once part of this network, the Gnutella client can respond to queries sent by other members of the network, generate queries itself, and

Microsoft Exhibit 1019

participate in file sharing. Queries are passed from client to client and responses are passed back over the same set of clients that the requests originated from. This prevents meaningful logging of IP addresses and queries, because the client attempting to log the request has no way of knowing which client made the original request. Each client is essentially just forwarding the request made by another member of the network. Queries therefore remain for the most part anonymous. The individual that made the query is hidden among the other members of the peer-to-peer network, as with the Crowds system.

File transfer in Gnutella is done directly instead of via intermediaries. This is done for performance reasons; however, it also means that file transfer is not anonymous. The individual copying the file is no longer hidden among the other network members. The IP address of the client copying the file can now be logged.

Let's say that client A wishes to copy a file that resides on client B. Gnutella client A contacts client B and a file transfer is initiated. Client B can now log A's IP address and the fact that A copied a particular file. Although this sort of logging may seem trivial and harmless, it led to the creation of the web site called the Gnutella Wall of Shame. This web site lists the IP addresses and domain names of computers that allegedly downloaded a file that was advertised as containing child pornography. The file did not actually contain child pornography, but just the fact that a client downloaded the file was enough to get it placed on the list. Of course, any web site claiming to offer specific content could perform the same violation of privacy.

Freenet

Freenet, described in Chapter 9, is a pure peer-to-peer anonymous publishing system. Files are stored on a set of volunteer file servers. This set of file servers is dynamic—servers can join and leave the system at any time. A published file is copied to a subset of servers in a store-and-forward manner. Each time the file is forwarded to the next server, the origin address field associated with the file can be changed to some random value. This means that this field is essentially useless in trying to determine where the file originated. Therefore, files can be published anonymously.

Queries are handled in exactly the same way—the query is handed from one server to another and the resulting file (if any) is passed back through the same set of servers. As the file is passed back, each server can cache it locally and serve it in response to future requests for that file. It is from this local caching that Freenet derives its resistance to censorship. This method of file transfer also prevents meaningful logging, as each server doesn't know the ultimate destination of the file.

Microsoft Exhibit 1019

Content certification

Now that we have downloaded a file using one of the previously described systems, how do we know it is the genuine article? This is exactly the same question we asked at the beginning of this chapter. However, for certain files we may not really care that we have been duped into downloading the wrong file. A good example of this is MP3 files. While we may have wasted time downloading the file, no real harm was done to our computer. In fact, several artists have made bogus copies of their work available on such file-sharing programs as Napster. This is an attempt to prevent individuals from obtaining the legitimate version of the MP3 file.

The "problem" with many of the publishing systems described in this book is that we don't know who published the file. Indeed this is actually a feature required of anonymous publishing systems. Anonymously published files are not going to be accompanied by a digital certificate and signature (unless the signature is associated with a pseudonym). Some systems, such as Publius, provide a tamper-check mechanism. However, just because a file passes a tamper check does not mean that the file is virus-free and has actually been uploaded by the person believed by the recipient to have uploaded it.

Trust and search engines

File-sharing and anonymous publishing programs provide for distributed, and in some cases fault tolerant, file storage. But for most of these systems, the ability to store files is necessary but not sufficient to achieve their goals. Most of these systems have been built with the hope of enabling people to make their files available to others. For example, Publius was designed to allow people to publish documents so that they are resistant to censorship. But publishing a document that will never be read is of limited use. As with the proverbial tree falling in the forest that nobody was around to hear, an unread document makes no sound—it cannot inform, motivate, offend, or entertain. Therefore, indexes and search engines are important companions to file-sharing and anonymous publishing systems.

As previously stated, all of these file-sharing and anonymous publishing programs are still in their infancy. Continuing this analogy, we can say that searching technologies for these systems are in the embryonic stage. Unlike the Web, which now has mature search engines such as Google and Yahoo!, the world of peer-to-peer search engines consists of ad hoc methods, none of which work well in all situations. Web search engines such as Google catalogue millions of web pages by having web crawlers (special computer programs) read web pages and catalogue them. This method will not work with many of the systems

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described in this book. Publius, for example, encrypts its content and only someone possessing the URL can read the encrypted file. It makes no sense for a web crawler to visit each of the Publius servers and read all the files stored on them. The encrypted files will look like gibberish to the web crawler.

The obvious solution is to somehow send a list of known Publius URLs to a special web crawler that knows how to interpret them. Of course, submitting the Publius URL to the web crawler would be optional, as one may not wish to widely publicize a particular document.

Creating a Publius web crawler and search engine would be fairly straightforward. Unfortunately this introduces a new way to censor Publius documents. The company or individual operating the Publius web crawler can censor a document by simply removing its Publius URL from the crawler's list. The owners of the search engine can not only log your query but can also control exactly what results are returned from the search engine.

Let us illustrate this with a trivial example. You go to the Publius search engine and enter the phrase "Windows 95." The search engine examines the query and decides to send you pages that only mention Linux. Although this may seem like a silly example, one can easily see how this could lead to something much more serious. Of course, this is not a problem unique to Publius search engines—this problem can occur with the popular web search engines as well. Indeed, many of the popular search engines sell advertisements that are triggered by particular search queries, and reorder search results so that advertisers' pages are listed at the top.

Distributed search engines

The problem with a centralized search engine, even if it is completely honest, is that it has a single point of failure. It presents an enticing target to anyone who wishes to censor the system. This type of attack has already been used to temporarily shut down Napster. Because all searches for MP3 files are conducted via the Napster server, just shut down the server and the system becomes useless.

This dramatically illustrates the need for a distributed index, the type of index that we find in Freenet. Each Freenet server keeps an index of local files as well as an index of some files stored in some neighboring servers. When a Freenet server receives a query it first checks to see if the query can be satisfied locally. If it cannot, it uses the local index to decide which server to forward the request to. The index on each server is not static and changes as files move through the system.

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One might characterize Gnutella as having a distributed index. However, each client in the network is concerned only with the files it has stored locally. If a query can be satisfied locally, the client sends a response. If not, it doesn't respond at all. In either case the previous client forwards its query to other members of the network. Therefore, one query can generate many responses. The query is essentially broadcast to all computers on the Gnutella network.

Each Gnutella client can interpret the query however it sees fit. Indeed, the Gnutella client can return a response that has nothing at all to do with the query. Therefore, the query results must be viewed with some suspicion. Again it boils down to the issue of trust.

In theory, an index of Publius documents generated by a web crawler that accepts submissions of Publius URLs could itself be published using Publius. This would prevent the index from being censored. Of course, the URL submission system and the forms for submitting queries to the index could be targeted for censorship.

Note that in many cases, indexes and search engines for the systems described in this book can be developed as companion systems without changing the underlying distributed system. It was not necessary for Tim Berners-Lee (the inventor of the World Wide Web) to build the many web search engines and indexes that have developed. The architecture of the Web was such that these services could be built on top of the underlying infrastructure.

Deniability

The ability to locate Publius documents can actually be a double-edged sword. On the one hand, being able to find a document is essential for that document to be read. On the other hand, the first step in censoring a document is locating it.

One of the features of Publius is that server administrators cannot read the content stored on their servers because the files are encrypted. A search engine could, in some sense, jeopardize this feature. Armed with a search engine, Publius administrators could conceivably learn that their servers are hosting something they find objectionable. They could then go ahead and delete the file from their servers. Therefore, a search engine could paradoxically lead to greater censorship in such anonymous publishing systems.

Furthermore, even if server administrators do not wish to censor documents, once presented with a Publius URL that indicates an objectionable document resides on their servers, they may have little choice under local laws. Once the server operators know what documents are on their servers, they lose the ability to deny knowledge of the kinds of content published with Publius.

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Some Publius server operators may wish to help promote free speech but may not wish to specifically promote or endorse specific speech that they find objectionable. While they may be willing to host a server that may be used to publish content that they would find objectionable, they may draw the line at publicizing that content. In effect, they may be willing to provide a platform for free speech, but not to provide advertising for the speakers who use the platform.

Table 15-2 summarizes the problems with censorship-resistant and file sharing systems we have discussed in this chapter.

Table 15-2. Trust issues in censorship-resistant publishing systems

Risk	Solution	Trust principle
Servers, proxies, ISPs, or other "nodes" you interact with may log your requests (making it possible for your actions to be traced).	Use a secure channel and/or an anonymity tool so other parties do not get access to information that might link you to a particular action.	Reduce risk, and reduce the number of people that must be trusted.
Proxies and search engines may alter content they return to you in ways they don't disclose.	Try multiple proxies (and compare results before trusting any of them) or run your own proxy.	Reduce risk, and reduce the number of people that must be trusted.
Multiple parties may collaborate to censor your document.	Publish your document in a way that requires a large number of parties to collaborate before they can censor successfully. (Only a small subset of parties needs to be trusted not to collaborate, and any subset of that size will do.)	Reduce the number of people that must be trusted.
Parties may censor your document by making it appear as if you updated your document when you did not.	Publish your document in a way that it cannot be updated, or publish your document in a way that requires a large number of parties to collaborate before they can make it appear that you updated your document. (Only a small subset of parties needs to be trusted not to collaborate, and any subset of that size will do.)	Reduce the number of people that must be trusted.
Publishers may flood disks with bogus content as part of a denial of service attack.	Impose limits or quotas on publishers; require publishers to pay for space with money, computation, space donations; establish a reputation system for publishers.	Reduce risk; look for positive reputations.

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Table 15-2. Trust issues in censorship-resistant publishing systems (continued)

Risk	Solution	Trust principle
Censors may use laws to try to force documents to be deleted.	Publish your document in a way that requires a large number of parties to collaborate before they can censor successfully. (Only a small subset of parties needs to be trusted not to collaborate, and any subset of that size will do.)	Reduce the number of people that must be trusted.
Censors may threaten publishers to get them to delete their own documents.	Publish your document in a way that even the publisher cannot delete it.	Reduce risk, and reduce the number of people that must be trusted.

Conclusions

In this chapter we have presented an overview of the areas where trust plays a role in distributed file-sharing systems, and we have described some of the methods that can be used to increase trust in these systems. By signing software they make available for download, authors can provide some assurance that their code hasn't been tampered with and facilitate the building of a reputation associated with their name and key. Anonymity tools and tools for establishing secure channels can reduce the need to trust ISPs and other intermediaries not to record or alter information sent over the Internet. Quota systems, CPU payment systems, and e-cash payment systems can reduce the risk of denial of service attacks. Search engines can help facilitate dissemination of files but can introduce additional trust issues.

There are several open issues. The first is the lack of existence of a global Public Key Infrastructure (PKI). Many people believe that such a PKI is not ever going to be possible. This has ramifications for trust, because it implies that people may never be able to trust signed code unless they have a direct relationship with the signer. While the problem of trusting strangers exists on the Net, strangely, it is also very difficult to truly be anonymous on the Internet. There are so many ways to trace people and correlate their online activity that the sense of anonymity that most people feel online is misplaced. Thus, there are two extremes of identity: both complete assurance of identity and total anonymity are very difficult to achieve. More research is needed to see how far from the middle we can push in both directions, because each extreme offers possibilities for increased trust in cyberspace.

Microsoft Exhibit 1019

chapter sixteen

Accountability

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One year after its meteoric rise to fame, Napster faces a host of problems. The best known of these problems is the lawsuit filed by the Recording Industry Association of America against Napster, Inc. Close behind is the decision by several major universities, including Yale, the University of Southern California, and Indiana University, to ban Napster traffic on their systems, thus depriving the Napster network of some of its highest-bandwidth music servers. The most popular perception is that universities are blocking Napster access out of fear of lawsuit. But there is another reason.

Napster users eat up large and unbounded amounts of bandwidth. By default, when a Napster client is installed, it configures the host computer to serve MP3s to as many other Napster clients as possible. University users, who tend to have faster connections than most others, are particularly effective servers. In the process, however, they can generate enough traffic to saturate a network. It was this reason that Harvard University cited when deciding to allow Napster, yet limit its bandwidth use.

Gnutella, the distributed replacement for Napster, is even worse: not only do downloads require large amounts of bandwidth, but searches require broadcasting to a set of neighboring Gnutella nodes, which in turn forward the request to other nodes. While the broadcast does not send the request to the entire Gnutella network, it still requires bandwidth for each of the many computers queried.

As universities limit Napster bandwidth or shut it off entirely due to bandwidth usage, the utility of the Napster network degrades. As the Gnutella network grows, searching and retrieving items becomes more cumbersome. Each service threatens to dig its own grave—and for reasons independent of the legality of trading MP3s. Instead, the problem is *resource allocation.*

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Problems in resource allocation come up constantly in offering computer services. Traditionally they have been solved by making users *accountable* for their use of resources. Such accountability in distributed or peer-to-peer systems requires planning and discipline.

Traditional filesystems and communication mediums use accountability to maintain centralized control over their respective resources—in fact, the resources allocated to users are commonly managed by "user accounts." Filesystems use quotas to restrict the amount of data that users may store on the systems. ISPs measure the bandwidth their clients are using—such as the traffic generated from a hosted web site—and charge some monetary fee proportional to this amount.

Without these controls, each user has an incentive to squeeze all the value out of the resource in order to maximize personal gain. If one user has this incentive, so do all the users.

Biologist Garrett Hardin labeled this economic plight the "tragedy of the commons."* The "commons" (originally a grazing area in the middle of a village) is any resource shared by a group of people: it includes the air we breathe, the water we drink, land for farming and grazing, and fish from the sea. The tragedy of the commons is that a commonly owned resource will be overused until it is degraded, as all agents pursue self-interest first. Freedom in a commons brings ruin to all; in the end, the resource is exhausted.

We can describe the problem by further borrowing from economics and political science. Mancur Olson explained the problem of collective actions and public goods as follows:

> "[U]nless the number of individuals in a group is quite small, or unless there is coercion or some other special device to make individuals act in their common interest, rational, self-interested individuals will not act to achieve their common or group interests.†

The usual solution for commons problems is to assign ownership to the resource. This ownership allows a party to profit from the resource, thus providing the incentive to care for it. Most real-world systems take this approach with a fee-for-service business model.

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* Garrett Hardin (1968), "The Tragedy of the Commons," *Science* 162, pp. 1243–1248.

† Mancur Olson (1982), "The Logic of Collective Action." In Brian Barry and Russell Hardin, eds., *Rational Man and Irrational Society*. Beverly Hills, CA: Sage, p. 44.

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Decentralized peer-to-peer systems have similar resource allocation and protection requirements. The total storage or bandwidth provided by the sum of all peers is still finite. Systems need to protect against two main areas of attack:

Denial of service (DoS) attacks
: Overload a system's bandwidth or processing ability, causing the loss of service of a particular network service or all network connectivity. For example, a web site accessed millions of times may show "503" unavailability messages or temporarily refuse connections.

Storage flooding attacks
: Exploit a system by storing a disproportionally large amount of data so that no more space is available for other users.

As the Napster and Gnutella examples show, attacks need not be malicious. System administrators must be prepared for normal peaks in activity, accidental misuse, and the intentional exploitation of weaknesses by adversaries. Most computers that offer services on a network share these kinds of threats.

Without a way to protect against the tragedy of the commons, collaborative networking rests on shaky ground. Peers can abuse the protocol and rules of the system in any number of ways, such as the following:

- Providing corrupted or low-quality information
- Reneging on promises to store data
- Going down during periods when they are needed
- Claiming falsely that other peers have abused the system in these ways

These problems must be addressed before peer-to-peer systems can achieve lasting success. Through the use of various accountability measures, peer-to-peer systems—including systems that offer protection for anonymity—may continue to expand as overlay networks through the existing Internet.

This chapter focuses on types of accountability that collaborative systems can use to protect against resource allocation attacks. The problem of accountability is usually broken into two parts:

Restricting access
: Each computer system tries to limit its users to a certain number of connections, a certain quantity of data that can be uploaded or downloaded, and so on. We will describe the technologies for doing this that are commonly called *micropayments*, a useful term even though at first it can be misleading. (They don't necessarily have to involve an exchange of money, or even of computer resources.)

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Selecting favored users

This is normally done through maintaining a *reputation* for each user the system communicates with. Users with low reputations are allowed fewer resources, or they are mistrusted and find their transactions are rejected.

The two parts of the solution apply in different ways but work together to create accountability. In other words, a computer system that is capable of restricting access can then use a reputation system to grant favored access to users with good reputations.

The difficulty of accountability

In simple distributed systems, rudimentary accountability measures are often sufficient. If the list of peers is generally static and all are known to each other by hostname or address, misbehavior on anyone's part leads to a permanent bad reputation. Furthermore, if the operators of a system are known, preexisting mechanisms such as legal contracts help ensure that systems abide by protocol.

In the real world, these two social forces—reputation and law—have provided an impetus for fair trade for centuries. Since the earliest days of commerce, buyers and merchants have known each others' identities, at first through the immediacy of face-to-face contact, and later through postal mail and telephone conversations. This knowledge has allowed them to research the past histories of their trading partners and to seek legal reprisal when deals go bad. Much of today's e-commerce uses a similar authentication model: clients (both consumers and businesses) purchase items and services from known sources over the Internet and the World Wide Web. These sources are uniquely identified by digital certificates, registered trademarks, and other addressing mechanisms.

Peer-to-peer technology removes central control of such resources as communication, file storage and retrieval, and computation. Therefore, the traditional mechanisms for ensuring proper behavior can no longer provide the same level of protection.

Special problems posed by peer-to-peer systems

Peer-to-peer systems have to treat identity in special ways for several reasons:

- The technology makes it harder to uniquely and permanently identify peers and their operators. Connections and network maps might be transient. Peers might be able to join and leave the system. Participants in the system might wish to hide personal identifying information.

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- Even if users have an identifying handle on the peer they're dealing with, they have no idea who the peer is and no good way to assess its history or predict its performance.
- Individuals running peer-to-peer services are rarely bound by contracts, and the cost and time delay of legal enforcement would generally outweigh their possible benefit.

We choose to deal with these problems—rather than give up and force everyone on to a centralized system with strong user identification—to pursue two valuable goals on the Internet: privacy and dynamic participation.

Privacy is a powerfully appealing goal in distributed systems, as discussed in Chapter 12, *Free Haven*. The design of many such systems features privacy protection for people offering and retrieving files.

Privacy for people offering files requires a mechanism for inserting and retrieving documents either anonymously or pseudonymously.* Privacy for people retrieving files requires a means to communicate—via email, Telnet, FTP, IRC, a web client, etc.—while not divulging any information that could link the user to his or her real-world persona.†

Dynamic participation has both philosophical and practical advantages. The Internet's loosely connected structure and explosive growth suggest that any peer-to-peer system must be similarly flexible and dynamic in order to be scalable and sustain long-term use. Similarly, the importance of ad hoc networks will probably increase in the near future as wireless connections get cheaper and more ubiquitous. A peer-to-peer system should therefore let peers join and leave smoothly, without impacting functionality. This design also decreases the risk of systemwide compromise as more peers join the system. (It helps if servers run a

* A pseudonymous identity allows other participants to link together some or all the activities a person does on the system, without being able to determine who the person is in real life. Pseudonymity is explored later in this chapter and in Chapter 12.

† In retrospect, the Internet appears not to be an ideal medium for anonymous communication and publishing. Internet services and protocols make both passive sniffing and active attack too easy. For instance, email headers include the routing paths of email messages, including DNS hostnames and IP addresses. Web browsers normally display user IP addresses; cookies on a client's browser may be used to store persistent user information. Commonly used online chat applications such as ICQ and Instant Messenger also divulge IP addresses. Network cards in promiscuous mode can read all data flowing through the local Ethernet. With all these possibilities, telephony or dedicated lines might be better suited for this goal of privacy protection. However, the ubiquitous nature of the Internet has made it the only practical consideration for digital transactions across a wide area, like the applications discussed in this book.

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variety of operating systems and tools, so that a single exploit cannot compromise most of the servers at once.)

Peer-to-peer models and their impacts on accountability

There are many different models for peer-to-peer systems. As the systems become more dynamic and diverge from real-world notions of identity, it becomes more difficult to achieve accountability and protect against attacks on resources.

The simplest type of peer-to-peer system has two main characteristics. First, it contains a fairly static list of servers; additions and deletions are rare and may require manual intervention. Second, the identities of the servers (and to some extent their human operators) are known, generally by DNS hostname or static IP host address. Since the operators can be found, they may have a legal responsibility or economic incentive—leveraged by the power of reputation—to fulfill the protocols according to expectation.

An example of such a peer-to-peer system is the Mixmaster remailer. A summary of the system appears in Chapter 7, *Mixmaster Remailers*. The original Mixmaster client software was developed by Lance Cottrell and released in 1995.* Currently, the software runs on about 30 remailer nodes, whose locations are published to the newsgroup *alt.privacy.anon-server* and at web sites such as *http://efga.org*.† The software itself can be found at *http://mixmaster.anonymizer.com*.

Remailer nodes are known by hostname and remain generally fixed. While anybody can start running a remailer, the operator needs to spread information about her new node to web pages that publicize node statistics, using an out-of-band channel (meaning that something outside the Mixmaster system must be used—most of the time, manually sent email). The location of the new node is then manually added to each client's software configuration files. This process of manually adding new nodes leads to a system that remains generally static. Indeed, that's why there are so few Mixmaster nodes.

A slightly more complicated type of peer-to-peer system still has identified operators but is dynamic in terms of members. That is, the protocol itself has support for adding and removing participating servers. One example of such a system is

* Lance Cottrell (1995) "Mixmaster and Remailer Attacks," *http://www.obscura.com/~loki/remailer/remailer-essay.html*.

† "Electronic Frontiers Georgia List of Public Mixmaster Remailers," *http://anon.efga.org/Remailers*.

Microsoft Exhibit 1019

Gnutella. It has good support for new users (which are also servers) joining and leaving the system, but at the same time, the identity and location of each of these servers is generally known through the hosts list, which advertises existing hosts to new ones that wish to join the network. These sorts of systems can be very effective, because they're generally easy to deploy (there's no need to provide any real protection against people trying to learn the identity of other participants), while at the same time they allow many users to freely join the system and donate their resources.

Farther still along the scale of difficulty lie peer-to-peer systems that have dynamic participants and pseudonymous servers. In these systems, the actual servers that store files or proxy communication live within a digital fog that conceals their geographic locations and other identifying features. Thus, the mapping of pseudonym to real-world identity is not known. A given pseudonym may be pegged with negative attributes, but a user can just create a new pseudonym or manage several at once. Since a given server can simply disappear at any time and reappear as a completely new entity, these sorts of designs require a micropayment system or reputation system to provide accountability on the server end. An example of a system in this category is the Free Haven design: each server can be contacted via a remailer reply block and a public key, but no other identifying features are available.

The final peer-to-peer model on this scale is a dynamic system with fully anonymous operators. A server that is fully anonymous lacks even the level of temporary identity provided by a pseudonymous system like Free Haven. Since an anonymous peer's history is by definition unknown, all decisions in an anonymous system must be based only on the information made available during each protocol operation. In this case, peers cannot use a reputation system, since there is no real opportunity to establish a profile on any server. This leaves a micropayment system as the only reasonable way to establish accountability. On the other hand, because the servers themselves have no long-term identities, this may limit the number of services or operations such a system could provide. For instance, such a system would have difficulty offering long-term file storage and backup services.

Purposes of micropayments and reputation systems

The main goal of accountability is to maximize a server's *utility* to the overall system while minimizing its potential *threat*. There are two ways to minimize the threat.

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- One approach is to limit our risk (in bandwidth used, disk space lost, or whatever) to an amount roughly equivalent to our benefit from the transaction. This suggests the fee-for-service or micropayment model mentioned at the beginning of the chapter.
- The other approach is to make our risk proportional to our trust in the other parties. This calls for a reputation system.

In the micropayment model, a server makes decisions based on fairly immediate information. Payments and the value of services are generally kept small, so that a server only gambles some small amount of lost resources for any single exchange. If both parties are satisfied with the result, they can continue with successive exchanges. Therefore, parties require little prior information about each other for this model, as the risk is small at any one time. As we will see later in this chapter, where we discuss real or existing micropayment systems, the notion of payment might not involve any actual currency or cash.

In the reputation model, for each exchange a server risks some amount of resources proportional to its trust that the result will be satisfactory. As a server's reputation grows, other nodes become more willing to make larger payments to it. The micropayment approach of small, successive exchanges is no longer necessary.

Reputation systems require careful development, however, if the system allows impermanent and pseudonymous identities. If an adversary can gain positive attributes too easily and establish a good reputation, she can damage the system. Worse, she may be able to "pseudospoof," or establish many seemingly distinct identities that all secretly collaborate with each other.

Conversely, if a well-intentioned server can incur negative points easily from short-lived operational problems, it can lose reputation too quickly. (This is the attitude feared by every system administrator: "Their web site happened to be down when I visited, so I'll never go there again.") The system would lose the utility offered by these "good" servers.

As we will see later in this chapter, complicated protocols and calculations are required for both micropayments and reputation systems. Several promising micropayment systems are in operation, while research on reputation systems is relatively young. These fields need to develop ways of checking the information being transferred, efficient tests for distributed computations, and, more broadly, some general algorithms to verify behavior of decentralized systems.

There is a third way to handle the accountability problem: ignore the issue and engineer the system simply to survive some faulty servers. Instead of spending

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time on ensuring that servers fulfill their function, leverage the vast resources of the Internet for redundancy and mirroring. We might not know, or have any way to find out, if a server is behaving according to protocol (i.e., whether that server is storing files and responding to file queries, forwarding email or other communications upon demand, and correctly computing values or analyzing data). Instead, if we replicate the file or functionality through the system, we can ensure that the system works correctly with high probability, despite misbehaving components. This is the model used by Napster, along with some of the systems discussed in this book, such as Freenet and Gnutella.

In general, the popular peer-to-peer systems take a wide variety of approaches to solving the accountability problem. For instance, consider the following examples:

- Freenet dumps unpopular data on the floor, so people flooding the system with unpopular data are ultimately ignored. Popular data is cached near the requester, so repeated requests won't traverse long sections of the network.
- Gnutella doesn't "publish" documents anywhere except on the publisher's computer, so there's no way to flood other systems. (This has a great impact on the level of anonymity actually offered.)
- Publius limits the submission size to 100K. (It remains to be seen how successful this will be; they recognize it as a problem.)
- Mojo Nation uses micropayments for all peer-to-peer exchanges.
- Free Haven requires publishers to provide reliable space of their own if they want to insert documents into the system. This economy of reputation tries to ensure that people donate to the system in proportion to how much space they use.

Junk mail as a resource allocation problem

The familiar problem of junk email (known more formally as *unsolicited commercial email*, and popularly as *spam*) yields some subtle insights into resource allocation and accountability. Junk mail abuses the unmetered nature of email and of Internet bandwidth in general. Even if junk email achieves only an extremely small success rate, the sender is still successful because the cost of sending each message is essentially zero.

Spam wastes both global and individual resources. On a broad scale, it congests the Internet, wasting bandwidth and server CPU cycles. On a more personal level, filtering and deleting spam can waste an individual's time (which, collectively, can represent significant person-hours). Users also may be faced with

Microsoft Exhibit 1019

metered connection charges, although recent years have seen a trend toward unmetered service and always-on access.

Even though the motivations for junk email might be economic, not malicious, senders who engage in such behavior play a destructive role in "hogging" resources. This is a clear example of the tragedy of the commons.

Just as some environmental activists suggest curbing pollution by making consumers pay the "real costs" of the manufacturing processes that cause pollution, some Internet developers are considering ways of stopping junk email by placing a tiny burden on each email sent, thus forcing the sender to balance the costs of bulk email against the benefits of responses. The burden need not be a direct financial levy; it could simply require the originator of the email to use significant resources. The cost of an email message should be so small that it wouldn't bother any individual trying to reach another; it should be just high enough to make junk email unprofitable. We'll examine such micropayment schemes later in this chapter.

We don't have to change the infrastructure of the Internet to see a benefit from email micropayments. Individuals can adopt personal requirements as recipients. But realistically, individual, nonstandard practices will merely reduce the usability of email. Although individuals adopting a micropayment scheme may no longer be targeted, the scheme would make it hard for them to establish relationships with other Internet users, while junk emailers would continue to fight over the commons.

Pseudonymity and its consequences

Many, if not most, of the services on the Internet today do not deal directly with legal identities. Instead, web sites and chat rooms ask their users to create a *handle* or *pseudonym* by which they are known while using that system. These systems should be distinguished from those that are fully anonymous; in a fully anonymous system, there is no way to refer to the other members of the system.

Problems with pseudospoofing and possible defenses

The most important difficulty caused by pseudonymity is *pseudospoofing*. A term first coined by L. Detweiler on the Cypherpunks mailing list, pseudospoofing means that one person creates and controls many phony identities at once. This is a particularly bad loophole in reputation systems that blithely accept input from just any user, like current web auction sites. An untrustworthy person can pseudospoof to return to the system after earning a bad reputation, and he can

Microsoft Exhibit 1019

even create an entire tribe of accounts that pat each other on the back. Pseudospoofing is a major problem inherent in pseudonymous systems.

Lots of systems fail in the presence of pseudospoofing. Web polls are one example; even if a web site requires registration, it's easy for someone to simply register and then vote 10, 15, or 1,500 times. Another example is a free web hosting site, such as GeoCities, which must take care to avoid someone registering under six or seven different names to obtain extra web space.

Pseudospoofing is hard in the real world, so most of us don't think about it. After all, in the real world, changing one's appearance and obtaining new identities is relatively rare, spy movies to the contrary. When we come online, we bring with us the assumptions built up over a lifetime of dealing with people who can be counted on to be the "same person" next time we meet them. Pseudospoofing works, and works so well, because these assumptions are *completely unjustified online*. As shown by the research of psychologist Sherry Turkle and others, multiple identities are common in online communities.

So what can we do about pseudospoofing? Several possibilities present themselves:

- Abandon pseudonymous systems entirely. Require participants in a peer-to-peer system to prove conclusively who they are. This is the direction taken by most work on Public Key Infrastructures (PKIs), which try to tie each online users to some legal identity. Indeed, VeriSign used to refer to its digital certificates as "driver's licenses for the information superhighway."

  This approach has a strong appeal. After all, why should people be allowed to "hide" behind a pseudonym? And how can we possibly have accountability without someone's real identity?

  Unfortunately, this approach is unnecessary, unworkable, and in some cases undesirable. It's unnecessary for at least three reasons:

  — Identity does not imply accountability. For example, if a misbehaving user is in a completely different jurisdiction, other users may know exactly who he or she is and yet be unable to do anything about it. Even if they are in the same jurisdiction, the behavior may be perfectly legal, just not very nice.

  — Accountability is possible even in pseudonymous systems. This point will be developed at length in the rest of this chapter.

  — The problem with pseudospoofing is not that someone acts under a "fake" name, but that someone acts under more than one name. If we could somehow build a system that ensured that every pseudonym was

Microsoft Exhibit 1019

controlled by a distinct person, a reputation system could handle the problem.

Furthermore, absolute authentication is unworkable because it requires verifying the legal identities of all participants. On today's Internet, this is a daunting proposition. VeriSign and other PKI companies are making progress in issuing their "digital driver's licenses," but we are a far cry from that end. In addition, one then has to trust that the legal identities have not themselves been fabricated. Verification can be expensive and leaves a system that relies on it open to attack if it fails.

Finally, this proposed solution is undesirable because it excludes users who either cannot or will not participate. International users of a system may not have the same ways of verifying legal identity. Other users may have privacy concerns.

- Allow pseudonyms, but ensure that all participants are distinct entities. This is all that is strictly necessary to prevent pseudospoofing. Unfortunately, it tends to be not much easier than asking for everyone's legal identity.
- Monitor user behavior for evidence of pseudospoofing. Remove or "expose" accounts that seem to be controlled by the same person. The effectiveness of this approach varies widely with the application. It also raises privacy concerns for users.
- Make pseudospoofing unprofitable. Give new accounts in a system little or no resources until they can prove themselves by doing something for the system. Make it so expensive for an adversary to prove itself multiple times that it has no inclination to pseudospoof. This is the approach taken by the Free Haven project, which deals with new servers by asking them to donate resources to the good of the system as a whole.

All of these alternatives are just rules of thumb. Each of them might help us combat the problems of pseudospoofing, but it's hard to reach a conclusive solution. We'll return to possible technical solutions later in this chapter when we describe the Advogato system.

Reputation for sale—SOLD!

Pseudonymous systems are based on the assumption that each pseudonym is controlled by the same entity for the duration of the system. That is, the adversary's pseudonyms stay controlled by the adversary, and the good guys' pseudonyms stay controlled by the good guys.

Microsoft Exhibit 1019

What happens if the adversary takes control of someone who already has a huge amount of trust or resources in the system? Allowing accounts to change hands can lead to some surprising situations.

The most prevalent example of this phenomenon comes in online multiplayer games. One of the best-known such games is Ultima Online. Players gallivant around the world of Brittania, completing quests, fighting foes, and traipsing around dungeons, in the process accumulating massive quantities of loot. Over the course of many, many hours, a player can go from a nobody to the lord and master of his own castle. Then he can sell it all to someone else.

Simply by giving up his username and password, an Ultima Online player can transfer ownership of his account to someone else. The new owner obtains all the land and loot that belonged to the old player. More importantly, she obtains the reputation built up by the old player. The transfer can be carried out independently of the game; no one need ever know that it happened. As far as anyone else knows, the game personality is the same person. Until the new owner does something "out of character," or until the news spreads somehow, there is no way to tell that a transfer has occurred.

This has led to a sort of cottage industry in trading game identities for cash online. Ultima Online game identities, or "avatars," can be found on auction at eBay. Other multiplayer online games admit the occurrence of similar transactions. Game administrators can try to forbid selling avatars, but as long as it's just a matter of giving up a username and password, it will be an uphill battle.

The point of this example is that reputations and identities do not bind as tightly to people online as they do in the physical world. Reputations can be sold or stolen with a single password. While people can be coerced or "turned" in the physical world, it's much harder. Once again, the assumptions formed in the physical world turn out to be misleading online.

One way of dealing with this problem is to embed an important piece of information, such as a credit card number, into the password for an account. Then revealing the password reveals the original user's credit card number as well, creating a powerful incentive not to trade away the password. The problem is that if the password is ever accidentally compromised, the user now loses not just the use of his or her account, but the use of a credit card as well.

Another response is to make each password valid only for a certain number of logins; to get a new password, the user must prove that he is the same person who applied for the previous password. This does not stop trading passwords, however—it just means the "original" user must hang around to renew the password each time it expires.

Microsoft Exhibit 1019

Common methods for dealing with flooding and DoS attacks

We've seen some examples of resource allocation problems and denial of service attacks. These problems have been around for a long while in various forms, and there are several widespread strategies for dealing with them. We'll examine them in this section to show that even the most common strategies are subject to attack—and such attacks can be particularly devastating to peer-to-peer systems.

Caching and mirroring

One of the simplest ways to maintain data availability is to mirror it. Instead of hosting data on one machine, host it on several. When one machine becomes congested or goes down, the rest are still available. Popular software distributions like the Perl archive CPAN and the GNU system have a network of mirror sites, often spread across the globe to be convenient to several different nations at once.

Another common technique is caching: If certain data is requested very often, save it in a place that is closer to the requester. Web browsers themselves cache recently visited pages.

Simple to understand and straightforward to implement, caching and mirroring are often enough to withstand normal usage loads. Unfortunately, an adversary bent on a denial of service attack can target mirrors one by one until all are dead.

Active caching and mirroring

Simple mirroring is easy to do, but it also has drawbacks. Users must know where mirror sites are and decide for themselves which mirror to use. This is more hassle for users and inefficient to boot, as users do not generally know their networks well enough to pick the fastest web site. In addition, users have little idea of how loaded a particular mirror is; if many users suddenly decide to visit the same mirror, they may all receive worse connections than if they had been evenly distributed across mirror sites.

In 1999, Akamai Technologies became an overnight success with a service that could be called active mirroring. Web sites redirect their users to use special "Akamaized" URLs. These URLs contain information used by Akamai to dynamically direct the user to a farm of Akamai web servers that is close to the user on

Microsoft Exhibit 1019

the network. As the network load and server loads change, Akamai can switch users to the best server farm of the moment.

For peer-to-peer systems, an example of active caching comes in the Freenet system for file retrieval. In Freenet, file requests are directed to a particular server, but this server is in touch with several other servers. If the initial server has the data, it simply returns the data. Otherwise, it forwards the request to a neighboring server which it believes more capable of answering the request, and keeps a record of the original requester's address. The neighboring server does the same thing, creating a chain of servers. Eventually the request reaches a server that has the data, or it times out. If the request reaches a server that has the data, the server sends the data back through the chain to the original requester. Every server in the chain, in addition, caches a copy of the requested data. This way, the next time the data is requested, the chance that the request will quickly hit a server with the data is increased.

Active caching and mirroring offer more protection than ordinary caching and mirroring against the "Slashdot effect" and flooding attacks. On the other hand, systems using these techniques then need to consider how an adversary could take advantage of them. For instance, is it possible for an adversary to fool Akamai into thinking a particular server farm is better- or worse-situated than it actually is? Can particular farms be targeted for denial of service attacks? In Freenet, what happens if the adversary spends all day long requesting copies of the complete movie *Lawrence of Arabia* and thus loads up all the local servers to the point where they have no room for data wanted by other people? These questions can be answered, but they require thought and attention.

For specific answers on a specific system, we might be able to answer these questions through a performance and security analysis. For instance, Chapter 14, *Performance*, uses Freenet and Gnutella as models for performance analysis. Here, we can note two general points about how active caching reacts to adversaries.

First, if the cache chooses to discard data according to which data was least recently used, the cache is vulnerable to an active attack. An adversary can simply start shoving material into the cache until it displaces anything already there. In particular, an adversary can simply request that random bits be cached. Active caching systems whose availability is important to their users should have some way of addressing this problem.

Next, guaranteeing service in an actively cached system with multiple users on the same cache is tricky. Different usage patterns fragment the cache and cause it to be less useful to any particular set of users. The situation becomes more

Microsoft Exhibit 1019

difficult when adversaries enter the picture: by disrupting cache coherency on many different caches, an adversary may potentially wreak more havoc than by mounting a denial of service attack on a single server.

One method for addressing both these problems is to shunt users to caches based on their observed behavior. This is a radical step forward from a simple least-recently-used heuristic. By using past behavior to predict future results, a cache has the potential to work more efficiently. This past behavior can be considered a special kind of reputation, a topic we'll cover in general later in this chapter.

But systems can also handle resource allocation using simpler and relatively well tested methods involving micropayments. In the next section, we'll examine some of them closely.

Micropayment schemes

Accountability measures based on micropayments require that each party offer something of value in an exchange. Consider Alice and Bob, both servers in a peer-to-peer system that involves file sharing or publishing. Alice may be inserting a document into the system and want Bob to store it for her. Alternatively, Alice may want Bob to anonymously forward some email or real-time Internet protocol message for her. In either case, Alice seeks some resource commodity—storage and bandwidth, respectively—from Bob. In exchange, Bob asks for a micropayment from Alice to protect his resources from overuse.

There are two main flavors of micropayments schemes. Schemes of the first type do not offer Bob any real redeemable value; their goal is simply to slow Alice down when she requests resources from Bob. She pays with a *proof of work* (POW), showing that she performed some computationally difficult problem. These payments are called *nonfungible*, because Bob cannot turn around and use them to pay someone else. With the second type of scheme, *fungible* micropayments, Bob receives a payment that holds some intrinsic or redeemable value. The second type of payment is commonly known as digital cash. Both of these schemes may be used to protect against resource allocation attacks.

POWs can prevent communication denial of service attacks. Bob may require someone who wishes to connect to submit a POW before he allocates any nontrivial resources to communication. In a more sophisticated system, he may start charging only if he detects a possible DoS attack. Likewise, if Bob charges to store data, an attacker needs to pay some (prohibitively) large amount to flood Bob's disk space. Still, POWs are not a perfect defense against an attacker with a

Microsoft Exhibit 1019

lot of CPU capacity; such an attacker could generate enough POWs to flood Bob with connection requests or data.

Varieties of micropayments or digital cash

The difference between micropayments and digital cash is a semantic one. The term "micropayment" has generally been used to describe schemes using small-value individual payments. Usually, Alice will send a micropayment for some small, incremental use of a resource instead of a single large digital cash "macropayment" for, say, a month's worth of service. We'll continue to use the commonly accepted phrase "micropayment" in this chapter without formally differentiating between the two types, but we'll describe some common designs for each type.

Digital cash may be either anonymous or identified. Anonymous schemes do not reveal Alice's identity to Bob or the bank providing the cash, while identified spending schemes expose her information. Hybrid approaches can be taken: Alice might remain anonymous to Bob but not to the bank or anonymous to everybody yet traceable. The latter system is a kind of pseudonymity; the bank or recipient might be able to relate a sequence of purchases, but not link them to an identity.

No matter the flavor of payment—nonfungible, fungible, anonymous, identified, large, or small—we want to ensure that a malicious user can't commit forgery or spend the same coin more than once without getting caught. A system of small micropayments might not worry about forgeries of individual micropayments, but it would have to take steps to stop large-scale, multiple forgeries.

Schemes identifying the spender are the digital equivalent of debit or credit cards. Alice sends a "promise of payment" that will be honored by her bank or financial institution. Forgery is not much of a problem here because, as with a real debit card, the bank ensures that Alice has enough funds in her account to complete the payment and transfers the specified amount to Bob. Unfortunately, though, the bank has knowledge of all of Alice's transactions.

Anonymous schemes take a different approach and are the digital equivalent of real cash. The electronic coin itself is worth some dollar amount. If Alice loses the coin, she's lost the money. If Alice manages to pay both Bob and Charlie with the same coin and not get caught, she's successfully double-spent the coin.

In the real world, government mints use special paper, microprinting, holograms, and other technologies to prevent forgery. In a digital medium, duplication is easy: just copy the bits! We need to find alternative methods to prevent

Microsoft Exhibit 1019

this type of fraud. Often, this involves looking up the coin in a database of spent coins. Bob might have a currency unique to him, so that the same coin couldn't be used to pay Charlie. Or coins might be payee-independent, and Bob would need to verify with the coin's issuing "mint" that it has not already been spent with Charlie.

With this description of micropayments and digital cash in mind, let's consider various schemes.

Nonfungible micropayments

Proofs of work were first advocated by Cynthia Dwork and Moni Naor* in 1992 as "pricing via processing" to handle resource allocation requests.

The premise is to make a user compute a moderately hard, but not intractable, computation problem before gaining access to some resource. It takes a long time to solve the problem but only a short time to verify that the user found the right solution. Therefore, Alice must perform a significantly greater amount of computational work to solve the problem than Bob has to perform to verify that she did it.

Dwork and Naor offer their system specifically as a way to combat electronic junk mail. As such, it can impose a kind of accountability within a distributed system.

To make this system work, a recipient refuses to receive email unless a POW is attached to each message. The POW is calculated using the address of the recipient and must therefore be generated specifically for the recipient by the sender. These POWs serve as a form of electronic postage stamp, and the way the recipient's address is included makes it trivial for the recipient to determine whether the POW is malformed. Also, a simple lookup in a local database can be used to check whether the POW has been spent before.

The computational problem takes some amount of time proportional to the time needed to write the email and small enough that its cost is negligible for an individual user or a mail distribution list. Only unsolicited bulk mailings would spend a large amount of computation cycles to generate the necessary POWs.

Recipients can also agree with individual users or mail distribution lists to use an access control list ("frequent correspondent list") so that some messages do not

* Cynthia Dwork and Moni Naor (1993), "Pricing via Processing or Combating Junk Mail," in Ernest F. Brickell, ed., *Advances in Cryptology— Crypto '92*, vol. 740 of *Lecture Notes in Computer Science*, pp. 139–147. Springer-Verlag,16–20 August 1992.

Microsoft Exhibit 1019

require a POW. These techniques are useful for social efficiency: if private correspondence instead costs some actual usage fee, users may be less likely to send email that would otherwise be beneficial, and the high bandwidth of the electronic medium may be underutilized.

Dwork and Naor additionally introduced the idea of a POW with a trap door: A function that is moderately hard to compute without knowledge of some secret, but easy to compute given this secret. Therefore, central authorities could easily generate postage to sell for prespecified destinations.

Extended types of nonfungible micropayments

Hash cash, designed by Adam Back in late 1997,* is an alternative micropayment scheme that is also based on POWs. Here, Bob calculates a *hash* or *digest*, a number that can be generated easily from a secret input, but that cannot be used to guess the secret input. (See Chapter 15, *Trust*.) Bob then asks Alice to guess the input through a brute-force calculation; he can set how much time Alice has to "pay" by specifying how many bits she must guess. Typical hashes used for security are 128 bits or 160 bits in size. Finding another input that will produce the entire hash (which is called a "collision") requires a prohibitive amount of time.

Instead, Bob requires Alice to produce a number for which some of the low-order bits match those of the hash. If we call this number of bits k, Bob can set a very small k to require a small payment or a larger k to require a larger payment. Formally, this kind of problem is called a "k-bit partial hash collision."

For example, the probability of guessing a 17-bit collision is 2^{-17}; this problem takes approximately 65,000 tries on average. To give a benchmark for how efficient hash operations are, in one test, our Pentium-III 800 MHz machine performed approximately 312,000 hashes per second.

Hash cash protects against double-spending by using individual currencies. Bob generates his hash from an ID or name known to him alone. So the hash cash coins given to Bob must be specific to Bob, and he can immediately verify their validity against a local spent-coin database.

Another micropayment scheme based on partial hash collisions is *client puzzles*, suggested by researchers Ari Juels and John Brainard of RSA Labs.† Client

* Adam Back, "Hash Cash: A Partial Hash Collision Based Postage Scheme," *http://www.cypherspace.org/~adam/hashcash*.

† A. Juels and J. Brainard, "Client Puzzles: A Cryptographic Defense Against Connection Depletion Attacks," *NDSS '99*.

Microsoft Exhibit 1019

puzzles were introduced to provide a cryptographic countermeasure against connection depletion attacks, whereby an attacker exhausts a server's resources by making a large number of connection requests and leaving them unresolved.

When client puzzles are used, a server accepts connection requests as usual. However, when it suspects that it is under attack, marked by a significant rise in connection requests, it responds to requests by issuing each requestor a puzzle: A hard cryptographic problem based on the current time and on information specific to the server and client request.*

Like hash cash, client puzzles require that the client find some k-bit partial hash collisions. To decrease the chance that a client might just guess the puzzle, each puzzle could optionally be made up of multiple subpuzzles that the client must solve individually. Mathematically, a puzzle is a hash for which a client needs to find the corresponding input that would produce it.†

Nonparallelizable work functions

Both of the hash collision POW systems in the previous section can easily be solved in parallel. In other words, a group of n machines can solve each problem in $1/n$ the amount of time as a single machine. Historically, this situation is like the encryption challenges that were solved relatively quickly by dividing the work among thousands of users.

Parallel solutions may be acceptable from the point of view of accountability. After all, users still pay with the same expected amount of burnt CPU cycles, whether a single machine burns m cycles, or n machines burn m cycles collectively. But if the goal of nonfungible micropayments is to ensure public access to Bob's resources, parallelizable schemes are weak because they can be overwhelmed by distributed denial of service attacks.

Let's actually try to make Alice wait for a fixed period between two transactions, in order to better protect Bob's resources. Consider a "proof of time": we desire a client to spend some amount of time, as opposed to work, to solve a given problem. An example is MIT's LCS35 Time Capsule Crypto-Puzzle. The time cap-

* "RSA Laboratories Unveils Innovative Countermeasure to Recent 'Denial of Service' Hacker Attacks," press release, *http://www.rsasecurity.com/news/pr/000211.html.*

† For example, by breaking a puzzle into eight subpuzzles, you can increase the amount of average work required to solve the puzzle by the same amount as if you left the puzzle whole but increased the size by three bits. However, breaking up the puzzle is much better in terms of making it harder to guess. The chance of correctly guessing the subpuzzle version is 2^{-8k}, while the chance of guessing the larger single version is just $2^{-(k+3)}$, achieved by hashing randomly selected inputs to find a collision without performing a brute-force search.

Microsoft Exhibit 1019

sule, sealed in 1999, will be opened either after 35 years or when the supplied cryptographic problem is solved, whichever comes first. The problem is designed to foil any benefit of parallel or distributed computing. It can be solved only as quickly as the fastest single processor available.

Time-lock puzzles, such as the LCS35 time capsule, were first presented by Ron Rivest, Adi Shamir, and David Wagner.* These types of puzzles are designed to be "intrinsically" or "inherently" sequential in nature. The problem LCS35 used to compute is:

$$2^{2^t} \bmod n$$

where n is the product of two large primes p and q, and t can be arbitrarily chosen to set the difficulty of the puzzle. This puzzle can be solved only by performing t successive squares modulo n. There is no known way to speed up this calculation without knowing the factorization of n. The reason is the same reason conventional computer encryption is hard to break: there is no existing method for finding two primes when only their product is known.

It's worth noting in passing that the previous construction is not *proven* to be nonparallelizable. Besides the product-of-two-primes problem, its security rests on no one knowing how to perform the repeated modular squaring in parallel. This problem is tied up with the "P vs. NC" problem in computational complexity theory and is outside the scope of this chapter. Similar to the better known "P vs. NP" problem, which concerns the question, "Which problems are easy?" the P vs. NC problem asks, "Which problems are parallelizable?"†

Fungible micropayments

All of the micropayment schemes we have previously described are nonfungible. While Alice pays Bob for resource use with some coin that represents a proof of work, he cannot redeem this token for something of value to him. While this micropayment helps prevent DoS and flooding attacks, there's no measure of "wealth" in the system. Bob has no economic incentive to engage in this exchange.

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* Ronald L. Rivest, Adi Shamir, and David A. Wagner (1996), "Time-Lock Puzzles and Timed-Release Crypto."

† Historical note: NC stands for Nick's Class, named after Nicholas Pippenger, one of the first researchers to investigate such problems. For more information, see Raymond Greenlaw, H. James Hoover, and Walter L. Ruzzo (1995), *Limits to Parallel Computation: P-Completeness Theory*. Oxford University Press.

Microsoft Exhibit 1019

Nonfungible micropayments are better suited for ephemeral resources, like TCP connections, than they are for long-term resources like data storage. Consider an attacker who wants to make a distributed datastore unusable. If an attacker is trying to fill up a system's storage capacity and is allowed to store data for a long time, the effects of DoS attacks can be cumulative. This is because the attacker can buy more and more space on the system as time goes on.

If micropayments just use up CPU cycles and cannot be redeemed for something of value, an attacker can slowly nibble at resources, requesting a megabyte now and then as it performs enough work to pay for the space. This can continue, bit by bit, until the attacker controls a large percentage of the total. Furthermore, the victim is unable to use these payments in exchange for other peers' resources, or alternatively to purchase more resources.

Enter redeemable payments. This compensation motivates users to donate resources and fixes the cost for resources in a more stable way.

Freeloading

The history of the Internet records a number of users who have freely donated resources. The hacker ethos and the free software movement can be relied on to provide resources to an extent. Johan Helsingius ran the initial "anonymous" remailer—*anon.penet.fi*—for its own sake. The Cypherpunks (Type I) and Mixmaster (Type II) remailers for anonymous email are run and maintained for free from around the globe. Processing for SETI@home and Distributed.net is also performed without compensation, other than the possibility of fame for finding an alien signature or cracking a key.

Unfortunately, not everybody donates equally. It is tempting for a user to "let somebody else pay for it" and just reap the rewards.

Peer-to-peer systems may combat this effect by incorporating coercive measures into their design or deployment, ensuring that users actually donate resources. This is not a trivial problem. Napster provides a good example: users need to connect to Napster only when actually searching for MP3 files; otherwise they can remain offline. Furthermore, users are not forced to publicly share files, although downloaded files are placed in a public directory by default.

A fairly recent analysis of Gnutella traffic showed a lot of freeloading. One well-known study* found that almost 70% of users share no files, and nearly 50% of all responses are returned by the top 1% of sharing hosts.

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* Eytan Adar and Bernardo A. Huberman, "Free Riding on Gnutella." Xerox Palo Alto Research Center, *http://www.parc.xerox.com/istl/groups/iea/papers/gnutella*.

Microsoft Exhibit 1019

Free Haven tackles this problem by attempting to ensure that users donate resources in amounts proportional to the resources they use. The system relies on accountability via reputation, which we discuss later. Mojo Nation, on the other hand, pays users Mojo—the system's private digital currency—for donated resources. Mojo has no meaning outside the system yet, but it can be leveraged for other system resources.

Fungible payments for accountability

Fungible micropayments are not used solely, or even largely, for economic incentives. Instead, they act as an accountability measure. Peers can't freeload in the system, as they can earn wealth only by making their own resources available (or by purchasing resource tokens via some other means). This is a more natural and more effective way to protect a system from flooding than proofs of work. In order to tie up resources protected by fungible payments, an adversary needs to donate a proportional amount of resources. The attempted denial of service becomes self-defeating.

If payments can actually be redeemed for real-world currencies, they provide yet another defense against resource misuse. It may still be true that powerful organizations (as opposed to a script-kiddie downloading DoS scripts from *http://rootshell.com*) can afford to pay enough money to flood a system. But now the victim can purchase additional physical disk space or bandwidth with the money earned. Since the prices of these computer resources drop weekly, the cost of successfully attacking the system increases with time.

Business arrangements, not technology, link digital cash to real-world currencies. Transactions can be visualized as foreign currency exchanges, because users need to convert an amount of money to digital cash before spending it. The Mark Twain Bank "issued" DigiCash eCash in the U.S. in the mid-1990s, joined by other banks in Switzerland, Germany, Austria, Finland, Norway, Australia, and Japan.* eCash can as easily be used for private currencies lacking real-world counterparts; indeed, Mojo is based on eCash technology (although without, in default form, the blinding operations that provide anonymity). The digital cash schemes we describe, therefore, can be used for both private and real-world currencies.

* "DigiCash Loses U.S. Toehold," CNET news article, *http://www.canada.cnet.com/news/0-1003-200-332852.html.*

Microsoft Exhibit 1019

Micropayment digital cash schemes

Ronald Rivest and Adi Shamir introduced two simple micropayment schemes, PayWord and MicroMint, in 1996.* PayWord is a credit-based scheme based on chains of paywords (hash values), while MicroMint represents coins by *k*-way hash function collisions. Both of these schemes follow the lightweight philosophy of micropayments: nickels and dimes don't matter. If a user loses a payment or is able to forge a few payments, we can ignore such trivialities. The security mechanisms in these schemes are not as strong nor expensive as the full macropayment digital cash schemes we will discuss later. At a rough estimate, hashing is about 100 times faster than RSA signature verification and about 10,000 times faster than RSA signature generation.

PayWord is designed for applications in which users engage in many repetitive micropayment transactions. Some examples are pay-per-use web sites and pay-per-minute online games or movies. PayWord relies on a broker (better known as a "bank" in many digital cash schemes), mainly for online verification, but seeks to minimize communication with the broker in order to make the system as efficient as possible.

It works like this. Alice establishes an account with a broker and is issued a digital certificate. When she communicates with vendor Bob for the first time each day, she computes and commits to a new payword chain w_1, w_2, ..., w_n. This chain is created by choosing some random w_n and moving backward to calculate each hash w_i from the hash w_{i+1}.

Alice starts her relationship with Bob by offering w_0. With each micropayment she moves up the chain from w_0 toward w_n. Just knowing w_0, vendor Bob can't compute any paywords and therefore can't make false claims to the broker. But Bob can easily verify the ith payment if he knows only w_{i-1}. Bob reports to the broker only once at the end of the day, offering the last (highest-indexed) micropayment and the corresponding w_0 received that day. The broker adjusts accounts accordingly.

As payword chains are both user- and vendor-specific, the vendor can immediately determine if the user attempts to double-spend a payword. Unfortunately, however, PayWord does not provide any transaction anonymity. As this is a credit-based system, the broker knows that Alice paid Bob.

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* R. Rivest and A. Shamir (1997), "PayWord and MicroMint: Two Simple Micropayment Schemes," Lecture Notes in Computer Science, vol. 1189, *Proc. Security Protocols Workshop*, Springer-Verlag, pp. 69–87.

Microsoft Exhibit 1019

MicroMint takes the different approach of providing less security, but doing so at a very low cost for unrelated, low-value payments. Earlier, we described k-bit collisions, in which Alice found a value that matched the lowest k bits in Bob's hash. MicroMint coins are represented instead by full collisions: all the bits of the hashes have to be identical. A k-way collision is a set of distinct inputs (x_1, x_2, ..., x_k) that all map to the same digests. In other words, hash(x_1) = hash(x_2) = ... = hash(x_k). These collisions are hard to find, as the hash functions are specifically designed to be collision-free!*

The security in MicroMint rests in an economy of scale: minting individual coins is difficult, yet once some threshold of calculations has been performed, coins can be minted with accelerated ease. Therefore, the broker can mint coins cost-effectively, while small-scale forgers can produce coins only at a cost exceeding their value.

As in PayWord, the MicroMint broker knows both Alice, to whom the coins are issued, and vendor Bob. This system is therefore not anonymous, allowing the broker to catch Alice if she attempts to double-spend a coin.

PayWord and MicroMint are just two representative examples of micropayment schemes. Many others exist. The point to notice in both schemes is the extreme ease of verification and the small space requirements for each coin. Not only are these schemes fast, but they remain fast even in environments with severe resource constraints or at larger amounts of money.

Micropayment schemes such as these make it possible to extend peer-to-peer applications beyond the desktop and into embedded and mobile environments. Routers can use micropayments to retard denial of service attacks with minimal extra computation and then store the proceeds. Music players can act as mobile Napster or Gnutella servers, creating ad hoc local networks to exchange music stored in their memories (and possibly make money in the process). These possibilities are just beginning to be explored.

Making money off others' work

Proofs of work can be exchanged for other resources in a system, even without a systemwide digital cash scheme. The key is to make a POW scheme in which Bob can take a POW submitted by Alice and pass it on to Charlie without expending any significant calculation of his own.

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* Given a hash function with an n-bit output (e.g., for SHA-1, n=160), we must hash approximately $2^{n(k-1)/k}$ random strings in order to find a k-way collision. This follows from the "birthday paradox" as explained in Rivest and Shamir, *ibid.*

Microsoft Exhibit 1019

This scheme was introduced by Markus Jakobsson and Ari Juels in 1999 as a *bread pudding protocol.** Bread pudding is a dish that originated with the purpose of reusing stale bread. In a similar spirit, this protocol defines a POW that may be reused for a separate, useful, and verifiably correct computation. This computation is not restricted to any specific problem, although the authors further specify a simple bread pudding protocol for minting MicroMint coins.

In this variant on MicroMint's original minting scheme, the broker no longer has to generate each individual payment made by each user. Instead, a single, valid coin can be redistributed by users in the system to satisfy each others' POWs. The fundamental idea is to make clients solve partial hash collisions, similar to the concept of hash cash. This computation is useful only to the mint, which holds some necessary secret. With enough of these POWs, the minter can offload the majority of computations necessary to minting a coin.

Effectively, Bob is asking Alice to compute part of a MicroMint coin, but this partial coin is useful only when combined with thousands or millions of other similar computations. Bob collects all of these computations and combines them into MicroMint coins. Without requiring systemwide fungible digital cash, Bob can reuse others' computation work for computations of value to him (and only to him). The scheme is extensible and can potentially be used with many different types of payment schemes, not just MicroMint.

Anonymous macropayment digital cash schemes

Up until now, we have described payments in which the value of each coin or token is fairly small. These make forgery difficult because it's useful only if it can be performed on a large scale. Now we will move to more complex schemes that allow large sums to be paid in a secure manner in a single transaction. These schemes also offer multiparty security and some protection for user privacy.

The macropayment digital cash schemes we are about to discuss offer stronger security and anonymity. However, these protections come at a cost. The computational and size requirements of such digital cash are much greater. In cryptographic literature, micropayments are usually considered extremely small (such as $0.01) and use very efficient primitives such as hash functions. In contrast, macropayment digital cash schemes use public key operations, such as exponentiation, which are much slower. The use of techniques from elliptic curve cryp-

* Markus Jakobsson and Ari Juels (1999), "Proofs and Work and Bread Pudding Protocols." In B. Preneel, ed., *Communications and Multimedia Security*. Kluwer Academic Publishers, pp. 258–272.

Microsoft Exhibit 1019

tography can alleviate this problem by making it possible to securely use much shorter keys.

Other clever tricks, such as "probabilistic" checking—checking selected payments on the grounds that large-scale forgery will be caught eventually—can help macropayment techniques compete with micropayment schemes. This is an active research area and a source of continual innovation. Macropayment schemes will prove useful when used with the reputation systems discussed later in this chapter in the section "Reputations," because reputation systems let us make large transactions without assuming incommensurate risk.

Pioneering work on the theoretical foundations of anonymous digital cash was carried out by David Chaum in the early 1980s.* Chaum held patents on his electronic cash system, and founded DigiCash in 1990 to implement his ideas, but he exclusively licensed his patents to Ecash Technologies in 1999.

The electronic cash system he developed is based on an extension of digital signatures, called blind signatures. A digital signature uses a PKI to authenticate that a particular message was sent by a particular person. (See Chapter 15 for a greater description of signatures and PKI.) A *blind* signature scheme allows a person to get a message signed by another party, while not revealing the message contents to that party or allowing the party to recognize the message later.

In digital cash, Alice creates some number called a *proto-coin* and "blinds" it by multiplying by a secret random number. She sends this blinded proto-coin to the bank, which cannot link it with the original proto-coin. The bank checks that she has a positive balance and signs the proto-coin with the assurance that "this is a valid coin," using a private key specific to the particular amount of money in the coin. The bank returns this to Alice and subtracts the corresponding amount from Alice's bank account. Alice then divides out the random number multiplier and finds herself with a properly minted coin, carrying a valid signature from the bank. This is just one way to do digital cash, but it will suffice for this discussion.

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* D. Chaum (1983), "Blind Signatures for Untraceable Payments," *Advances in Cryptology—Crypto '82*, Springer-Verlag, pp. 199–203. D. Chaum (1985), "Security Without Identification: Transaction Systems to Make Big Brother Obsolete," *Communications of the ACM*, vol. 28, no. 10, pp. 1030–1044. D. Chaum, A. Fiat, and M. Naor (1988), "Untraceable Electronic Cash," *Advances in Cryptology—Crypto '88, Lecture Notes in Computer Science*, no. 403, Springer-Verlag, pp. 319–327. D. Chaum (August 1992), "Achieving Electronic Privacy" (invited), *Scientific American*, pp. 96–101, *http://www.chaum.com/articles/Achieving_Electronic_Privacy.htm*.

Microsoft Exhibit 1019

In real life, the digital cash transaction would be like Alice slipping a piece of paper into a carbon-lined envelope, representing the blinded proto-coin. The bank just writes its signature across the envelope, which imprints a carbon signature onto the inside paper. Alice removes the paper and is left with a valid coin.

Alice can then spend this coin with Bob. Before accepting it, Bob needs to check with the issuing bank that the coin hasn't already been spent, a process of online verification. Afterwards, Bob can deposit the coin with the bank, which has no record of to whom that coin was issued. It saw only the blinded proto-coin, not the underlying "serial" number.

This digital cash system is both anonymous and untraceable. Its disadvantage, however, is that coins need to be verified online during the transaction to prevent double-spending. This slows down each transaction.

Stefan Brands proposed an alternative digital cash scheme in 1993.* It forms the core of a system implemented and tested by CAFE, a European project with both academic and commercial members. His patents are currently held by the Montreal-based privacy company Zero-Knowledge Systems, Inc.

Brands's digital cash scheme allows offline checking of double-spending for fraud tracing, with obvious performance benefits compared to online verification. It is also well suited for incorporation into smart cards, and the underlying technology provides an entire framework for privacy-protecting credential systems.

Brands's scheme uses a restrictive blind signature protocol rather than a normal blind signature protocol as proposed by Chaum. In the digital cash context, this certificate is a valid coin, represented as three values—secret key, public key, and digital signature—certified by the bank. A key aspect of this protocol is that the bank knows—and encodes attributes into—*part* of Alice's secret key, but it has no idea what the corresponding public key and certificate look like (except that they are consistent with the known part of the secret key). At the end of the issuing protocol, Alice uses techniques somewhat similar to Chaum's protocol to generate a valid coin.

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* Stefan Brands (1993), "Untraceable Off-Line Cash in Wallet with Observers," *Advances in Cryptology—Crypto '93*, *Lecture Notes in Computer Science*, no. 773, Springer-Verlag, pp. 302–318. Stefan Brands (2000), *Rethinking Public Key Infrastructures and Digital Certificates: Building in Privacy*. MIT Press. Stefan Brands, online book chapter from *Rethinking Public Key Infrastructures and Digital Certificates: Building in Privacy*, *http://www.xs4all.nl/~brands/cash.html*.

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Payment is very different in Brands's system. Alice's coin contains her secret key, so she doesn't actually give her coin to the vendor Bob. Instead, she proves to Bob that she is the proper holder of that coin (that is, that she knows the secret key associated with the public key), without actually disclosing its contents. This type of payment, a signed proof of knowledge transcript, is a fundamental shift from Chaum's e-cash model, in which the coin is merely an "immutable" bit string.

Privacy is maintained together with honesty by Brands's system in a very clever manner. If Alice only spends the coin once, the bank can gain no information as to her identity. After all, during the issuing protocol, the bank saw only part of Alice's secret key, not the public key used to verify Alice's payment transcript signature. Nor did the bank see its own signature on that public key. Yet if Alice double-spends the coin, the bank can use it to extract her identity!

We won't provide the math necessary to understand the security in this system, but you can understand why it works by comparing it to a Cartesian x-y plane. The first random number challenge used during payment provides one point (x_0,y_0) on this plane. An infinite number of lines can pass through this one point. If Alice uses the same coin to pay Charlie, a different random number is used. Now we know a second (x_1,y_1) point, and two points uniquely define a line. In the same way, Alice's identity will be exposed if she spends the coin twice.

Brands's scheme—useful for both digital cash and credentials—can be used to encode other useful information, such as negotiable attributes exposed during payment or high-value secrets that can prevent lending. A "high-value secret" refers to the same strategy we discussed when trying to prevent people from sharing their accounts—if a certificate to do X includes the user's credit card number, the user will be less willing to loan the certificate to others.

The "negotiable attributes" are an extension of a powerful idea—that of "credentials without identity." If you have a credential without identity, you have a way of proving that you belong to a certain class of people without actually having to prove anything about who you are. For example, you may have a credential which certifies that you are over 21 but doesn't include your name. The entity that authorized your age wouldn't want you to be able to lend this certificate to someone else. Therefore, we utilize these high-value secrets: the user needs to know the secret in order to use the over-21 certificate. Brands's scheme takes this farther and allows you to selectively reveal or hide various certifications on the fly, thereby allowing you to negotiate your degree of privacy.

Microsoft Exhibit 1019

One final note: whether a peer-to-peer system uses micropayments or macropayments, system designers must consider the possibility that these can be DoS targets in themselves. Perhaps an attacker can flood a system with cheaply minted counterfeit coins, eating up computational resources through verification-checking alone. The extent of this problem depends largely on the computational complexity of coin verification. Many of the systems we describe—hash cash, client puzzles, MicroMint, PayWord—can very quickly verify coins with only a single or a few hash operations. Public key operations, such as modular exponentiation, can be significantly more expensive. Again, digital cash schemes are an active area of cryptographic research; before specifying a scheme it is worth checking out the proceedings of the Financial Cryptography, CRYPTO, and EUROCRYPT conferences.

The use and effectiveness of micropayments in peer-to-peer systems

So far, we have spent quite a bit of time describing various micropayment and digital cash schemes. Our discussion is not meant as exhaustive, yet it provides some examples of various cryptographic primitives and technologies used for electronic cash: public key encryption, hash functions, digital signatures, certificates, blinding functions, proofs of knowledge, and different one-way and trap door problems based on complexity theory. The list reads like a cryptographic cookbook. Indeed, the theoretical foundations of digital cash—and the design of systems—have been actively researched and developed over the past two decades.

Only in the past few years, however, have we begun to see the real-world application of micropayments and digital cash, spurred by the growth of the Internet into a ubiquitous platform for connectivity, collaboration, and even commerce. Electronic cash surely has a place in future society. But its place is not yet secured. We are not going to try to predict either how fast or how widespread its adoption will be; that depends on too many economic, social, and institutional factors.

Instead, we'll focus on how micropayments might be useful for peer-to-peer systems: what issues the developers of peer-to-peer systems need to consider, when certain digital cash technologies are better than others, how to tell whether the micropayments are working, and how to achieve stronger or weaker protections as needed.

Microsoft Exhibit 1019

Identity-based payment policies

When designing a policy for accepting micropayments, a peer-to-peer system might wish to charge a varying amount to peers based on identity. For instance, a particular peer might charge less to local users, specified friends, users from academic or noncommercial subnets, or users within specified jurisdictional areas.

Such policies, of course, depend on being able to securely identify peers in the system. This can be hard to do both on the Internet as a whole (where domain names and IP addresses are routinely spoofed) and, in particular, on systems that allow anonymity or pseudonymity. Chapter 15 discusses this issue from several general angles, and Chapter 12 discusses how we try to solve the problem in one particular pseudonymous system, Free Haven. Many other systems, like ICQ and other instant messaging services, use their own naming schemes and ensure some security through passwords and central servers. Finally, systems with many far-flung peers need a reputation system to give some assurance that peers won't abuse their privileges.

General considerations in an economic analysis of micropayment design

Designers choosing a micropayment scheme for a peer-to-peer system should not consider merely the technical and implementation issues of different micropayment schemes, but also the overall economic impact of the entire system. Different micropayment schemes have different economic implications.

A classic economic analysis of bridge tolls serves as a good analogy for a peer-to-peer system. In 1842, the French engineer Jules Dupuit reached a major breakthrough in economic theory by arguing the following: the economically efficient toll on an uncongested bridge is zero, because the extra cost from one more person crossing the bridge is zero. Although bridge building and maintenance is not free—it costs some money to the owner—it is socially inefficient to extract this money through a toll. Society as a whole is worse off because some people choose not to cross due to this toll, even though their crossing would cost the owner nothing, and they would not interfere with anyone else's crossing (because the bridge is uncongested). Therefore, everyone should be allowed to cross the bridge for free, and the society should compensate the bridge builder through a lump-sum payment.*

* Arsene Jules Etienne Dupuit (1842), "On Tolls and Transport Charges," *Annales des Ponts et Chaussees*, trans. 1962, IEP.

Microsoft Exhibit 1019

This bridge example serves as a good analogy to a peer-to-peer system with micropayments. Tolls should be extracted only in order to limit congestion and to regulate access to people who value crossing the most. Given the same economic argument, resource allocation to limit congestion is the only justifiable reason for micropayments in a peer-to-peer system. Indeed, excessive micropayments can dissuade users from using the system, with negative consequences (known as "social inefficiencies") for the whole society. Users might not access certain content, engage in e-commerce, or anonymously publish information that exposes nefarious corporate behavior.

This analysis highlights the ability of micropayments to prevent attacks and adds the implied ability to manage congestion. Congestion management is a large research area in networking. Micropayments can play a useful role in peer-to-peer systems by helping peers prioritize their use of network bandwidth or access to storage space. Users who really want access will pay accordingly. Of course, such a system favors wealthy users, so it should be balanced against the goal of providing a system with the broadest social benefits. Reputation can also play a role in prioritizing resource allocation.

Most economic research relevant for micropayments has focused on owner-side strategies for maximizing profit. AT&T researchers compared flat-fee versus pay-per-use fee methods where the owner is a monopolist and concluded that more revenues are generated with a flat-fee model. Similar research at MIT and NYU independently concluded that content bundling and fixed fees can generate greater profits per good.*

We try to take a broader view. We have to consider the well-being of all economic agents participating in and affected by the system. Three general groups come to mind in the case of a peer-to-peer system: The owner of the system, the participants, and the rest of society.

How does a micropayment scheme impact these three agents? Participants face direct benefits and costs. The owner can collect fees or commissions to cover the fixed cost of designing the system and the variable costs of its operation. The rest of society can benefit indirectly by synergies made possible by the system, knowledge spillovers, alternative common resources that it frees up, and so on.

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* P. C. Fishburn and A. M. Odlyzko (1999), "Competitive Pricing of Information Goods: Subscription Pricing Versus Pay-Per-Use," *Economic Theory*, vol. 13, pp. 447–470, *http://www.research.att.com/~amo/doc/competitive.pricing.ps*. Y. Bakos and E. Brynjolfsson (December 1999), "Bundling Information Goods: Pricing, Profits, and Efficiency," *Management Science*, *http://www.stern.nyu.edu/~bakos/big.pdf*.

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To simplify our discussion, we assume that whatever benefits participants also benefits society. Furthermore, we can realistically assume a competitive market, so that the owner is probably best off serving the participants as well as possible. Therefore, we focus on the cost/benefit analysis for the participant.

We believe that a focus on costs and benefits to participants is more suited to the peer-to-peer market than the literature on information services, for several reasons. First, peer-to-peer system owners do not enjoy the luxury of dictating exchange terms, thanks to competition. Second, nonfungible micropayments do not generate revenues for the owner; it is not always worthwhile to even consider the benefit to the owner. Third, we expect that a large amount of resources in peer-to-peer systems will be donated by users: people donate otherwise unused CPU cycles to SETI@home calculations, unused bandwidth to forward Mixmaster anonymous email, and unused storage for Free Haven data shares. For these situations, the sole role of micropayments is to achieve optimal resource allocation among participants. In other words, micropayments in a peer-to-peer system should be used only to prevent congestion, where this concept covers both bandwidth and storage limitations.

Moderating security levels: An accountability slider

Poor protection of resources can hinder the use of a peer-to-peer system on one side; attempts to guard resources by imposing prohibitive costs can harm it on the other. Providing a widely used, highly available, stable peer-to-peer system requires a balance.

If a peer-to-peer system wishes only to prevent query-flooding (bandwidth) attacks, the congestion management approach taken by client puzzles—and usable with any form of micropayment—answers the problem. Query-flooding attacks are transient; once the adversary stops actively attacking the system, the bandwidth is readily available to others.

As we have suggested, other forms of congestion are cumulative, such as those related to storage space. Free Haven seeks "document permanence," whereby peers promise to store data for a given time period (although the Free Haven trading protocol seeks to keep this system dynamic, as discussed in Chapter 12). If we wait until the system is already full before charging micropayments, we've already lost our chance to adequately protect against congestion.

This is the freeloading problem: we wish to prevent parasitic users from congesting the system without offering something of value in return. Furthermore, an adversary can attempt to flood the system early to fill up all available space. Therefore, for systems in which resource pressures accrue cumulatively,

Microsoft Exhibit 1019

micropayments should always be required. Free Haven's answer is to require that peers offer storage space proportional to that which they take up. (Even though cash-based micropayments are not used, the idea of payment by equivalent resources is similar.)

The alternative to this design is the caching approach taken by systems such as Freenet. Old data is dropped as newer and more "popular" data arrives. This approach does not remove the resource allocation problem, however; it only changes the issue. First, flooding the system can flush desirable data from nearby caches as well. System designers simply try to ensure that flooding will not congest the resources of more distant peers. Second, freeloading is not as much of a concern, because peers are not responsible for offering best-effort availability to documents. However, if many peers rely on a few peers to store data, only the most popular data remains. Social inefficiencies develop if the system loses data that could be desired in the long run because short-term storage is insufficient to handle the requirements of freeloading peers. Furthermore, disk space is only one of several resources to consider. Massive freeloading can also affect scalability through network congestion.

Always charging for resources can prevent both freeloading and attacks. On the other hand, excessive charges are disadvantageous in their own right. So it would be useful to find a balance.

Consider an *accountability slider*: Peers negotiate how much payment is required for a resource within a general model specified by the overall peer-to-peer system. Using only a micropayment model, systems like Free Haven or Publius may not have much leeway. Others, like Freenet, Gnutella, or Jabber, have notably more. When we later add the concept of reputation, this accounting process becomes even more flexible.

Each of the micropayment schemes described earlier in this chapter can be adapted to provide a sliding scale:

Hash cash
: Partial hashes can be made arbitrarily expensive to compute by choosing the desired number of bits of collision, denoted by k. No matter how big k gets, systems providing the resources can verify the requests almost instantly.

Client puzzles
: The work factor of these puzzles is also based on the number of bit collisions. The number of subpuzzles can also be increased to limit the possibility of random guessing being successful; this is especially important when k becomes smaller.

Microsoft Exhibit 1019

Time puzzles

The LCS35 time-lock includes a parameter t that sets the difficulty of the puzzle.

MicroMint

The "cost" of a coin is determined by its number of colliding hash values. The greater the k-way collision, the harder the coin is to generate.

PayWord

In the simplest adaptation, a commitment within PayWord can be a promise of varying amount. However, PayWord is designed for iterative payments. To be able to use the same PayWord chain for successive transactions, we want to always pay with coins of the same value. Therefore, to handle variable costs, we can just send several paywords for one transaction. The very lightweight cost of creating and verifying paywords (a single hash per payword) also makes this multiple-coin approach suitable for macropayment schemes.

Anonymous digital cash

Coins can have different denominations. In Chaum's design, the bank uses a different public key to sign the coin for different denominations. In Brands's model, the denomination of the coin is encoded within the coin's attributes; the bank's public key is unique to currency, not denomination. Brands's digital cash system also allows negotiable attributes to be revealed or kept secret during payment. This additional information can play a role in setting the accountability slider.

By negotiating these various values, we can change the level of accountability and security offered by a peer-to-peer system. Based on overall system requirements, this process can be fixed by the system designers, changed by the administrators of individual peer machines, or even fluctuate between acceptable levels at runtime!

While payment schemes can be used in a variety of peer-to-peer situations—ranging from systems in which peers are fully identified to systems in which peers are fully anonymous—they do involve some risk. Whenever Alice makes an electronic payment, she accepts some risk that Bob will not fulfill his bargain. When identities are known, we can rely on existing legal or social mechanisms. In fully anonymous systems, no such guarantee is made, so Alice attempts to minimize her risk at any given time by making small, incremental micropayments. However, there is another possibility for pseudonymous systems, or identified systems for which legal mechanisms should only be used as a last resort: reputation schemes. In this next section, we consider the problem of reputation in greater depth.

Microsoft Exhibit 1019

Reputations

While micropayments provide an excellent mechanism for anonymous exchange, a number of systems call for more long-term pseudonymous or even public relationships. For instance, in the case of transactions in which one party promises a service over a long period of time (such as storing a document for three years), a simple one-time payment generally makes one party in the transaction vulnerable to being cheated. A whistleblower or political dissident who publishes a document may not wish to monitor the availability of this document and make a number of incremental micropayments over the course of several years, since this requires periodic network access and since continued micropayments might compromise anonymity. (While third-party escrow monitoring services or third-party document sponsors might help to solve this issue, they introduce their own problems.) In addition, some systems might want to base decisions on the observed behavior of entities—how well they actually perform—rather than simply how many resources they can provide.

In the real world, we make use of information about users to help distribute resources and avoid poor results. Back before the days of ubiquitous communication and fast travel, doing business over long distances was a major problem. Massive amounts of risk were involved in, say, sending a ship from Europe to Asia for trade. Reputations helped make this risk bearable; large banks could issue letters of credit that could draw on the bank's good name both in Europe and Asia, and they could insure expeditions against loss. As the bank successfully helped expeditions finance their voyages, the bank's reputation grew, and its power along with it. Today's business relationships still follow the same path: two parties make a decision to trust each other based on the reputations involved.

While the online world is different from the brick-and-mortar world, it too has benefited from reputations—and can continue to benefit.

The main difference between reputation-based trust systems and micropayment-based trust systems is that, in reputation-based trust systems, parties base their decisions in part on information provided by third parties. Peers are motivated to remain honest by fear that news of misdealings will reach yet other third parties.

Reputation systems are not useful in all situations. For instance, if there were thousands of buyers and one or two vendors, being able to track vendor performance and reliability would not help buyers pick a good vendor. On the other hand, tracking performance might provide feedback to the vendor itself on areas in which it might need improvement. This in turn could result in better performance down the road, but only if the vendor acted on this feedback.

Microsoft Exhibit 1019

Similarly, in fields in which a given buyer generally doesn't perform transactions frequently, the benefits of a reputation system are more subtle. A buyer might benefit from a real estate reputation system, since she expects to rent from different people over time. Even for a domain in which she expects to do just one transaction over her whole lifetime (such as laser eye surgery), she would probably contribute to a reputation site—first out of altruism, and second in order to give the surgeon an incentive to do well.

This is the tragedy of the commons in reverse: when the cost of contributing to a system is low enough, people will contribute to it for reasons not *immediately* beneficial to themselves.

Chapter 17, *Reputation*, describes some of the practical uses for a reputation system and the difficulties of developing such a system. That chapter focuses on the solution found at Reputation Technologies, Inc. In this chapter we'll give some background on reputation and issues to consider when developing a reputation system.

Early reputation systems online

The online world carries over many kinds of reputation from the real world. The name "Dennis Ritchie" is still recognizable, whether listed in a phone book or in a posting to *comp.theory*. Of course, there is a problem online—how can you be sure that the person posting to *comp.theory* is *the* Dennis Ritchie? And what happens when "Night Avenger" turns out to be Brian Kernighan posting under a pseudonym? These problems of identity—covered earlier in this chapter—complicate the ways we think about reputations, because some of our old assumptions no longer hold.

In addition, new ways of developing reputations evolve online. In the bulletin board systems of the 1980s and early 1990s, one of the more important pieces of data about a particular user was her upload/download ratio. Users with particularly low ratios were derided as "leeches," because they consumed scarce system resources (remember, when one user was on via a phone line, no one else could log in) without giving anything in return. As we will see, making use of this data in an automated fashion is a promising strategy for providing accountability in peer-to-peer systems.

Codifying reputation on a wide scale: The PGP web of trust

Human beings reason about reputations all the time. A large-scale peer-to-peer application, however, cannot depend on human judgments for more than a negligible portion of its decisions if it has any hope of scalability. Therefore, the

Microsoft Exhibit 1019

next step in using reputations is to make their development and consequences automatic.

We've already mentioned the value of knowing upload/download ratios in bulletin board systems. In many systems, gathering this data was automatic. In some cases, the consequences were automatic as well: drop below a certain level and your downloading privileges would be restricted or cut off entirely. Unfortunately, these statistics did not carry over from one BBS to another—certainly not in any organized way—so they provided for reputations only on a microscopic scale.

One of the first applications to handle reputations in an automated fashion on a genuinely large scale was the "web of trust" system introduced in Phil Zimmermann's Pretty Good Privacy (PGP). This was the first program to bring public key cryptography to the masses. In public key cryptography, there are two keys per user. One is public and can be used only to encrypt messages. The other key is private and can be used only to decrypt messages. A user publishes his public key and keeps the private key safe. Then others can use the public key to send him messages that only he can read.

With public key cryptography comes the key certification problem, a type of reputation issue. Reputations are necessary because there is no way to tell from the key alone which public key belongs to which person.

For example, suppose Alice would like people to be able to send encrypted messages to her. She creates a key and posts it with the name "Alice." Unbeknownst to her, Carol has also made up a key with the name "Alice" and posted it in the same place. When Bob wants to send a message to Alice, which key does he choose? This happens in real life; as an extreme example, the name "Bill Gates" is currently associated with dozens of different PGP keys available from popular PGP key servers.

So the key certification problem in PGP (and other public key services) consists of verifying that a particular public key really does belong to the entity to whom it "should" belong. PGP uses a system called a web of trust to combat this problem. Alice's key may have one or more certifications that say "Such and such person believes that this key belongs to Alice." These certifications exist because Alice knows these people personally; they have established to their satisfaction that Alice really does own this key. Carol's fake "Alice" key has no such certifications, because it was made up on the spot.

When Bob looks at the key, his copy of PGP will assign it a trust level based on how many of the certifications are made by people he knows. The higher the trust level, the more confidence Bob can have in using the key.

Microsoft Exhibit 1019

A perennial question about the web of trust, however, is whether or not it scales. Small groups of people can create a web of trust easily, especially if they can meet each other in person. What happens when we try to make the web of trust work for, say, a consumer and a merchant who have never met before? The conventional wisdom is that the web of trust does not scale. After all, there is a limit to how many people Alice and Bob can know.

The most frequently cited alternative to the web of trust is a so-called Public Key Infrastructure. Some trusted root party issues certificates for keys in the system, some of which go to parties that can issue certificates in turn. The result is to create a certification tree. An example is the certificate system used for SSL web browsers; here VeriSign is one of the trusted root certificate authorities responsible for ensuring that every public key belongs to the "right" entity. A hierarchical system has its own problems (not least the fact that compromise of the root key, as may recently have happened to Sun Microsystems,* is catastrophic), but at least it scales, right?

As it turns out, the web of trust may not be as unworkable as conventional wisdom suggests. A study of publicly available PGP keys in 1997 showed that on average, only six certificates linked one key to another.† This "six degrees of separation" or "small-world" effect (also discussed in Chapter 14) means that for a merchant and a user who are both good web of trust citizens—meaning that they certify others' keys and are in turn certified—the odds are good that they will have reason to trust each others' keys. In current practice, however, most commercial sites elect to go with VeriSign. The one major commercial exception is Thawte's Freemail Web of Trust system.‡

A more serious problem with PGP's implementation of the web of trust, however, comes with key revocation. How do you tell everyone that your key is no longer valid? How do you tell everyone that your certificate on a key should be changed? For that matter, what exactly did Bob mean when he certified Charlie's key, and does Charlie mean the same thing when he certifies David's key?

A more sophisticated—but still distributed—approach to trust management that tries to settle these questions is the Rivest and Lampson Simple Distributed

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* Sun Security Bulletin 198, "Revocation of Sun Microsystems Browser Certificates," "How to Detect or Remove the Invalid Certificate," *http://sunsolve5.sun.com/secbull/certificate_howto.html*. Computer Emergency Response Team Bulletin CA-2000-19, *http://www.cert.org/advisories/CA-2000-19.html*.

† Neal McBurnett, "PGP Web of Trust Statistics," *http://bcn.boulder.co.us/~neal/pgpstat*.

‡ Thawte, "Personal Certificates for Web and Mail: The Web of Trust," *http://www.thawte.com/certs/personal/wot/about.html*.

Microsoft Exhibit 1019

Security Infrastructure/Simple Public Key Infrastructure (SDSI/SPKI). A thorough comparison of this with PGP's web of trust and PKI systems is given by Yang, Brown, and Newmarch.*

All of this brings up many issues of trust and public key semantics, about which we refer to Khare and Rifkin.† The point we're interested in here is the way in which the web of trust depends on reputation to extend trust to new parties.

Who will moderate the moderators: Slashdot

The news site Slashdot.org is a very popular news service that attracts a particular kind of "Slashdot reader"—lots of them. Each and every Slashdot reader is capable of posting comments on Slashdot news stories, and sometimes it seems like each and every one actually does. Reputations based on this interaction can help a user figure out which of the many comments are worth reading.

To help readers wade through the resulting mass of comments, Slashdot has a moderation system for postings. Certain users of the system are picked to become moderators. Moderators can assign scores to postings and posters. These scores are then aggregated and can be used to tweak a user's view of the posts depending on a user's preferences. For example, a user can request to see no posts rated lower than 2.

The Slashdot moderation system is one existing example of a partially automated reputation system. Ratings are entered by hand, using trusted human moderators, but then these ratings are collected, aggregated, and displayed in an automatic fashion.

Although moderation on Slashdot serves the needs of many of its readers, there are many complaints that a posting was rated too high or too low. It is probably the best that can be done without trying to maintain reputations for moderators themselves.

Reputations worth real money: eBay

The eBay feedback system is another example of a reputation service in practice. Because buyers and sellers on eBay usually have never met each other, neither has much reason to believe that the other will follow through on their part of the deal. They need to decide whether or not to trust each other.

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* Yinan Yang, Lawrie Brown, and Jan Newmarch, "Issues of Trust in Digital Signature Certificates," *http://www.cs.adfa.oz.au/~yany97/auug98.html*.

† Rohit Khare and Adam Rifkin, "Weaving a Web of Trust," *http://www.cs.caltech.edu/~adam/papers/trust.html*.

Microsoft Exhibit 1019

To help them make the decision, eBay collects feedback about each eBay participant in the form of ratings and comments. After a trade, eBay users are encouraged to post feedback about the trade and rate their trading partner. Good trades, in which the buyer and seller promptly exchange money for item, yield good feedback for both parties. Bad trades, in which one party fails to come through, yield bad feedback which goes into the eBay record. All this feedback can be seen by someone considering a trade.

The idea is that such information will lead to more good trades and fewer bad trades—which translates directly into more and better business for eBay. As we will see, this isn't always the case in practice. It is the case often enough, however, to give eBay a reputation of its own as the preeminent web auction site.

A reputation system that resists pseudospoofing: Advogato

Another example of reputations at work is the "trust metric" implemented at *http://www.advogato.org*, which is a portal for open source development work. The reputation system is aimed at answering the fundamental question, "How much can you trust code from person X?" This question is critical for people working on open source projects, who may have limited time to audit contributed code. In addition, in an open source project, attempts by one contributor to fix the perceived "mistakes" of another contributor may lead to flame wars that destroy projects. As of this writing, the open source development site *http://www.sourceforge.net* (host to Freenet) is considering using a similar reputation system.

The stakes at Advogato are higher than they are at Slashdot. If the Slashdot moderation system fails, a user sees stupid posts or misses something important. If the Advogato trust metric incorrectly certifies a potential volunteer as competent when he or she is not, a software project may fail. At least, this would be the case if people depended on the trust metric to find and contact free software volunteers. In practice, Advogato's trust metric is used mostly for the same application as Slashdot's: screening out stupid posts.

The method of determining trust at Advogato also contains features that distinguish it from a simple rating system like Slashdot moderation. In particular, the Advogato trust metric resists a scenario in which many people join the system with the express purpose of boosting each others' reputation scores.*

Advogato considers trust relationships as a directed flow graph. That is, trust relationships are represented by a collection of nodes, edges, and weights. The

* Raph Levien, "Advogato's Trust Metric," *http://www.advogato.org/trust-metric.html*.

Microsoft Exhibit 1019

system is illustrated in Figure 16-1 (we omit weights for simplicity). The nodes are the people involved. An edge exists between A and B if A trusts B. The weight is a measure of how much A trusts B.

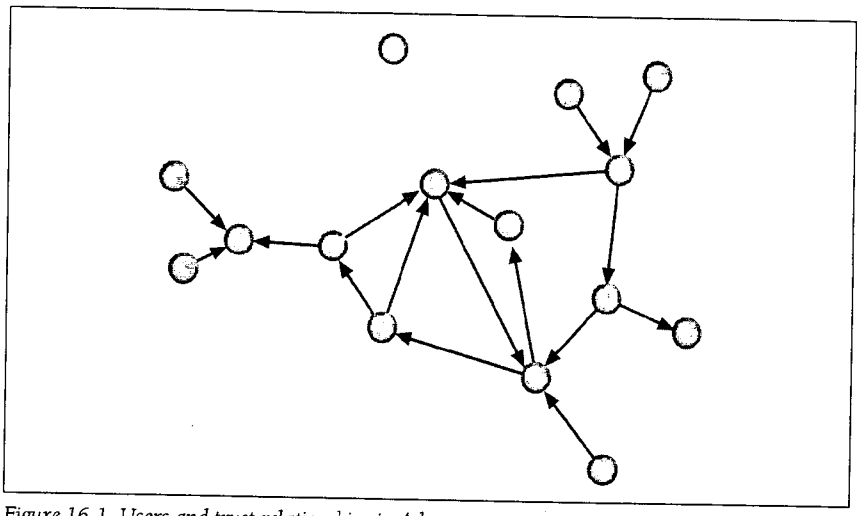

Figure 16-1. Users and trust relationships in Advogato

What we are interested in, however, is not just how much A trusts B, but how much B is trusted in general. So Advogato measures how much trust "flows to" B, by designating a few special trusted accounts as a source and by designating B as a sink. It then defines a flow of trust from the source to B as a path from the source to B. Advogato assigns numbers to edges on the path that are less than or equal to the edge weights. The edge weights act as constraints on how much trust can be "pushed" between two points on the flow path. Finally, the trust value of B is defined as the maximum amount of trust that can be pushed from the source to B—the maximum flow.

Calculating flows through networks is a classic problem in computer science. Advogato uses this history in two ways. First, the Ford-Fulkerson algorithm lets the system easily find the maximum flow, so B's trust value can always be computed.* Second, a result called the "maxflow-mincut theorem" shows that the Advogato system resists the pseudospoofing attacks described earlier, in the sec-

* Thomas H. Cormen, Charles E. Leiserson, and Ronald L. Rivest, *Introduction to Algorithms* (MIT Press and McGraw-Hill, Cambridge, MA, 1990).

Microsoft Exhibit 1019

tion "Problems with pseudospoofing and possible defenses." Even if one entity joins under many different assumed names, none of these names will gain very much more trust than if each had joined alone.

Pseudospoofing is resisted because each of the new names, at least at first, is connected only to itself in the graph. No one else has any reason whatsoever to trust it. Therefore, there is no trust flow from the source to any of the pseudospoofing nodes, and none of them are trusted at all. Even after the pseudospoofing nodes begin to form connections with the rest of the graph, there will still be "trust bottlenecks" that limit the amount of trust arriving at any of the pseudospoofing nodes.

This property is actually quite remarkable. No matter how many fake names an adversary uses, it is unable to boost its trust rating very much. The downside is that nodes "close" to the source must be careful to trust wisely. In addition, it may not be readily apparent what kinds of weights constitute high trust without knowing what the entire graph looks like.

System successes and failures

Reputation in the brick-and-mortar world seems to work quite well; spectacular failures, such as the destruction of Barings Bank by the actions of a rogue trader, are exceptions rather than the rule. Which reputation-based systems have worked online, and how well have they worked?

The Slashdot and Advogato moderation systems seem to work. While it is difficult to quantify what "working" means, there have been no spectacular failures so far. On the other hand, the eBay fraud of mid-2000* shows some of the problems with reputation systems used naively.

In the eBay case, a group of people engaged in auctions and behaved well. As a result, their trust ratings went up. Once their trust ratings were sufficiently high to engage in high-value deals, the group suddenly "turned evil and cashed out." That is, they used their reputations to start auctions for high-priced items, received payment for those items, and then disappeared, leaving dozens of eBay users holding the bag.

As for PGP's web of trust, it has been overtaken by hierarchical PKIs, like those offered by VeriSign, as a widespread means of certifying public keys. In this case, peer-to-peer did not automatically translate into success.

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* "eBay, Authorities Probe Fraud Allegations," CNET news article, *http://www.canada.cnet.com/news/0-1007-200-1592233.html.*

Microsoft Exhibit 1019

Scoring systems

Reputation systems depend on *scores* to provide some meaning to the ratings as a whole. As shown in Chapter 17, scores can be very simple or involve multiple scales and complicated calculations.

In a reputation system for vendors, buyers might give ratings—that is, numbers that reflect their satisfaction with a given transaction—for a variety of different dimensions for each vendor. For instance, a given vendor might have good performance in terms of response time or customer service, but the vendor's geographic location might be inconvenient. Buyers provide feedback on a number of these rating dimensions at once, to provide a comprehensive view of the entity. The job of the reputation system is to aggregate these ratings into one or more published scores that are meaningful and useful to participants in the system. A good scoring system will possess many of the following qualities:

Accurate for long-term performance
: The system reflects the confidence (the likelihood of accuracy) of a given score. It can also distinguish between a new entity of unknown quality and an entity with bad long-term performance.

Weighted toward current behavior
: The system recognizes and reflects recent trends in entity performance. For instance, an entity that has behaved well for a long time but suddenly goes downhill is quickly recognized and no longer trusted.

Efficient
: It is convenient if the system can recalculate a score quickly. Calculations that can be performed incrementally are important.

Robust against attacks
: The system should resist attempts of any entity or entities to influence scores other than by being more honest or having higher quality.

Amenable to statistical evaluation
: It should be easy to find outliers and other factors that can make the system rate scores differently.

Private
: No one should be able to learn how a given rater rated an entity except the rater himself.

Smooth
: Adding any single rating or small number of ratings doesn't jar the score much.

Microsoft Exhibit 1019

Understandable

It should be easy to explain to people who use these scores what they mean—not only so they know how the system works, but so they can evaluate for themselves what each score implies.

Verifiable

A score under dispute can be supported with data.

Note that a number of these requirements seem to be contradictory. We will explore the benefits and trade-offs from each of them over the course of the rest of this section.

Attacks and adversaries

Two questions determine how we evaluate the security of reputation systems. First, what information needs to be protected? Second, who are the adversaries?

The capabilities of potential adversaries and the extent to which they can damage or influence the system dictate how much energy should be spent on security. For instance, in the case of Free Haven, if political dissidents actually began using the system to publish their reports and information, government intelligence organizations might be sufficiently motivated to spend millions of dollars to track the documents to their sources.

Similarly, if a corporation planning a $50 million transaction bases its decisions on a reputation score that Reputation Technologies, Inc., provides, it could be worth many millions of dollars to influence the system so that a particular company is chosen. Indeed, there are quite a few potential adversaries in the case of Reputation Technologies, Inc. A dishonest vendor might want to forge or use bribes to create good feedback to raise his resulting reputation. In addition to wanting good reputations, vendors might like their competitors' reputations to appear low. Exchanges—online marketplaces that try to bring together vendors to make transactions more convenient—would like their vendors' reputations to appear higher than those of vendors that do business on other exchanges. Vendors with low reputations—or those with an interest in people being kept in the dark—would like reputations to appear unusably random. Dishonest users might like the reputations of the vendors that they use to be inaccurate, so that their competitors will have inaccurate information.

Perhaps the simplest attack that can be made against a scoring system is called *shilling*. This term is often used to refer to submitting fake bids in an auction, but it can be considered in a broader context of submitting fake or misleading ratings. In particular, a person might submit positive ratings for one of her friends

Microsoft Exhibit 1019

(positive shilling) or negative ratings for her competition (negative shilling). Either of these ideas introduces more subtle attacks, such as negatively rating a friend or positively rating a competitor to try to trick others into believing that competitors are trying to cheat.

Shilling is a very straightforward attack, but many systems are vulnerable to it. A very simple example is the AOL Instant Messenger system. You can click to claim that a given user is abusing the system. Since there is no support for detecting multiple comments from the same person, a series of repeated negative votes will exceed the threshold required to kick the user off the system for bad behavior, effectively denying him service. Even in a more sophisticated system that detects multiple comments by the same person, an attacker could mount the same attack by assuming a multitude of identities.

Vulnerabilities from overly simple scoring systems are not limited to "toy" systems like Instant Messenger. Indeed, eBay suffers from a similar problem. In eBay, the reputation score for an individual is a linear combination of good and bad ratings, one for each transaction. Thus, a vendor who has performed dozens of transactions and cheats on only 1 out of every 4 customers will have a steadily rising reputation, whereas a vendor who is completely honest but has only done 10 transactions will be displayed as less reputable. As we have seen, a vendor could make a good profit (and build a strong reputation!) by being honest for several small transactions and then being dishonest for a single large transaction.

Weighting ratings by size of transaction helps the issue but does not solve it. In this case, large transactions would have a large impact on the reputation score of a vendor, and small transactions would have only a small impact. Since small transactions don't have much weight, vendors have no real incentive to be honest for them—making the reputation services useless for small buyers. Breaking reputation into many different dimensions, each representing the behavior of the vendor on a given *category* of transaction (based on cost, volume, region, etc.), can solve a lot of these problems. See the section "Scoring algorithms", later in this chapter for more details and an analysis of this idea.

Aspects of a scoring system

The particular scoring system or algorithm used in a given domain should be based on the amount of information available, the extent to which information must be kept private, and the amount of accuracy required.

Microsoft Exhibit 1019

In some situations, such as verifying voting age, a fine-grained reputation measurement is not necessary—simply indicating who seems to be sufficient or insufficient is good enough.

In a lot of domains, it is very difficult to collect enough information to provide a comprehensive view of each entity's behavior. It might be difficult to collect information about entities because the volume of transactions is very low, as we see today in large online business markets.

But there's a deeper issue than just whether there are transactions, or whether these transactions are trackable. More generally: does there exist some sort of proof (a receipt or other evidence) that the rater and ratee have actually interacted?

Being able to prove the existence of transactions reduces problems on a wide variety of fronts. For instance, it makes it more difficult to forge large numbers of entities or transactions. Such verification would reduce the potential we described in the previous section for attacks on AOL Instant Messenger. Similarly, eBay users currently are able to directly purchase a high reputation by giving eBay a cut of a dozen false transactions which they claim to have performed with their friends. With transaction verification, they would be required to go through the extra step of actually shipping goods back and forth.

Proof of transaction provides the basis for Amazon.com's simple referral system, "Customers who bought this book also bought..." It is hard to imagine that someone would spend money on a book just to affect this system. It happens, however. For instance, a publishing company was able to identify the several dozen bookstores across America that are used as sample points for the New York Times bestseller list; they purchased thousands of copies of their author's book at these bookstores, skyrocketing the score of that book in the charts.*

In some domains, it is to most raters' perceived advantage that everyone agree with the rater. This is how chain letters, Amway, and Ponzi schemes get their shills: they establish a system in which customers are motivated to recruit other customers. Similarly, if a vendor offered to discount past purchases if enough future customers buy the same product, it would be hard to get honest ratings for that vendor. This applies to all kinds of investments; once you own an investment, it is not in your interest to rate it negatively so long as it holds any value at all.

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* David D. Kirkpatrick, "Book Agent's Buying Fuels Concern on Influencing Best-Seller Lists," *New York Times Abstracts*, 08/23/2000, Section C, p. 1, col. 2, c. 2000, New York Times Company.

Microsoft Exhibit 1019

Collecting ratings

One of the first problems in developing reputation systems is how to collect ratings. The answer depends highly on the domain, of course, but there are a number of aspects that are common across many domains.

The first option is simply to observe as much activity as possible and draw conclusions based on this activity. This can be a very effective technique for automated reputation systems that have a lot of data available. If you can observe the transaction flow and notice that a particular vendor has very few repeat customers, he probably has a low reputation. On the other hand, lack of repeat customers may simply indicate a market in which any given buyer transacts infrequently. Similarly, finding a vendor with many repeat customers might indicate superior quality, or it might just indicate a market in which one or a few vendors hold a monopoly over a product. Knowledge of the domain in question is crucial to knowing how to correctly interpret data.

In many circumstances it may be difficult or impossible to observe the transaction flow, or it may be unreasonable to expect parties to take the initiative in providing feedback. In these cases, a reasonable option is to solicit feedback from parties involved in each transaction. This can be done either by publicizing interest in such feedback and providing incentives to respond, or even by actively going to each party after an observed transaction and requesting comments. Reputation Technologies, Inc., for instance, aggressively tries to obtain feedback after each transaction.

Tying feedback to transactions is a very powerful way of reducing vulnerabilities in the system. It's much more difficult for people to spam positive feedback, since each item of feedback has to be associated with a particular transaction, and presumably only the latest piece of feedback on a given transaction would actually count.

On the surface, it looks like this requires an exchange or other third-party transaction moderator, to make it difficult to simply fabricate a series of several thousand transactions and exploit the same vulnerability. However, vendors could provide blinded receipts for transactions—that is, the vendors would not be able to identify which buyer was providing the ratings. Without such a receipt, the reputation system would ignore feedback from a given buyer. Thus, buyers could not provide feedback about a vendor without that vendor's permission. There are a number of new problems introduced by this idea, such as how to respond if vendors fail to provide a receipt, but it seems to address many of the difficult issues about shilling in a decentralized environment.

Microsoft Exhibit 1019

A final issue to consider when collecting ratings is how fair the ratings will be—that is, how evenly distributed are the raters out of the set of people who have been doing transactions? If the only people who have incentive to provide ratings are those that are particularly unhappy with their transaction, or if only people with sufficient technical background can navigate the rating submission process, the resulting scores may be skewed to the point of being unusable. One solution to this could involve incentives that lead more people to respond; another approach is to simply collect so much data that the issue is no longer relevant. (The Slashdot moderation system, for instance, depends on the participation of huge numbers of independent moderators.) But systematic errors or biases in the ratings will generally defeat this second approach.

Bootstrapping

One of the tough questions for a reputation-based trust system is how to get started. If users make choices based on the scores that are available to them, but the system has not yet collected enough data to provide useful scores, what incentive do buyers have to use the system? More importantly, what incentive do they have to contribute ratings to the system?

Free Haven can avoid this problem through social means. Some participants will be generous and willing to try out new nodes just to test their stability and robustness. In effect, they will be performing a public service by risking some of their reputation and resources evaluating unknown nodes. However, businesses, particularly businesses just getting started in their fields and trying to make a name for themselves, won't necessarily be as eager to spend any of their already limited transaction volume on trying out unknown suppliers.

The way to present initial scores for entities depends on the domain. In some noncommercial domains, it might be perfectly fine to present a series of entities and declare no knowledge or preference; in others, it might be more reasonable to list only those entities for which a relatively certain score is known. Reputation Technologies needs to provide some initial value to the users; this can be done by asking vendors to provide references (that is, by obtaining out-of-band information) and then asking those references to fill out a survey describing overall performance of and happiness with that vendor. While this bootstrapping information may not be as useful as actual transaction-based feedback (and is more suspect because the vendors are choosing the references), it is a good starting point for a new system.

Bootstrapping is a much more pronounced issue in a centralized system than in a decentralized system. This is because in a decentralized system, each user develops his own picture of the universe: he builds his trust of each entity based

Microsoft Exhibit 1019

on his own evidence of past performance and on referrals from other trusted parties. Thus, every new user effectively joins the system "at the beginning," and the process of building a profile for new users is an ongoing process throughout the entire lifetime of the system. In a centralized environment, on the other hand, ratings are accumulated across many different transactions and over long periods of time. New users trust the centralized repository to provide information about times and transactions that happened before the user joined the system.

In a newly developed system, or for a new entity in the system, the choice of the default reputation score is critical. If it's easy to create a new identity (that is, pseudonym), and new users start out with an average reputation, users who develop a bad reputation are encouraged to simply drop their old identities and start over with new ones. One way to deal with this problem is to start all new users with the lowest possible reputation score; even users with a bad track record will then have an incentive to keep their current identities.

Another approach to solving this problem is to make it difficult to create a new identity. For instance, this can be done by requiring some proof of identity or a monetary fee for registration. Tying the user to her real-world identity is the simplest, and probably the most effective, way to reduce abuse—but only if it's appropriate for that system.

Personalizing reputation searches

The user interface—that is, the way of presenting scores and asking questions—is a crucial element of a reputation system. Scores cannot simply be static universal values representing the overall quality of an individual. Since a score is an attempt to predict future performance, each score must be a prediction for a particular context. That is, the user interface must allow participants to query the system for the likelihood of a successful transaction for their particular situation. The more flexibility a client has, the more powerful and useful the system is (so long as users can still understand how to use it).

The user interface must also display a *confidence value* for each score—that is, how likely the score is to reflect the reality of the subject's behavior. The mechanism for generating this confidence value depends on the domain and the scoring algorithm. For instance, it might reflect the number of ratings used to generate the score, the standard deviation of the set of ratings, or the level of agreement between several different scoring algorithms that were all run against the ratings set. Confidence ratings are a major topic in Chapter 17.

Not only does a confidence value allow users to have a better feel for how firm a given score is, but it can also allow a more customized search. That is, a user

Microsoft Exhibit 1019

might request that only scores with a certain minimum confidence value be displayed, which would weed out new users as well as users with unusual (widely varying) transaction patterns.

In some domains, qualitative statements (like verbal reviews) can enhance the value of a quantitative score. Simply providing a number may not feel as genuine or useful to users—indeed, allowing for qualitative statements can provide more flexibility in the system, because users providing feedback might discuss topics and dimensions which are difficult for survey authors to anticipate. On the other hand, it is very difficult to integrate these statements into numerical scores, particularly if they cover unanticipated dimensions. Also, as the number of statements increases, it becomes less useful to display all of them. Choosing which statements to display not only requires manual intervention and choice, but might also lead to legal liabilities. Another problem with providing verbal statements as part of the score is the issue of using this scoring system in different countries. Statements may need to be translated, but numbers are universal.

Scoring algorithms

As we've seen in the previous sections, there are many different aspects to scoring systems. While we believe that query flexibility is perhaps the most crucial aspect to the system, another important aspect is the actual algorithm used to aggregate ratings into scores. Such an algorithm needs to answer most of the requirements that we laid out in the section "Scoring systems." Broadly speaking, the scoring algorithm should provide accurate scores, while keeping dishonest users from affecting the system and also preventing privacy leaks (as detailed in the next section).

Keeping dishonest users from affecting the system can be done in several ways. One simple way is to run statistical tests independent of the actual aggregation algorithm, to attempt to detect outliers or other suspicious behavior such as a clique of conspiring users. Once this suspicious behavior has been identified, system operators can go back, manually examine the system, and try to prune the bad ratings. While this appears to be a very time-intensive approach that could not possibly be used in a deployed system, eBay has used exactly this method to try to clean up their system once dishonest users have been noticed.*

A more technically sound approach is to weight the ratings by the *credibility* of each rater. That is, certain people contribute more to the score of a given entity based on their past predictive ability. Google makes use of this idea in its Internet

* "eBay Feedback Removal Policy," *http://pages.ebay.com/help/community/fbremove.html*.

Microsoft Exhibit 1019

search engine algorithm. Its algorithm counts the number of references to a given page; the more pages that reference that page, the more popular it is. In addition, the pages that are referenced from popular pages are also given a lot of weight. This simple credibility metric produces much more accurate responses for web searches.

By introducing the notion of local credibility rather than simple global credibility for each entity, the system can provide a great deal of flexibility and thus stronger predictive value. Local credibility means that a rating is weighted more strongly if the situation in which that rating was given is similar to the current situation. For instance, a small farmer in Idaho looking into the reputation of chicken vendors cares more about the opinion of a small farmer in Arkansas than he does about the opinion of the Great Chinese Farming Association. Thus, the algorithm would generate a score that more accurately reflects the quality of the vendor according to other similar buyers. Similarly, if Google knew more about the person doing the web search, it could provide an even more accurate answer. Before being bought by Microsoft, firefly.net offered a service based on this idea.

One of the problems with incorporating credibility into the scoring algorithm is that, in some domains, an individual's ability to perform the protocol honestly is very separate from an individual's ability to predict performance of others.

In the Free Haven system, for instance, a server may be willing to store documents and supply them to readers, but keep no logs about transactions or trades (so it has no idea which other servers are behaving honestly). In the case of Reputation Technologies, one vendor might be excellent at providing high-quality products on time, leading to a high reputation score, but possess only average skill at assessing other vendors. Indeed, a consulting firm might specialize in predicting performance of vendors but not actually sell any products of its own.

One way to solve this problem is to have separate scores for performance and credibility. This makes it more complex to keep track of entities and their reputations, but it could provide tremendous increases in accuracy and flexibility for scoring systems.

Weighting by credibility is not the only way to improve the accuracy and robustness of the scoring algorithm. Another approach is to assert that previous transactions should carry more weight in relation to how similar they are to the current transaction. Thus, a vendor's ability to sell high-quality Granny Smith apples should have some bearing on his ability to sell high-quality Red Delicious apples. Of course, this could backfire if the vendor in question specializes only in Granny Smith apples and doesn't even sell Red Delicious apples. But in

Microsoft Exhibit 1019

general, weighting by the so-called *category* of the transaction (and thus the vendor's reputation in related categories) is a very powerful idea. Separating reputations into categories can act as a defense against some of the subtle shilling attacks described above, such as when a vendor develops a good reputation at selling yo-yos and has a side business fraudulently selling used cars.

The category idea raises very difficult questions. How do we pick categories? How do we know which categories are related to which other categories, and how related they are? Can this be automated somehow, or do the correlation coefficients have to be estimated manually?

In the case of Free Haven, where there is only one real commodity—a document—and servers either behave or they don't, it might be feasible to develop a set of categories manually and allow each server to manually configure the numbers that specify how closely related the categories are. For instance, one category might be files of less than 100K that expire within a month. A strongly related category would be files between 100K and 200K that expire within a month; perhaps we would say that this category is 0.9-related to the first. A mostly unrelated category would be files more than 500MB in size that expire in 24 months. We might declare that this category is 0.05-related to the first two.

With some experience, an algorithm might be developed to tweak the correlation coefficients on the fly, based on how effective the current values have been at predicting the results of future transactions. Similarly, we might be able to reduce the discrete categories into a single continuous function that converts "distance" between file size and expiration date into a correlation coefficient.

Reputation Technologies is not so lucky. Within a given exchange, buyers and sellers might barter thousands of different types of goods, each with different qualities and prices; the correlation between any pair of categories might be entirely unclear. To make matters worse, each vendor might only have a few transactions on record, leaving data too sparse for any meaningful comparison.

While we've presented some techniques to provide more accuracy and flexibility in using ratings, we still haven't discussed actual algorithms that can be used to determine scores. The simplest such algorithm involves treating reputations as probabilities. Effectively, a reputation is an estimate of how likely a future transaction in that category is to be honest. In this case, scores are simply computed as the weighted sum of the ratings.

More complex systems can be built out of neural networks or data-clustering techniques, to try to come up with ways of applying nonlinear fitting and optimizing systems to the field of reputation. But as the complexity of the scoring algorithm increases, it becomes more and more difficult for actual users of these

Microsoft Exhibit 1019

systems to understand the implications of a given score or understand what flaws might be present in the system.

Finally, we should mention the *adversarial approach* to scoring systems. That is, in many statistical or academic approaches, the goal is simply to combine the ratings into as accurate a score as possible. In the statistical analysis, no regard is given for whether participants in the system can conspire to provide ratings that break the particular algorithm used.

A concrete example might help to illustrate the gravity of this point. One of the often referenced pitfalls of applying neural networks to certain situations comes from the U.S. military. They wanted to teach their computers how to identify tanks in the battlefield. Thus they took a series of pictures that included tanks, and a series of pictures that did not include tanks. But it turns out that one of the sets was taken during the night, and the other set was taken during the day. This caused their high-tech neural network to learn not how to identify a tank but how to distinguish day from night. Artificial intelligence developers need to remember that there are a number of factors that might be different in a set of samples, and their neural network might not learn quite what they want it to learn.

But consider the situation from our perspective: what if the Russian military were in charge of providing the tank pictures? Is there a system that can be set up to resist bad data samples? Many would consider that learning how to identify a tank under those circumstances is impossible. How about if the Russians could provide only half of the pictures? Only a tenth? Clearly this is a much more complicated problem. When developing scoring systems, we need to keep in mind that simply applying evaluation techniques that are intended to be used in a "clean" environment may introduce serious vulnerabilities.

Privacy and information leaks

Yet another issue to consider when designing a good scoring system is whether the system will be vulnerable to attacks that attempt to learn about the tendencies or patterns of entities in the system. In a business-oriented domain, knowledge about transaction frequency, transaction volume, or even the existence of a particular transaction might be worth a lot of time and money to competitors. The use of a simple and accurate scoring algorithm implies that it should be easy to understand the implication of a vendor's score changing from 8.0 to 9.0 over the course of a day. Perhaps one or more ratings arrived regarding large transactions, and those ratings were very positive.

Microsoft Exhibit 1019

The objectives of providing timeliness and accuracy in the scoring algorithm and of maintaining privacy of transaction data seem to be at odds. Fortunately, there are a number of ways to help alleviate the leakage problems without affecting accuracy too significantly. We will describe some of the more straightforward of these techniques in this section.

The problem of hiding transaction data for individual transactions is very similar to the problem of hiding source and destination data for messages going through mix networks.* More specifically, figuring out what kind of rating influenced a published score by a certain amount is very similar to tracking a message across a middleman node in a mix network. In both cases, privacy becomes significantly easier as transaction volume increases. Also in both cases, adversaries observe external aspects of the system (in the case of the scoring system, the change in the score; in the case of the mix network, the messages on the links to and from the mix node) to try to determine the details of some particular message or group of messages (or the existence of any message at all).

One common attack against the privacy of a scoring system is a *timing attack*. For instance, the adversary might observe transactions and changes in the scores and then try to determine the rating values that certain individuals submitted. Alternatively, the adversary might observe changes in the scores and attempt to discover information about the timing or size of transactions. These attacks are like watching the timings of messages going through various nodes on a mix network, and trying to determine which incoming message corresponds to which outgoing message.

A number of solutions exist to protect privacy. First of all, introducing extra latency between the time that ratings are submitted and the time when the new score is published can make timing correlation more difficult. (On the other hand, this might reduce the quality of the system, because scores are not updated immediately.) Another good solution is to queue ratings and process them in bulk. This prevents the adversary from being able to determine which of the ratings in that bulk update had which effect on the score.

A variant of this approach is the *pooling* approach, in which some number of ratings are kept in a pool. When a new rating arrives, it is added to the pool and a rating from the pool is chosen at random and aggregated into the score. Obviously, in both cases, a higher transaction volume makes it easier to provide timely score updates.

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* D. Chaum (1981), "Untraceable Electronic Mail, Return Addresses, and Digital Pseudonyms." *Communications of the ACM*, vol. 24, no. 2, pp.84–88.

Microsoft Exhibit 1019

An active adversary can respond to bulk or pooled updates with what is known as an *identification flooding attack*. He submits ratings with known effect, and watches for changes in the score that are not due to those ratings. This approach works because he can "flush" the few anonymous ratings that remain by submitting enough known ratings to fill the queue. This attack requires the adversary to produce a significant fraction of the ratings during a given time period.

But all this concern over privacy may not be relevant at all. In some domains, such as Free Haven, the entire goal of the reputation system is to provide as much information about each pseudonymous server as possible. For instance, being able to figure out how Alice performed with Bob's transaction is always considered to be a good thing. In addition, even if privacy is a concern, the requirement of providing accurate, timely scores may be so important that no steps should be taken to increase user privacy.

Decentralizing the scoring system

Many of the issues we've presented apply to both centralized and decentralized reputation systems. In a decentralized system such as Free Haven, each server runs the entire reputation-gathering system independently. This requires each node to make do with only the information that it has gathered firsthand, and it generally requires a broadcast mechanism in order for all nodes to keep their information databases synchronized.

Another approach is to decentralize the scoring system itself, spreading it among the entire set of machines participating in the system. In this section, we present two ways of decentralizing a scoring system. The first exploits redundancy along with user flexibility to reduce the risk from cheating or compromised servers. The second is a more traditional approach to decentralizing a system, but it also brings along the more traditional problems associated with decentralization, such as high bandwidth requirements and difficult crypto problems.

Multiple trusted parties

Assume there is a set of scorers around the world, each independently run and operated. When a transaction happens, the vendor chooses a subset of the scorers and constructs a set of *tickets*. Each ticket is a receipt allowing the buyer to rate the vendor at a particular scorer. The receipts are blinded so that the vendor is not able to link a ticket with any given buyer.

At this point, the buyer can decide to which scorer or scorers he wishes to submit his ratings. Since each scorer could potentially use its own algorithm and have its own prices or publishing habits, each scorer might have its own set of

Microsoft Exhibit 1019

trade-offs based on accuracy, privacy, and security. This technique allows the vendor to veto some of the scorers first. Then the rater chooses from among the remaining scorers. Thus, the ratings will only be submitted to mutually agreeable scorers.

We could extend this scenario to allow both parties in the transaction to provide tickets to each other, creating a more symmetric rating process. This approach introduces complications, because both parties in the transaction need to coordinate and agree on which tickets will be provided before the transaction is completed. There also needs to be some mechanism to enforce or publicize if one side of the transaction fails to provide the promised receipts.

The beauty of decentralizing the scoring system in this manner is that every individual in the system can choose which parts of the system they want to interact with. Participants in transactions can list scorers whom they trust to provide accurate scores, raters can choose scorers whom they trust not to leak rating information, and users looking for scores on various entities can choose scorers whom they trust to provide accurate scores.

Of course, this decentralization process introduces the issue of meta-reputation: how do we determine the reputations of the reputation servers? This sort of reputation issue is not new. Some Mixmaster nodes are more reliable than others,* and users and operators keep uptime and performance lists of various nodes as a public service. We expect that reputation scoring services would similarly gain (external) reputations based on their reliability or speed.

True decentralization

In this scenario, both sides of the transaction obtain blinded receipts as above. Apart from these *raters*, the system also consists of a set of *collectors* and a set of *scorers*. They are illustrated in Figure 16-2.

After the transaction, each rater splits up his rating using Shamir's secret sharing algorithm (described in Chapter 11, *Publius*) or some other *k*-of-*n* system. At this point, the rater submits one share of her rating to each collector. This means that the collectors together could combine the shares to determine her rating, but separately they can learn no information. It is the job of the scorers to provide useful information to clients: when a client does a reputation query for a specific category (situation), the scorer does the equivalent of an *encrypted database query* on the set of collectors.†

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* "Electronic Frontiers Georgia Remailer Uptime List," *http://anon.efga.org*.

† Tal Malkin (1999), MIT Ph.D. thesis, "Private Information Retrieval and Oblivious Transfer."

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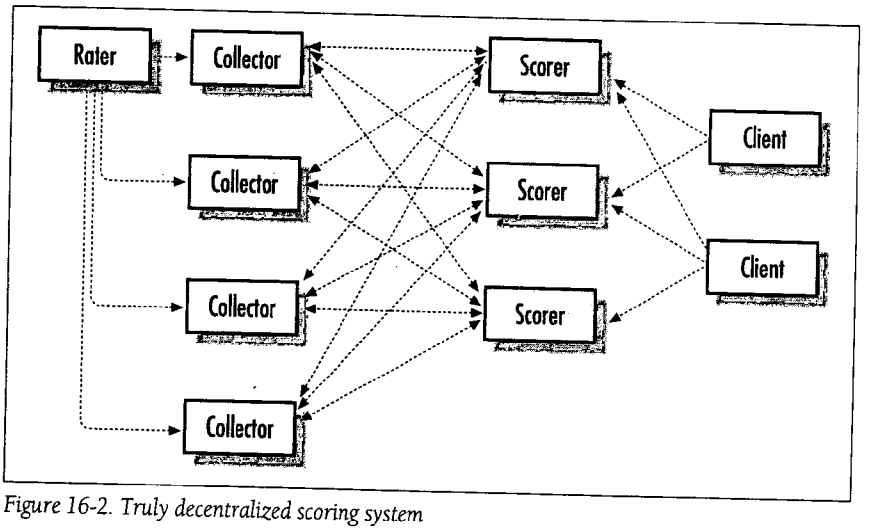

Figure 16-2. Truly decentralized scoring system

A number of technical challenges need to be solved in order to make this scheme work. First of all, the collectors need to have some mechanism for authenticating a rating without reading it. Similarly, they need to have some way to authorize a rater to put his share onto the system without their knowing the author of a given rating. Without this protection, malicious raters could simply flood the system with data until it overflowed.

Once these problems are solved, we need to come up with some sort of computationally feasible and bandwidth-feasible way of communication between the scorers and the collectors. We also need a set of rules that allow the scorers to get the information they need to answer a given query without allowing them to get too much information and learn more than they ought to learn about raters.

With this decentralization comes some subtle questions. Can scorers "accidentally forget" to include a specific rating when they're computing a score? Said another way, is there some way of allowing scorers to provide proof that they included a certain rating in the calculation of the score, without publishing the actual ratings that were used? This question is similar to the question of allowing mix nodes to prove that they forwarded a given message without yielding information that might help an adversary determine the source or destination of the message.*

* Masayuki Abe (1998), "Universally Verifiable MIX-Network with Verification Work Independent of the Number of MIX Servers," *EUROCRYPT '98*, Springer-Verlag LNCS.

Microsoft Exhibit 1019

A case study: Accountability in Free Haven

As described in Chapter 12, the Free Haven project is working toward creating a decentralized and dynamically changing storage service that simultaneously protects the anonymity of publishers, readers, and servers, while ensuring the availability of each document for a lifetime specified by the publisher. Our goals of strong anonymity and long-term persistent storage are at odds with each other. Providing as much anonymity as possible while still retaining sufficient accountability is a very difficult problem. Here we will describe the accountability requirements in greater detail than in Chapter 12 and discuss some approaches to solving them.

Our job is two-fold: We want to keep people from overfilling the bandwidth available from and between servers, and we want to keep people from overfilling the system with data. We will examine each of these goals separately.

Micropayments

In general, there are a number of overall problems with using micropayments in peer-to-peer systems. This general analysis will help motivate our discussion of using micropayments in the Free Haven context. We'll talk about them, then try to apply them to Free Haven.

The difficulty of distributed systems: How to exchange micropayments among peers

Consider the simple approach to micropayments introduced early in this chapter, in the section "Micropayment schemes." Alice wants resources operated by Bob. Alice pays Bob with some micropayments. Bob provides Alice with the access she purchased to his resources.

This sounds like a great model for economically-based distribution that provides both accountability and effective congestion-management of resources. However, the problem is rarely so simple in the case of peer-to-peer distributed systems on the Internet. The reason is that many intermediaries may be involved in a transaction—and one doesn't know who they are before the transaction starts, or perhaps even after the transaction is finished.

Consider an anonymous remailer like Mixmaster. Alice sends an email to Bob through a number of intermediate proxy remailers, which strip all identifying information from the message before transmitting it. This design is used to distribute trust across operational and often jurisdictional lines. Only a very powerful adversary—able to observe large sections of the network and use advanced

Microsoft Exhibit 1019

traffic analysis techniques—should be able to link the sender and recipient of any given message. Hence, we achieve an essentially anonymous communications path for email.

Consider again the Gnutella routing protocol. Alice seeks some piece of information contained in the network. She sends out a query to all peers that she knows about (her "friends"); these peers in turn propagate the request along, branching it through the network. Hopefully, before the time-to-live (TTL) of the query expires, the request traverses enough intermediate hops to find Bob, who responds with the desired information. The Freenet routing protocol works similarly, covering some fraction of the surrounding network over the course of the search.

These examples highlight a design quite common in peer-to-peer systems, especially for systems focusing on anonymity (by distributing trust) or searching (by distributing content). That is, endpoint peers are not the only ones involved in an operation; Alice and Bob are joined by any number of intermediate peers. So how should we handle micropayments? What are the entities involved in a transaction? Four possible strategies are illustrated in Figure 16-3:

End-to-end model

The simplest approach is to make Alice send Bob some form of payment and not worry about what happens to any intermediaries. This model works fine for operations that do not make use of intermediate nodes. But if intermediate peers are involved, they lack any protection from attack. Bob might even be fictitious. Alice can attack any number of intermediate peers by routing her queries through them, using up their bandwidth or wiping out the data in Freenet-style data caches. This problem is precisely our motivation for using micropayments!

Pairwise model

Recognizing the problems of the end-to-end model, we can take a step upward in complexity and blindly throw micropayments into every transaction between every pair of peers. One long route can be modeled as a number of pairwise transactions. This model might appear to protect each recipient of payments, but it is also fundamentally flawed.

Using fungible micropayments, each peer earns one unit from its predecessor and then spends one unit on its successor. Assuming equal costs throughout the network, Alice is the only net debtor and Bob the only net creditor. But if a single malicious operator is in charge of both Alice and Bob, these two peers have managed to extract work from the intermediate nodes without paying—a more subtle DoS or flooding attack!

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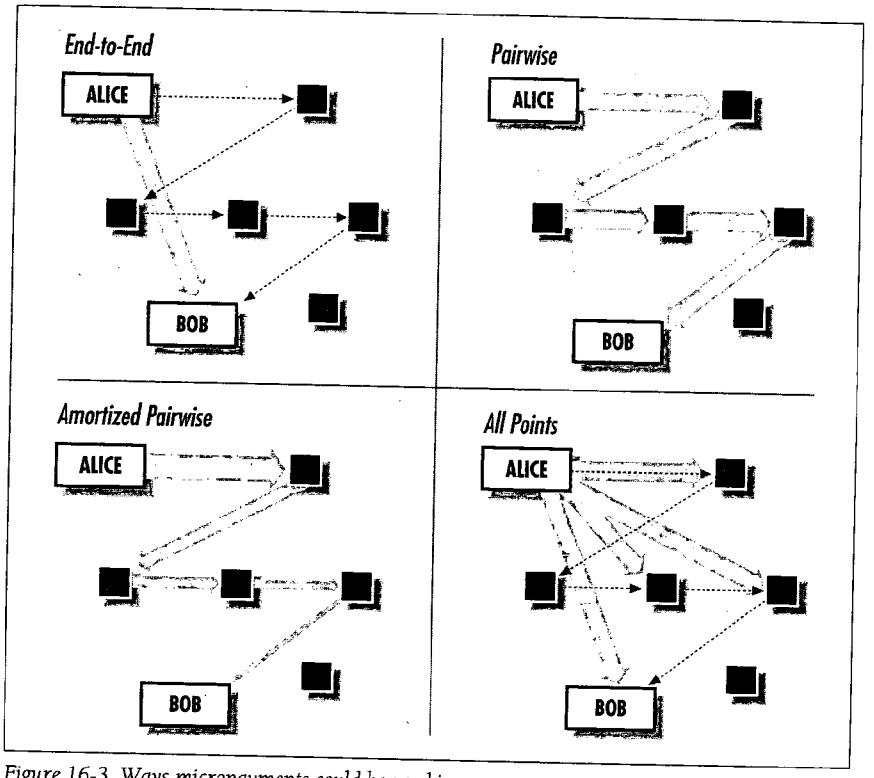

Figure 16-3. Ways micropayments could be used in a peer-to-peer communication path

Using nonfungible micropayments, Alice remains a net debtor, but so are all intermediate peers. Alice can make use of greater computational resources (centralized or distributed) to flood intermediate peers with POWs. Being properly-behaving nodes, these peers attempt to make good on the micropayment exchange, and start churning out POWs for the next hop in the protocol... and churning... and churning. Eventually Alice can exhaust the resources of a whole set of smaller, honest peers.

Amortized pairwise model

Taking what we learned about the risks of the pairwise model, we can design a more sophisticated one that amortizes Alice's cost throughout the network route by iteratively decreasing the cost of transactions as they move through the system. Say Alice pays X with four units of micropayment, X pays Y with three units, Y pays Z with two units, and Z finally pays Bob only one unit.

In the case of nonfungible POWs, we still lose. First of all, Alice can still make use of greater wealth, economies of scale, distributed computing, etc.,

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in order to attack intermediate nodes. While the load decreases as it moves though the system, peers X, Y, and Z still need to devote some of their own resources; they may be unable to afford that load.

For fungible payments, this model appears more promising. Intermediate nodes end up as net creditors: their resources are paid for by the cut they take from Alice's initial lump-sum payment.

But this model has another weakness from a security point of view: we leak information regarding the route length. We mentioned the Mixmaster mix net at the beginning of this section; the system allows a sender to specify the number and identity of intermediate remailers. This number of hops and their corresponding identities are unknown to all other parties.* But if we use amortized payments, each peer in the chain has to know the amount it is given and the function used to decrease payments, so intermediate peers can extrapolate how many hops are in the route as well as their relative positions in the chain.

Furthermore, Alice may not know the route length. If a system uses Gnutella- or Freenet-type searching, Alice has no idea how many hops are necessary before the query reaches Bob.

As Alice's query branches out through the network, payments could become prohibitively expensive. For example, in Gnutella, we can estimate the number of nodes that a query traverses by treating the network as a binary tree rooted at the originating node, where the query traverses the first k levels (k is the query's time-to-live (TTL)). This gives a total of $2^{k+1}-1$ nodes visited by the query—and all of these nodes want to be paid. Thus the amount of nodes to pay is exponential in the TTL. Indeed, in reality the branching factor for the tree will be much greater than 2, leading to even more nodes that need payment. Freenet searching may be much more efficient; for more details, see Chapter 14.

All points model

These previous problems lead us to settle on an all points model. Alice pays each peer engaged in the protocol, intermediate and endpoint alike. Of course, we immediately run into the same problem we had in the previous model, where Alice may not know which peers are involved, especially during a search. But let's assume for this discussion that she knows which nodes she'll be using.

* We ignore the possibility of traffic analysis here and assume that the user chooses more than one hop.

Microsoft Exhibit 1019

This solution is ideal for such fully identified systems. The cost of resource use falls solely upon its instigating requestor.

Anonymous systems add a few difficulties to using this model. First of all, we lose some of our capacity to use interactive payment models. For the forward-only Mixmaster mix net, intermediate nodes cannot tell Alice what client puzzle she should solve for them because only the first hop knows Alice's identity. Therefore, payments must be of a noninteractive variety.

To stop double-spending, the scheme must use either a centralized bank server model (such as Chaumian e-cash) or have recipient-specific information encoded in the payment (such as hash cash). This recipient-specific information should further be hidden from view, so as to protect an eavesdropper from being able to piece together the route by looking at the micropayments. Recipient-hiding cryptosystems* help ensure that the act of encrypting the micropayment does not itself leak information about to whom the data is encrypted.

In short, the all points payment model—while offering advantages over the prior three models—presents its own difficulties.

Micropayment schemes can help ensure accountability and resource allocation in peer-to-peer systems. But the solution requires careful design and a consideration of all security problems: there are no simple, off-the-shelf solutions.

Micropayments in the Free Haven context

Most Free Haven communication is done by broadcast. Every document request or reputation referral is sent to every other Free Haven server. Even if we can solve the micropayments issue for mix nets as described above, we still need to ease the burden of multiple payments incurred by each server each time it sends even a single Free Haven broadcast.

The first step is to remember that our communications channel already has significant latency. Nobody will care if we introduce a little bit more. We can queue the broadcasts and send a batch of them out every so often—perhaps once an hour. This approach makes the problem of direct flooding less of a problem, because no matter how many broadcasts we do in the system, our overall use of the mix net by the n Free Haven servers is limited to n^2 messages per hour. We assume that the size of the message does not dramatically increase as we add more broadcasts to the batch; given that each Free Haven communication is very small, and given the padding already present in the mix net protocol, this seems like a reasonable assumption.

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* David Hopwood, "Recipient-Hiding Blinded Public-Key Encryption," draft manuscript.

Microsoft Exhibit 1019

However, batching helps the situation only a little. For several reasons—the lack of a widely deployed electronic cash system, our desire to provide more frictionless access regardless of wealth, and the complex, central-server model used by most fungible payment systems to issue coins—nonfungible POW micropayments are better suited for Free Haven. Likewise, nonfungible payments work best with the expensive all-points payment scheme. We still have the problem, therefore, that every server must pay each intermediate node used to contact every other server each hour.

It is conceivable that spreading the waste of time for each message over the hour would produce a light enough load. Servers could simply do the computation with idle cycles and send out a batch of broadcasts whenever enough calculations have been performed.

We can solve this more directly by thinking of the server Alice as a mailing list that uses pay-per-send email as described earlier in this chapter, in the section "Nonfungible micropayments." In this case, users attach special tickets to messages sent to Alice, so they don't have to perform a timewasting computation. Similarly, we might be able to introduce into the mix net protocol a "one free message per hour" exception. But making this exception introduces a difficult new problem—our primary purpose is to maintain the anonymity of the senders and recipients through the mix net, but at the same time we want to limit each server to sending only one message per recipient in each hour. Thus, it seems that we need to track the endpoints of each message in order to keep count of who sent what.

Having Alice distribute blinded tickets as an end-to-end solution doesn't work easily either, as these tickets are used with the intermediate mix net nodes. The tickets would need to assure the nodes of both Alice's identity as a Free Haven server and her certification of the user's right to mail her, while still maintaining the pseudonymity of both parties.

The alternative is to have node-specific tickets for our all points model. More precisely, each mix net node issues a limited number of blinded tickets for each hour and user. This design also adds the functionality of a prepaid payment system, if we want one. Project Anon, an anonymous communications project, suggests such a technique.* It's important to note that most blind signature techniques use interactive protocols, which are less suitable for our type of application.

* Oliver Berthold, Hannes Federrath, and Marit Köhntopp (2000), "Anonymity and Unobservability in the Internet," *Workshop on Freedom and Privacy by Design/Conference on Computers, Freedom and Privacy 2000*, Toronto, Canada, April 4–7.

Microsoft Exhibit 1019

Introducing a free message every hour to the mix net protocol also allows for smooth integration of another Free Haven feature: we want to allow anonymous users to proxy a document retrieve request through certain (public) Free Haven servers. Specifically, a user generates a one-time mix net reply block and a one-time key pair and passes these to a Free Haven node along with a handle to the document being requested. This Free Haven node broadcasts the query to all other servers, just as in a normal retrieve operation. Because bundling extra broadcasts into each hourly message is virtually free, we can allow these extra anonymous requests without much extra cost. Of course, a concerted flood of document requests onto a server could cause its hourly message to be very large; public Free Haven servers may have to drop document requests after a certain threshold or find some other mechanism for limiting this threat of flooding.

Overall, providing bandwidth accountability along with anonymity is a tough problem. What we describe above does not provide any clear solution for an environment in which we want to maintain strong anonymity. This discussion may help to explain why current mix net implementations don't use micropayments to address accountability. Further research is certainly necessary.

Reputation systems

The Free Haven reputation solution has two parts: first, we need to notice servers that drop data early, and second, we need to develop a process for "punishing" these servers.

It's very difficult to notice if a server drops data early, and we still haven't solved the problem completely. The buddy system laid out in Chapter 12 is our current approach, and it may well be good enough. After all, we simply have to provide a system that is difficult to *reliably* fool—it doesn't have to catch every single instance of misbehavior.

As for punishing misbehaving servers, that's where our reputation scheme comes in. The first step in developing a solution that uses reputation systems is to examine the situation more thoroughly and try to understand our goals and limitations. Every situation contains features that make it hard to develop a reputation solution and features that make it easier.

We expect the Free Haven domain to include a number of generous individuals who will take some risks with their reputations and resources. Since disk space is very cheap and getting cheaper, and there's no significant loss if a single trade goes bad, the Free Haven environment is relatively lenient.

Microsoft Exhibit 1019

Ratings in the reputation system are tied to transactions and include digitally signed receipts. So we can be pretty certain that either a given transaction actually did happen, or the two parties are conspiring. At regular intervals, each Free Haven server broadcasts a "reputation referral," a package of ratings of other servers. Nodes should broadcast reputation referrals in several circumstances:

- When they log the honest completion of a trade
- When they check to see if a buddy to a share they hold is still available and find that it is missing
- When there's a substantial change in the reputation or credibility of a given server, compared to the last reputation referral about that server

How often to broadcast a referral can be a choice made by each server. Sending referrals more often allows that server to more easily distribute its current information and opinions to other servers in the network. On the other hand, frequent broadcasts use more bandwidth, and other servers may ignore servers that talk too much.

Servers get most of their information from their own transactions and trades. After all, those are the data points that they are most certain they can trust. Each server keeps its own separate database of information that it knows, based on information it has observed locally and information that has come to it. Thus every server can have a different view of the universe and a different impression of which servers are reputable and which aren't. Indeed, these local decisions introduce a lot of flexibility into the design: Each server operator can choose her own thresholds for trust, broadcast frequency, which trades are accepted or offered, etc. These decisions can be made to suit her particular situation, based, for instance, on available bandwidth and storage or the expected amount of time that she'll be running a Free Haven server.

Since each server is collecting referrals from other servers (and some of those servers may be trying to discredit good servers or disrupt the system in other ways), we need a robust algorithm for combining the referrals. Each server operator can use an entirely separate algorithm, but realistically speaking, most of them will use a default configuration recommended by the Free Haven designers.

Some ways of choosing a good algorithm are described earlier in this chapter, in the section "Scoring systems." In Free Haven, we don't expect to have to focus on very many parameters in order to get a reasonable score. Our basic approach to developing a score for a given server is to iterate through each rating available on that server and weight each rating based on how important and relevant it appears to be. Parameters that we might want to examine while weighting a score include the following:

Microsoft Exhibit 1019

How recent is this rating?

Newer ratings should get more weight.

How similar (in terms of size and expiration date) is this rating to the transaction I'm currently considering?

Similar ratings should get more weight.

In my experience, has this server accurately predicted the behavior that I have observed?

This deals with the credibility of the rater.

How often does the server send referrals?

If a server is verbose, we might choose to assign a lower weight to each rating. On the other hand, if this is the first referral we've ever gotten from this server, we might regard it with skepticism.

How long has the rating server been a Free Haven server?

We will probably have greater confidence in servers that have been part of the system for a long time.

As explained in Chapter 12, each server needs to keep two values to describe each other server it knows about: reputation and credibility. Reputation signifies a belief that the server in question will obey the Free Haven Protocol. Credibility represents a belief that the referrals from that server are valuable information. For each of these two values, each server also needs to maintain a confidence rating. This indicates how firmly the server believes in these values, and indicates how much a value might move when a new rating comes in.

When new servers want to join the system, they must contact certain servers that are acting as *introducers*. These introducers are servers that are willing to advertise their existence in order to introduce new servers to the rest of the servnet. Introducing consists simply of broadcasting a reputation referral with some initial reputation values. Each introducer can of course choose her own initial values, but considering the discussion in the section "Problems with pseudospoofing and possible defenses," earlier in this chapter, it seems most reasonable to broadcast an initial referral value of zero for both reputation and credibility.

At first glance, it seems that we do not need to worry about information leaks from the compiled scores—after all, the entire goal of the system is to communicate as much information as possible about the behavior history of each pseudonym (server). But a closer examination indicates that a large group of ratings might reveal some interesting attributes about a given server. For instance, by looking at the frequency and quantity of transactions, we might be able to learn that a given server has a relatively large hard drive. We currently believe that leaking this type of information is acceptable.

Microsoft Exhibit 1019

Other considerations from the case study

Alice has paid for some resource. But did she get what she paid for? This question deals with the problem of trust, discussed more fully in Chapter 15. But given our discussion so far, we should note a few issues that apply to various distributed systems.

In the case of data storage, at a later date, Alice can query Bob for her document and verify its checksum in order to be sure Bob has properly stored her document. She cannot be sure Bob has answered all requests for that document, but she may be more convinced if Bob can't determine that she's the one doing the query.

A distributed computation system can check the accuracy of the results returned by each end user. As we saw earlier in this chapter, some problems take a lot longer to solve than a checker takes to verify the answer. In other situations, we can use special algorithms to check the validity of aggregate data much faster than performing each calculation individually. For example, there are special batch verification methods for verifying many digital signatures at once that run much faster than checking each signature individually.* On the other hand, sometimes these schemes leave themselves open to attack.†

The methods we've described take advantage of particular properties of the problem at hand. Not all problems are known to have these properties. For example, the SETI@home project would benefit from some quick method of checking correctness of its clients. This is because malicious clients have tried to disrupt the SETI@home project in the past. Unfortunately, no quick, practical methods for checking SETI@home computations are currently known.‡

Verifying bandwidth allocation can be a trickier issue. Bandwidth often goes hand-in-hand with data storage. For instance, Bob might host a web page for Alice, but is he always responding to requests? A starting point for verification is to sample anonymously at random and gain some statistical assurance that Bob's server is up. Still, the Mixmaster problem returns to haunt us. David Chaum, who proposed mix nets in 1981,§ suggested that mix nodes publish the outgo-

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* Mihir Bellare, Juan A. Garay, and Tal Rabin (1998), "Fast Batch Verification for Modular Exponentiation and Digital Signatures," *EUROCRYPT '98*, pp. 236–250.

† Colin Boyd and Chris Pavlovski (2000), "Attacking and Repairing Batch Verification Schemes," *ASIACRYPT 2000*.

‡ David Molnar (September 2000), "The SETI@home Problem," *ACM Crossroads*, *http://www.acm.org/crossroads/columns/onpatrol/september2000.html*.

§ D. Chaum, "Untraceable Electronic Mail, Return Addresses, and Digital Pseudonyms," *op. cit.*

Microsoft Exhibit 1019

ing batch of messages. Alternatively, they could publish some number per message, selected at random by Alice and known only to her. This suggestion works well for a theoretical mix net endowed with a public bulletin board, but in Internet systems, it is difficult to ensure that the mix node actually sends out these messages. Even a bulletin board could be tampered with.

Above, we have described some approaches to addressing accountability in Free Haven. We can protect against bandwidth flooding through the use of micropayments in the mix net that Free Haven uses for communication, and against data flooding through the use of a reputation system. While the exact details of these proposed solutions are not described here, hopefully the techniques described to choose each accountability solution will be useful in the development of similar peer-to-peer publication or storage systems.

Conclusion

Now we've seen a range of responses to the accountability problem. How can we tell which ones are best? We can certainly start making some judgments, but how does one know when one technique is better suited than another?

Peer-to-peer remains a fuzzy concept. A strict definition has yet to be accepted, and the term covers a wide array of systems that are only loosely related (such as the ones in this book). This makes hard and fast answers to these questions very difficult. When one describes operating systems or databases, there are accepted design criteria that all enterprise systems should fulfill, such as security and fault tolerance. In contrast, the criteria for peer-to-peer systems can differ widely for various distributed application architectures: file sharing, computation, instant messaging, intelligent searching, and so on.

Still, we *can* describe some general themes. This chapter has covered the theme of accountability. Our classification has largely focused on two key issues:

- Restricting access and protecting from attack
- Selecting favored users

Dealing with resource allocation and accountability problems is a fundamental part of designing any system that must serve many users. Systems that do not deal with these problems have found and will continue to find themselves in trouble, especially as adversaries find ways to make such problems glaringly apparent.

Microsoft Exhibit 1019

With all the peer-to-peer hype over the past year—which will probably be spurred on by the publication of this book—we want to note a simple fact: peer-to-peer won't save you from dealing with resource allocation problems.

Two examples of resource allocation problems are the Slashdot effect and distributed denial of service attacks. From these examples, it's tempting to think that somehow being peer-to-peer will save a system from thinking about such problems—after all, there's no longer any central point to attack or flood!

That's why we began the chapter talking about Napster and Gnutella. Unfortunately, as can be seen in Gnutella's scaling problems, the massive amounts of Napster traffic, and flooding attacks on file storage services, being peer-to-peer doesn't make the problems go away. It just makes the problems different. Indeed, it often makes the problems harder to solve, because with peer-to-peer there might be no central command or central store of data.

The history of cryptography provides a cautionary tale here. System designers have realized the limits of theoretical cryptography for providing practical security. Cryptography is not pixie dust to spread liberally and without regard over network protocols, hoping to magically achieve protection from adversaries. Buffer overflow attacks and unsalted dictionary passwords are only two examples of easy exploits. A system is only as secure as its weakest link.

The same assertion holds for decentralized peer-to-peer systems. A range of techniques exists for solving accountability and resource allocation problems. Particularly powerful are reputation and micropayment techniques, which allow a system to collect and leverage local information about its users. Which techniques should be used depends on the system being designed.

Acknowledgments

First and foremost, we'd like to thank our editor, Andy Oram, for helping us to make all of these sections fit together and make sense. Without him, this would still be just a jumble of really neat ideas. We would also like to thank a number of others for reviews and comments: Robert Zeithammer for economic discussion of micropayments, Jean-François Raymond and Stefan Brands for micropayments, Nick Mathewson and Blake Meike for the reputations section, and Daniel Freedman, Marc Waldman, Susan Born, and Theo Hong for overall edits and comments.

Microsoft Exhibit 1019

chapter seventeen

Reputation

Richard Lethin, Reputation Technologies, Inc.

Reputation is the memory and summary of behavior from past transactions. In real life, we use it to help us set our expectations when we consider future transactions. A buyer depends on the reputation of a seller when he considers buying. A student considers the reputation of a university when she considers applying for admission, and the university considers the student's reputation when it decides whether to admit her. In selecting a candidate, a voter considers the reputation of a politician for keeping his word.

The possible effect on one's reputation also influences how one behaves: an individual might behave properly or fairly to ensure that her reputation is preserved or enhanced. In situations without reputation, where there is no prospect of memory after the transaction, behavior in the negotiation of the transaction can be zero-sum. This is the classic used car salesman situation in which the customer is sold a lemon at an unreasonable price, because once the customer drives off the lot, the salesman is never going to see her again.

A trade with a prospective new partner is risky if we don't know how he behaved in the past. If we know something about how he's behaved in the past, and if our prospect puts his reputation on the line, we will be more willing to trade. So reputation makes exchange freer, smoother, and more liquid, removing barriers of risk aversion that interfere with trade's free flow.

Reputation does all this without a central authority. Naturally, therefore, reputation turns up frequently in any discussion about distributed entities interacting peer-to-peer—a situation that occurs at many levels over the Internet. Some of these levels are close to real life, such as trade in the emerging e-marketplaces and private exchanges. Others are more esoteric, such as the interaction of anonymous storage servers in the Free Haven system described in Chapter 12. Chapter 16, *Accountability*, includes a discussion of the value of reputation.

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The use of reputation as a distributed means of control over fairness is a topic of much interest in the research literature. Economists and game theorists have analyzed the way reputation motivates fair play in repeated games, as opposed to a single interaction, which often results in selfish behavior as the most rational choice. Researchers in distributed artificial intelligence look to reputation as a system to control the behavior of distributed agents that are supposed to contribute collectively to intelligence. Researchers in computer security look at deeper meanings of trust, one of which is reputation.

In this chapter, I will present a commercial system called the Reputation Server™* that tries to bring everyday aspects of reputation and trust into online transactions. While not currently organized in a peer-to-peer fashion itself, the service has the potential to become more distributed and prove useful to peer-to-peer systems as well as traditional online businesses.

The Reputation Server is a computer system available to entities engaging in a prospective transaction—a third party to a trade that can be used by any two parties who want reputation to serve as motivation for fair dealing.

The server accepts feedback on the performance of the entities after each transaction is finished and stores the information for use by future entities. It also provides scores summarizing the history of transactions that an entity has engaged in. The Reputation Server, by holding onto the histories of transactions, acts as the memory that helps entities build reputations.

Examples of using the Reputation Server

A North American buyer of textiles might be considering purchasing from a new supplier in China. The buyer can check the Reputation Server for scores based on feedback from other buyers who have used that supplier. If the scores are good enough to go forward, the buyer will probably still insist that the trade be recorded in a transaction context on the Reputation Server—that the seller be willing to let others see feedback about its performance—in order to make it costly for the seller to perform poorly in the transaction. Without the Reputation Server, the buyer has to rely solely on other means of reducing risk, such as costly product inspections or insurance.†

* Reputation Server™ is a trademark of Reputation Technologies, Inc.

† These other risk reduction techniques can also be used with the Reputation Server.

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But the motivation to use the Reputation Server is not exclusively on the buyer's side: A reliable seller may insist on using the Reputation Server so that the trade can reinforce his reputation.

In some cases, the Reputation Server may be the only way to reduce risk. For example, two entities might want to trade in a securely pseudonymous manner, with payment by a nonrepudiable anonymous digital cash protocol. Product inspection might be unwanted because it reveals the entity behind the pseudonym. Once the digital cash is spent, there's no chance of getting a refund. Reputation helps ease some of the buyer's concern about the risk of this transaction: she can check the reputation of the pseudonym, and she has the recourse of lowering that reputation should the transaction go bad. Thus, the inventors of anonymous digital cash have long recognized the interdependence of pseudonymous commerce systems and reputations. Also, the topic gets attention in the Cypherpunks *Cyphernomicon* as an enabling factor in the adoption of anonymous payment technologies.*

But more mundane risks can also make using the Reputation Server worthwhile. The example I started with in this section, of a buyer in North America purchasing textiles from China, has some aspects of functional anonymity: even though the buyer and seller aren't actively hiding from each other, they don't know each other because of the geographic, political, cultural, class, and language barriers that separate them. Reputation Servers can be the social network that is otherwise lacking and that enforces good behavior or allows the system to correct itself. As the Internet bridges the traditional barriers to create new relationships, the need for Reputation Servers grows.

At first, the implementation of this system seems trivial: just a database, some messaging, and some statistics. However, the following architecture discussion will reveal that the issues are quite complex. With keen competition and high-value transactions, the stakes are high. This makes it important to consider the design carefully and take a principled approach.

Reputation domains, entities, and multidimensional reputations

To understand how the Reputation Server accomplishes its task, you have to start with the abstraction of a *reputation domain*, which is a context in which a

* Tim May (1994). *The Cyphernomicon*, Sections 15.2–4, archived in various places on the Net, e.g., *http://swissnet.ai.mit.edu/6805/articles/crypto/cypherpunks/cyphernomicon/CP-FAQ* 1994.

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sequence of trades will take place and in which reputations are formed and used. A domain is created, administered, and owned by one entity. For example, a consultant integrating the software components for a business-to-business, online e-marketplace might create a reputation domain for that e-marketplace on the Reputation Server. Thousands of businesses that will trade in the e-marketplace can use the same domain. Or someone might create a smaller domain consisting of auto mechanics in Cambridge and the car owners that purchase repairs. Or someone might create a domain for the anonymous servers forming Free Haven.

The domain owner can specify the domain's rules about which entities can join, the definition of reputation within that domain, which information is going to be collected, who can access the data, and what they can access. Reputations form within the domain according to the specified configuration. For the moment, we assume that there is no information transfer among domains: A reputation within one domain is meaningless in another domain.*

Entities in a Reputation Server correspond to the parties for whom reputations will be forming and the parties who will be providing feedback. Entities might correspond to people, companies, software agents, or Pretty Good Privacy (PGP) public keys. They exist outside the domains, so it is possible for an entity to be a participant in multiple domains.

The domain has a great degree of latitude in how it defines reputation. This definition might be a simple scalar quantity representing an overall reputation, or a multidimensional quantity representing different aspects of an entity's performance in transactions. For example, one of the dimensions of a seller's reputation might be a metric measuring the quality of goods a seller ships; another might be the ability to ship on time. The scoring algorithms do not depend on what the individual dimensions "mean"; the dimensions are measures within a range, and the domain configuration simply names them and hooks them up to sources and readers.

The notion of a domain is powerful, even for definitions that might be considered too small to be meaningful. For example, a domain with only one buyer seems solipsistic (self-absorbed) but can in fact be quite useful to an entity for privately monitoring its suppliers. The domain can provide a common area for the storage and processing of quality, docking, and exception information that might otherwise be used by only one small part of the buyer's organization or simply lost outright.

* This constraint is relaxed later in the chapter.

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Reputation information about a supplier might be kept internal to the buyer if the buyer thinks this is of strategic importance (that is, if knowing which supplier is good or bad in particular areas conveys a competitive advantage to the buyer). On the other hand, if the buyer is willing to share the reputation information he has taken the trouble to accumulate, it could be useful so that a seller can attract other buyers. For example, ACME computer company might allow its ratings of suppliers to be shared outside to help its suppliers win other buyers; this benefits ACME by allowing its suppliers to amortize fixed costs, and it might even be able to negotiate preferred terms from the supplier to realize this benefit.

Identity as an element of reputation

Before gaining a reputation, an entity needs to have an identity that is made known to the Reputation Server. The domain defines how identities are determined.

Techniques for assuring an entity's identity are discussed in other areas of this book, notably Chapter 15, *Trust*, and Chapter 18, *Security*. An entity's identity, for instance, might be a certified public key or a simple username validated with password login on the Reputation Server.

Some properties of identities can influence the scoring system. One of the most critical questions is whether an entity can participate under multiple identities. Multiple participation might be difficult to prevent, because entities might be trivially able to adopt a new identity in a marketplace. In this situation, with weak identities, we have to be careful how we distinguish a bad reputation from a new reputation. This is because we may create a moral hazard: the gain from cheating may exceed the loss to reputation if the identity can be trivially discarded and a new identity trivially constructed. Weak identities also have implications for credibility, because it becomes hard to distinguish true feedback from feedback provided by the entity itself.

While it is possible to run a reputation domain for weak identities, it is easier to do so for strong identities. Reputation domains with weak identities require the system to obtain and process more data, while strong identities allow the system to "bootstrap" online reputations with some grounding in the real world.

Interface to the marketplace

We use the term marketplace loosely: generally it corresponds to an online e-marketplace, but a marketplace might also correspond to the distributed block

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trading that is taking place in Free Haven or the private purchasing activity of the single buyer who has set up a private reputation domain. While some marketplaces, such as eBay, include an embedded reputation system, our Reputation Server exists outside the marketplace so that it can serve many marketplaces of different types.

The separation of the Reputation Server from the marketplace creates relatively simple technical issues as well as more complex business issues. We discuss some of the business issues later in the section "Long-term vision." The main technical issue is that the marketplace and the Reputation Server need to communicate. This is easy to solve: The Internet supports many protocols for passing messages, such as email, HTTP, and MQ. The XML language is excellent for exchanging content-rich messages.

One of the simple messages that the marketplace can send to the Reputation Server indicates the completion of a transaction. This message identifies the buyer and seller entities and gives a description of the type of transaction and the monetary value of the transaction. The description is important: A reputation for selling textiles might not reflect on the ability to sell industrial solvents.

The transaction completion message permits the Reputation Server to accept feedback on the performance of entities in the transaction. For some domains, it also triggers the Reputation Server to send out a request for feedback on the transaction. In the most rudimentary case, the request for feedback and the results could be in electronic mail messages. Since a human being has to answer the email request for feedback, some messages may be discarded and only some transactions will get feedback. For this reason, obviously, it is preferable to automate the collection. So some businesses may interface the trader's Enterprise Resource Planning (ERP) systems into the Reputation Server. For automated peer-to-peer protocols like Free Haven, an automated exchange of feedback will be easier to generate.

The marketplace and the Reputation Server will also exchange other, more complex messages. For example, the marketplace might send a message indicating the start of a potential transaction. Some transactions take a long time from start to finish, perhaps several weeks. Providing the Reputation Server with an early indication of the prospective transaction allows the Reputation Server to provide supplementary services, such as messages indicating changes in reputation of a prospective supplier before the transaction is consummated.

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Scoring system

One of the most interesting aspects of the Reputation Server is the scoring system, the manner in which it computes reputations from all of the feedback that is has gathered.

Why bother computing reputations at all? If, as asserted in the first sentence of this chapter, "Reputation is the memory and summary of behavior from past transactions," why not simply make the reputation be the complete summary of all feedback received, verbatim? Some online auctions do in fact implement this, so that a trader can view the entire chain of feedback for a prospective partner. This is okay when the trader has the facility to process the history as part of a decision whether to trade or not.

But more often, there is good reason for the Reputation Server to add value by processing the chain into a simple reputation score for the trader. First, the feedback chain may be sensitive information, because it includes a description of previous pricing and the good traded. Scoring algorithms can mask details and protect the privacy of previous raters. This trade-off between hiding and revealing data is more subtle than encryption. Encryption seeks to transform data so that, to the unauthorized reader, it looks as much like noise as possible. With reputation, there is a need to simultaneously mask private aspects of the transaction history—even to the authorized reader—while allowing some portion of the history through so it can influence the reputation. Some of this is accomplished simply by compressing the multiple dimensionality of the history into a single point, perhaps discretizing or adding another noise source to the point to constrain its dimensionality.

Furthermore, the Reputation Server has a more global view of the feedback data set than one can learn from viewing a simple history listing, and it can include other sources of information to give a better answer about reputation. Stated bluntly, the Reputation Server can process a whole bunch of data, including data outside the history. For example, the Reputation Server may have information about the credibility of feedback sources derived from the performance of those sources in other contexts.

Reputation metrics

The Reputation Server is a platform for multiple scoring functions, and each domain can choose the kinds of scoring used and the functions that compute the scores.

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A number of reputation metrics have been proposed in the literature. Some simply provide ad hoc scales, dividing reputations into discrete steps or assigning boundaries and steps arbitrarily. While ad hoc definitions of reputation can seem reasonable at first, they can have undesirable properties.* For example, simply incrementing reputation by one for each good transaction and decrementing by one for each bad transaction allows a reputation to keep growing indefinitely if a seller cheats one buyer out of every four. If the seller does a lot of volume, she could have a higher reputation in this system than someone who trades perfectly but has less than three quarters the volume. Other reputation metrics can have high sensitivity to lies or losses of information.

Other approaches to reputation are principled.† One of the approaches to reputation that I like is working from statistical models of behavior, in which reputation is an unbound model parameter to be determined from the feedback data, using Maximum Likelihood Estimation (MLE). MLE is a standard statistical technique: it chooses model parameters that maximize the likelihood of getting the sample data.

The reputation calculation can also be performed with a Bayesian approach. In this approach, the Reputation Server makes explicit prior assumptions about a probability distribution for the reputation of entities, either the initial distribution that is assumed for every new entity or the distribution that has previously been calculated for entities. When new scores come in, this data is combined with the previous distribution to form a new posterior distribution that combines the new observations with the prior assumptions.

Our reputation scores are multidimensional vectors of continuous quantities. An entity's reputation is an ideal to be estimated from the samples as measured by the different entities providing feedback points. An entity's reputation is accompanied by an expression of the confidence or lack of confidence in the estimate.

Our reputation calculator is a platform that accepts different statistical models of how entities might behave during the transaction and in providing feedback. For example, one simple model might assume that an entity's performance rating follows a normal distribution (bell) curve with some average and standard deviation. To make things even simpler, one can assume that feedback is always given

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* Raph Levien and Alexander Aiken (1998), "Attack-Resistant Trust Metrics for Public Key Certification," *Proceedings of the 7th USENIX Security Symposium*, UNIX Assoc., Berkeley, CA, pp. 229–241.

† Michael K. Reiter and Stuart G. Stubblebine, "Authentication Metric Analysis and Design," *ACM Transactions on Information Systems and Security*, vol. 2, no. 2, pp. 138–158.

Microsoft Exhibit 1019

honestly and with no bias. In this case, the MLE is a linear least squares fit of the feedback data.

This platform will accept more sophisticated reputation models as the amount of data grows. Some of the model enhancements our company is developing are described in the following list:

- Allowing dynamic reputation. Without this, reputation is considered a static quantity with feedback data providing estimates. If an entity's reputation changes, the estimate of reputation changes only with the processing of more feedback data. When we incorporate drift explicitly, confidence in the reputation estimate diminishes without feedback data.
- Incorporating source feedback models. With multiple ratings given by the same party, we can estimate statistically their bias in providing feedback. This might even permit the identification of sources that are not truthful.
- Allowing performance in one context to project the entity's ability to perform in another context. For instance, the ability to sell shoes is some prediction of the ability to sell clothes.

The rate of reputation drift, the related weight assigned to more recent feedback, biases, the estimate of the credibility of sources, and contextual correlation become additional free parameters to be chosen by the MLE solver. Getting good estimates of these parameters requires more data, obviously.

A property of this approach is that reputation does not continue increasing arbitrarily as time advances; it stays within the bounds established when the reputation domain was configured. Additional data increase the data points on which the extracted parameters are based, so as a trader earns more feedback, we usually offer greater confidence in her reputation. Confidence is not being confused with the estimate of reputation.

It's interesting to think about how to incorporate the desire to punish poor performance quickly (making reputation "hard to build up, and easy to tear down") into the model-based approach. It seems reasonable to want to make the penalty for an entity's behaving in a dishonest way severe, to deter that dishonest behavior. With an ad hoc reputation-scoring function, positive interactions can be given fewer absolute reward points than absolute punishment points for negative behavior. But how is the ratio of positive to negative feedback chosen? There are a number of approaches that permit higher sensitivity to negative behavior.

One approach is to increase the amount of history transmitted with the reputation so the client's decision function can incorporate it. If recent negative behavior

Microsoft Exhibit 1019

is of great concern, the reputation model can include a drift component that results in more weight toward recent feedback. Another approach is to weight positive and negative credibility differently, giving more credence to warnings.

The design choices (including ad hoc parameter choices) depend intimately on the goals of the client and the characteristics of the marketplace. Such changes could be addressed by adapting the model to each domain, by representing the assumptions as parameters that each domain can tune or that can be extracted mechanically, and perhaps even by customizing the reputation component in a particular client.

How is MLE calculated? For simple models, MLE can be calculated analytically, by solving the statistical equations algebraically. Doing MLE algebraically has advantages: The answer is exact, updates can be computed quickly, and it is easier to break up the calculation in a distributed version of a Reputation Server. But an exact analytical solution may be hard to find, nonexistent, or computationally expensive to solve, depending on the underlying models. In that case, it may be necessary to use an approximation algorithm. However, some of these algorithms may be difficult to compute in a distributed manner, so here a centralized Reputation Server may be better than a distributed one.

Credibility

One of the largest problems for the Reputation Server is the credibility of its sources. How can a source of feedback be trusted? Where possible, cryptographic techniques such as timestamps and digital signatures are used to gain confidence that a message originates from the right party. Even if we establish that the message is truly from the correct feedback source, how do we know that the source is telling the truth? This is the issue of source credibility, and it's a hairy, hairy problem.

We address this in our Reputation Server by maintaining credibility measures for sources. These credibility measures factor into the scoring algorithms that form reputations—both our estimated reputation and the confidence that our service has in the estimate. Credibility measures are initialized based on heuristic judgments, and then updated over time using the Bayesian/MLE framework previously described. Sources that prove reliable over time increase their credibility. Sources that do not prove reliable find their credibility diminished.

This process can be automated through the MLE solver and folded into the scoring algorithm. Patterns of noncredible feedback are identified by the algorithm and given lower weights. Doing this, though, requires something more than the

Microsoft Exhibit 1019

accumulated feedback from transactions; we should have an external reference or benchmark source of credible data. One way that we solve this is by allowing the domain configuration to designate benchmark sources. The Reputation Server assigns high credibility to those sources because the designation indicates that there is something special backing them up, such as a contractual arrangement, bonding of the result, or their offline reputation. In a sense, credibility flows from these benchmark sources to bootstrap the credibility of other sources.

Interdomain sharing

Popular online marketplaces such as auctions have rudimentary reputation systems, providing transaction feedback for participants. These marketplaces strongly protect their control over the reputations that appear on their site, claiming they are proprietary to the marketplace company! The marketplaces fight cross-references from other auctions and complete copying of reputations with lawsuits, and they discourage users from referring to their reputations from other auctions.

These practices raise the question: Who owns your reputation? The popular auction sites claim that they own your reputation: It is their proprietary information. It is easy to understand why this is the case. Portable reputations would be a threat to the auction sites, because they reduce a barrier to buyers and suppliers trading on competitor auctions. Portable reputations make it more difficult for auctions to get a return from their investment in technology development and marketing that helped build the reputation.

The Reputation Server supports auction sites by isolating the reputation domains unless the owners of the domains permit sharing. In cases where the sharing can be economically beneficial, the scoring algorithms can permit joining the data of two domains to achieve higher confidence reputations. This is performed only with the permission of the domain owners.

Bootstrapping

One obstacle to the use of the Reputation Server is a bootstrapping or chicken-and-egg problem. While the server is of some use even when empty of transaction histories (because it serves as a place where entities can put their reputations on the line), it can be difficult to convince a marketplace to use it until some reputation information starts to appear.

Microsoft Exhibit 1019

Consequently, our server offers features to bootstrap reputations similar to the way reputations might be bootstrapped in a real-world domain: through the use of references. A supplier entering the system can supply the names of trade references and contact information for those references. The server uses that contact information to gather the initial ratings. While the reference gathering process is obviously open to abuse, credibility metrics are applied to those initial references. To limit the risk of trusting the references from outside the reputation system, those credibility metrics can signal that the consequent reputation is usable only for small transactions. As time passes and transactions occur within the reputation system, the feedback from transactions replaces the reference-based information in the computation of the reputation.

Long-term vision

Business theorists have observed that the ability to communicate broadly and deeply through the Internet at low cost is driving a process whereby large businesses break up into a more competitive system of smaller component companies. They call this process "deconstruction."* This process is an example of Coase's Law, which states that other things being equal, the cost of transacting—negotiating, paying, dealing with errors or fraud—between firms determines the optimal size of the firm.† When business transactions between firms are expensive, it's more economical to have larger firms, even though larger firms are considered less efficient because they are slower to make decisions. When transactions are cheaper, smaller firms can replace the larger integrated entity.

As an example, Evans and Wurster point to the financial industry. Where previously a bank provided all services like investments and mortgages, there are now many companies on the Internet filling small niches of the former service. Aggregation sites find the best mortgage rate out of hundreds of banks, investment news services are dedicated solely to investment news feeds, and so on. Even complex processes like the manufacturing of automobiles—already spread over chains of multiple companies for manufacturing parts, chassis, subsystems—could be further deconstructed into smaller companies.‡

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* Philip Evans and Thomas Wurster (2000). *Blown to Bits: How the New Economics of Information Transforms Strategy*. Harvard Business School Press.

† Ronald Coase (1960). "The Problem of Social Cost," *Journal of Law and Economics*, vol. 3, pp. 1–44.

‡ Clayton M. Christenson (1997) *The Innovator's Dilemma*. Harvard Business School Press.

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With more entities, there is an increased need for tracking reputations at the interaction points between them. At the extreme, a firm might completely deconstruct: One vision is that the substations that currently make up a factory can become independent entities, all transacting in real time and automatically to accomplish the manufacturing task that previously occurred in the single firm. The Reputation Server, as one of the components reducing the cost of transacting between firms, serves as a factor to assist in this deconstruction, which results in lower manufacturing costs.

Central Reputation Server versus distributed Reputation Servers

The first version of the Reputation Server is a centralized web server with a narrow messaging interface. One could well argue that it should be decentralized so that the architecture conforms to our ultimate goal: to provide fairness in a non-centralized manner for peer-to-peer networks.

Can we design a network of distributed Reputation Servers? Yes, in some cases, such as when the reputation metric computation can be executed in a distributed fashion and can give meaningful results with partial information. Not all reputation metrics have these properties, however, so if the design goal of a distributed server is important, we should choose one that does.

Summary

Reputation is a subtle and important part of trade that motivates fair dealing. We have described technologies for translating the reputation concept into electronic trade, applicable to business transactions and peer-to-peer interaction. The Reputation Server provides these technologies. Scoring algorithms based on MLE and Bayesian techniques estimate reputations based on feedback received when trades occur. We describe enhancements for addressing the credibility of sources. Reputation domains, which are an abstraction mapped to the client marketplace, serve to store the configuration of rules about how reputations form for that marketplace, allowing the Reputation Server to be a platform for many different reputation systems.

Microsoft Exhibit 1019

chapter eighteen

Security

Jon Udell, BYTE.com, and Nimisha Asthagiri and Walter Tuvell, Groove Networks

Security is hard enough in traditional networks that depend on central servers. It's harder still in peer-to-peer networks, particularly when you want to authenticate your communication partners and exchange data only with people you trust. Earlier chapters stressed protection for users' anonymity. The need to assert identity is actually more common than the need to hide it, though the two are not mutually exclusive. As shown in Chapter 16, *Accountability*, systems that assign pseudonyms to users need not absolve users of responsibility. This chapter touches on the interplay of identity and pseudonymity too, but will mainly focus on how to authenticate users and ensure they can communicate securely in a peer-to-peer system.

At Groove Networks Inc., we've developed a system that provides a type of strong security consistent with Groove's vision of a peer-to-peer system. The details are described in this chapter. We hope that our work can serve not only as proof that traditional conservative security principles can coexist with a novel distributed system, but also as a guide to developers in other projects. Groove is a peer-to-peer groupware system. Before we focus on its security architecture, we should first explain its goals and the environment in which it operates. Using Groove, teams of collaborators form spontaneous *shared spaces* in which they collect the documents, messages, applications, and application-specific data related to group projects. The software (which is available for Windows now and for Linux soon) works identically for users on a LAN, behind corporate firewalls, behind DSL or cable-modem Network Address Translation (NAT), on dial-up connections with dynamic IP addresses, or in any combination of such circumstances. The key benefits of Groove shared spaces are:

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Spontaneity

Groove needs no administrator. Nobody has to wait for IT to create the support system for a project. Users can do this for themselves, easily and right away.

Security

Shared spaces are, in effect, instant virtual private networks (VPNs).

Context

The shared space provides a context that helps users understand the nature, purpose, and history of all the messages and documents related to an activity.

Synchronization

Shared spaces synchronize automatically among all members' devices and among all Groove devices owned by each member. Users can work offline; changes automatically synchronize when they reconnect.

Granularity

Groove users don't typically exchange whole documents (though conventional file sharing is supported). Rather, they exchange incremental edits to documents. Groove-aware applications can even enable shared editing in real time.

Groove is really a new kind of Internet-based platform that delivers basic support for collaboration—in particular, security and synchronization. Users automatically enjoy these services with no special effort. Developers can build on them without needing to reinvent the wheel. In terms of data synchronization, Groove arguably breaks new technical ground with its distributed, transactional, serverless XML object store. But in terms of security—the focus of this chapter—Groove relies on tried-and-true techniques. What's novel isn't the algorithms and protocols, but rather the context in which they are used. Groove enables spontaneous peer-to-peer computing while at the same time abolishing the human factors problems that bedevil real-world security.

The environment in which Groove does all this is a hostile one. Firewall/NAT barriers often separate members of a group. Even within a group, people do not necessarily trust one another and do not typically share a common directory service or Public Key Infrastructure (PKI). People aren't always online, and when they are, they're not always using the same computer. People connect to the Net in different ways, using channels with very different bandwidths and latencies, so that, for example, an encrypted message may arrive before the message bearing its decryption key. Groups are dynamic; membership is fluid and constantly changing. The unit of secured data—that is, data that is authenticated, encrypted, and guaranteed not to have been tampered with—is not typically a

Microsoft Exhibit 1019

whole document, but rather an incremental change (or *delta*), possibly an individual keystroke.

In the face of this hostile environment, Groove makes an impressive set of security guarantees to users. Here are some of them:

- Strong security is always in force. No user or administrator can accidentally or intentionally turn it off.
- All shared-space data is confidential. It's encrypted not only on the wire, where it's readable and writable by only group members, but also on disk, where it's readable and writable by only the owner of that copy of the data.
- No group member can impersonate another group member or tamper with the contents of any group message.
- A lost message can be recovered from any member, with assurance of the integrity of the recovered message and proof of its true originator.
- No nonmember or former member who has been uninvited from the group can eavesdrop on or tamper with group communication.

How Groove implements these security guarantees, thereby accomplishing its mission to deliver flexible and secure groupware in a hostile environment, is the subject of this chapter. We'll explore the implementation in detail, but first let's consider how and why Groove is like and unlike other groupware solutions.

Groove versus email

The world's dominant groupware application is email. Like Groove, email enables users to create primitive "shared spaces" that contain both messages and documents (i.e., attachments). Nobody needs to ask an administrator to create one of these shared spaces. We do it quite naturally by addressing messages to individuals and groups. Because firewalls are always permeable to email, we can easily form spaces that include people behind our own firewalls and people behind foreign firewalls. Email enables us to modify group membership on the fly by adjusting the *To:* and *Cc:* headers of our messages, adding or dropping members as needed. This is powerful stuff. It's no wonder we depend so heavily on it.

To the extent that we exchange sensitive information in email, though, we incur serious risks. People worry about the efficacy of the SSL encryption that guards against theft of a credit card number during an online shopping transaction. Yet they're oddly unconcerned about sending completely unencrypted personal and business secrets around in email. Secrets stored on disk typically enjoy no more

Microsoft Exhibit 1019

protection than do secrets sent over the wire, a fact deeply regretted by the Qualcomm executive whose notebook computer was recently stolen.

Although it is convenient in many important ways, email is terribly inconvenient in others. The shared space of a group email exchange is a fragmentary construct. There is no definitive transcript that gathers all project-related messages and documents into a single container that's the same for all current (and future!) group members. Newsgroups, web forums, and web-accessible mail archives (such as Hypermail) or document archives (such as CVS) can make collaboration a more coherent and controlled exercise. But the IT support needed for these solutions is often missing within organizations, and especially across organizational boundaries.

There is, to be sure, an emerging breed of hosted collaborative solutions that make shared spaces a do-it-yourself proposition for end users. Anyone can go to eGroups (*http://www.eGroups.com*), for example, and create a project space for shared messages and documents. But eGroups provides only modest guarantees as to the privacy of such spaces, and none with respect to the integrity and authenticity of messages exchanged therein. What's more, services similar to eGroups fail the convenience test when users are connected poorly, or not at all. In these cases, users wind up manually replicating data to their local PCs—a procedure that is arduous, error-prone, and thus insecure.

Security, as cryptographer and security consultant Bruce Schneier likes to observe, is a process. When that process is too complex—which is to say, when it requires just about any effort or thought—people will opt out, with predictably disastrous results.

Collaboration places huge demands on any security architecture. It's a convenient fiction to believe that we are all safe behind our corporate firewalls, where we can form the groups in which we do our work, and create and exchange the documents that are the product of that work. But we never were safe behind the firewall, and the fiction grows less believable all the time as email worms burrow through firewalls and wreak havoc.

Furthermore, in a company of any substantial size, the firewall-protected realm cannot usefully be regarded as an undifferentiated zone of trust. Real people doing real work will want to form spontaneous workgroups; these workgroups ought to be isolated from one another. When we rely only on the firewall, we create the kind of security architecture that hackers call "crunchy on the outside, soft and chewy on the inside."

We need more granular security, distributed at the workgroup level rather than centralized in the firewall. Historically, people could form password-protected

Microsoft Exhibit 1019

group spaces on departmental servers or even among their own peer-enabled PCs. But if the internal network is compromised, a sniffer anywhere on the LAN can scoop up all the unencrypted data that it can see. Likewise, if a server or desktop PC is compromised, the intruder (possibly a person with unauthorized physical access, possibly a virus) can scoop up all available unencrypted data.

The LAN, in any case, is a construct that few companies have successfully exported beyond the firewall to the homes, hotel rooms, public spaces, and foreign corporate zones in which employees are often doing their collaborative work. In theory, virtual private networks extend the LAN to these realms. In practice, for many companies that doesn't yet happen. When it does, there is typically only protection on the wire, not complementary protection on the disk.

So far, all these models assume that collaboration is an internal affair—that we work in groups under the umbrella of a single corporate security infrastructure. For many real-world collaborative projects, that assumption is plainly false. Consider the project that produced this chapter. Two of the authors (Nimisha Asthagiri and Walt Tuvell) are employees of Groove Networks, Inc. Another (Jon Udell) is an independent contractor. Beyond this core team, there was the editor (Andy Oram, an employee of O'Reilly & Associates, Inc.), and a group of reviewers with various corporate and academic affiliations. Projects like this aren't exceptions. They're becoming the norm.

To support our project, one of the authors created a Groove shared space. There, we used a suite of applications to collaborate on the writing of this chapter: persistent chat, a shared text editor, a discussion tool, and an archive of highly confidential Groove Networks security documents. As users of the shared space, we didn't have to make any conscious decisions or take any explicit actions to ensure the secure transmission and storage of our data. Under the covers, of course, were powerful security protocols that we'll explore in this chapter.

Why secure email is a failure

Before we dive into the details of Groove's security system, let's look again at the big picture. It's instructive to ask, "Why couldn't ordinary secure email support the kind of border-crossing collaboration we've been touting?" PGP, after all, has been widely available for years. Likewise S/MIME, which lies dormant within the popular mail clients. These are strong end-to-end solutions, delivering both on-the-wire and on-disk encryption. Why don't we routinely and easily use these tools to secure our shared email spaces? Because it's just too hard. In the case of PGP, users must acquire the software and integrate it with their email programs.

Microsoft Exhibit 1019

Then they confront a daunting user interface which, according to a study called *Why Johnny Can't Encrypt*,* few are able to master. S/MIME, though built into common email programs (Netscape Messenger and Microsoft Outlook Express), requires users to acquire client certificates (VeriSign calls them "digital IDs") that unleash signing and encryption.†

The next weakness of email shared spaces is that they aren't as coherent as we need, or as functional. Email is a good way to exchange interpersonal messages but a poor medium for group discussion and document archiving. When we ask it to serve these functions—as we often do, lacking other convenient tools—the result is a mess. Documents and pieces of conversations end up scattered across a bunch of computers. People get confused and waste time because they can't find everything related to the collaboration in one place; there's no single, consistent view of the project's data.

Finally, email can be a little too spontaneous for our own good. Information can leak out of an email shared space when anyone "ccs" someone else. That kind of spontaneity is a wonderful thing, and it's vital for effective collaboration. But it's not always a good idea to enable anybody in a shared space to include anybody else. People can leak information because of malice, poor judgment, or just operator error. The kinds of groups that form in email shared spaces are just too loosely defined. There's no way to balance the necessary freedom of spontaneous group formation with the equally necessary control of a centrally determined policy that governs modes of group formation.

The authors of this article could have used the S/MIME capabilities of our respective mail readers instead of Groove. And, in fact, we tried that experiment. But even for the three of us, all well versed in crypto software, S/MIME presented daunting configuration and use challenges. In any case, S/MIME only

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* *http://www.cs.cmu.edu/~alma/johnny.pdf.*

† To the extent that we have any routine on-the-wire encryption at all on the Web, it's in the form of SSL-protected shopping carts. This works because although servers have certificates that authenticate them to clients (and that enable the SSL handshake to occur), almost no clients have certificates that reciprocally authenticate them to servers. It's unfortunate that this is so. E-commerce ought to have much more robust client authentication than it does. But the Public Key Infrastructure (PKI) gymnastics that server administrators are required to perform go way beyond what normal people are willing to put up with. So we settle for one-way, server-only authentication on the secure Web. And the only reason we have at least this much security is that it was possible to make it a no-brainer, an out-of-the-box default for a web shopper. You click the link, you see the golden key, it's "secure"—at least in a limited sense of that term, notwithstanding the risk of a hostile takeover of your PC, spoofing of the server's identity (since nobody actually checks the certificate sent from the server), or capture of your credit card number after decryption on the server.

Microsoft Exhibit 1019

governs the email domain. It doesn't empower users to form the coherent but replicated multiapplication workspace that makes Groove so effective. For us, the benefits of a secure shared space far outweighed the learning curve presented by Groove. The same will hold true for most typical Groove users.

The solution: A Groove shared space

Now we can start to put in place the foundations of a better data exchange system. A shared space is a copy of an XML object store. Incremental changes to objects are transmitted to all Groove devices participating in a shared space in the form of Groove delta messages. These messages may carry pieces of application data (a line of text in a chat, a stroke in a sketch) or pieces of administrative data (an invitation to join a shared space, a cryptographic key). The distributed communications engine ensures that delta messages are reliably delivered to and stored on each node. It adapts as needed (sometimes with the assistance of a central relay service) in order to reach nodes that are offline, that don't have fixed IP addresses, or that are behind firewalls and NATs.

Users see none of this plumbing.* They just interact with a viewer/editor called the transceiver. Figure 18-1 is a picture of a Groove transceiver displaying the shared space that we used to collaborate on this chapter.

The shared space is defined by a set of members, a set of tools, and the data created by members using those tools. In our case, the primary tools used for the collaboration included a persistent chat session, several threaded discussions, several notepads, a calendar, and a file archive. The tools are Groove-aware, which means that any change made in a notepad, discussion, or calendar propagates to all instances of the shared space. For an individual member, this synchronization may involve two or more devices (such as a desktop PC and a notebook PC).

Security characteristics of a shared space

In normal operation, a shared space has a fixed (steady state) population of members. Delta messages are being exchanged among the devices (one or more per user) that access the shared space. The kind of security that governs these messages may vary, for two different reasons. First, the Groove system can sup-

* In its base version, Groove does not require nor support the notion of a shared-space administrator. However, Groove Networks expects to offer an enterprise version of Groove that gives IT personnel more control over shared-space policy.

Microsoft Exhibit 1019

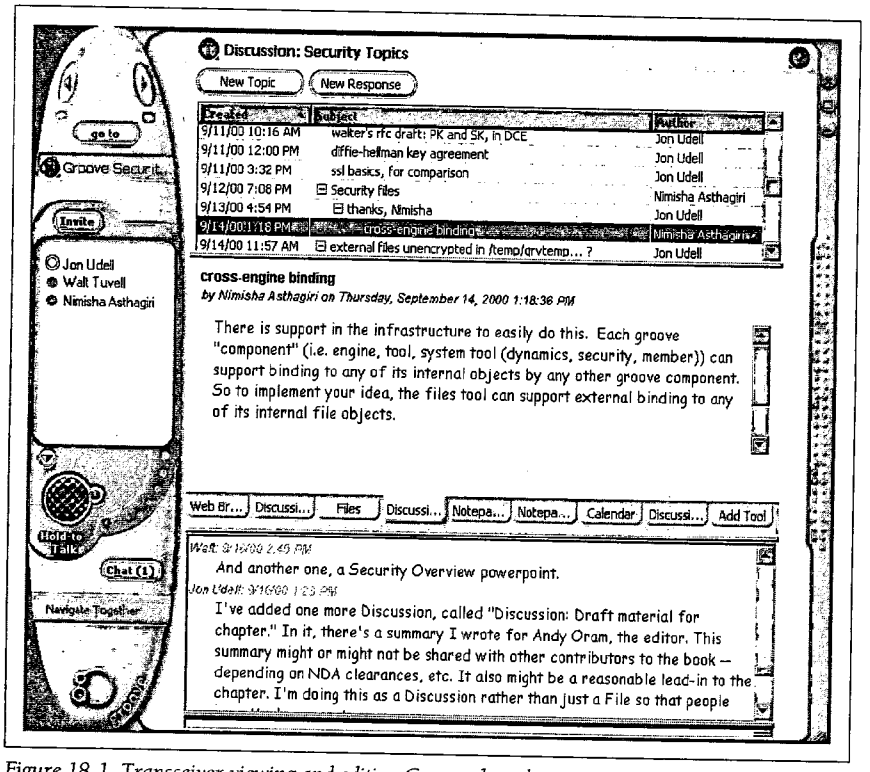

Figure 18-1. Transceiver viewing and editing Groove shared space

port different cryptographic algorithms (and key lengths) for each shared space. The default asymmetric algorithm for authentication and key exchange is ElGamal with a 1536-bit modulus. The default symmetric algorithm for bulk encryption on the wire is MARC4* with a 192-bit key.† But hooks exist to enable someone to use these algorithms in one shared space and other algorithms in a different shared space. These other algorithms might be stronger versions of ElGamal/MARC4 (though the Groove defaults are already massively strong) or different algorithms altogether, such as RSA/Blowfish. A Groove user

* Because the symbol RC4 is trademarked by RSA Data Security, Inc. (now RSA Security, Inc.), Groove uses the symbol MARC4, which stands for "Modified-Alleged-RC4." "Alleged" refers to a freely available algorithm that's plug-compatible with RSA Data Security, Inc.'s RC4. "Modified" means that the first 256 bytes of the keystream are discarded, to thwart a weak-key attack.

† In symmetric encryption, a single key is used for both encryption and decryption; in asymmetric encryption, the decryptor keeps a private key and the encryptor uses a public key. Symmetric encryption is more efficient, hence it's used wherever possible, subject to security considerations. For more information, see Chapter 15, *Trust*.

Microsoft Exhibit 1019

can choose different cryptographic algorithms to transact business with different organizations in parallel shared spaces. Imagine a consultant who works with the NSA on some projects and with a university on others.

Security for shared spaces can also vary in another way. The protocols that govern message encryption and authenticity/integrity can vary independently along these dimensions:

Encryption/confidentiality
: Who can read delta messages exchanged within the shared space? In theory, encryption can be turned off, just as SSL can use a null cipher. In practice, it's not interesting or useful to do this, and Groove doesn't.

Data-origin authenticity/integrity
: Is every delta message guaranteed to have come from its purported sender (and, perforce, not to have been tampered with)? Again, it's possible to turn off authenticity/integrity, although Groove doesn't. Messages are always authenticated and integrity-protected. There are, however, two flavors. A delta message may be guaranteed only to have come from some unspecified member of the group, or guaranteed to have come from a specific member. Each shared space, for its lifetime, uses one or another of these modes. The first, which we'll call "mutual trust," is the most common and least computationally expensive. The second, which we'll call "mutual suspicion," is less common and involves more overhead.

Mutually-trusting shared spaces

Let's assume a shared space in "mutual trust" mode—the most convenient and likely setup—and review the security model from that perspective. We'll also show why you might want to switch to "suspicious" mode.

Anatomy of a mutually-trusting shared space

Every Groove user maintains an account on one or more devices. An account is a container of identities. A user can project a single identity into all shared spaces but is not restricted to a single identity. Carol can be just "Carol" in spaces she shares with her friends, but "Carol Smith, Marketing Director" in spaces she shares with colleagues. Technically, a member of a shared space is an instantiation of an identity as a participant in that space; the same identity may be a member of many shared spaces.

Each identity is defined by two public/private key pairs—one for signing and verification, one for encryption and decryption. The private half of each of these

Microsoft Exhibit 1019

key pairs is stored in the account. The public half is stored in the shared space, accessible to all members, and also (optionally) in each user's Groove Contacts (an address book), to protect and authenticate Groove instant messages. These Groove instant messages are exchanged among Groove identities, but outside the context of a shared space. So they're encrypted with one-time symmetric keys that are exchanged via the identity keys. By contrast, the delta messages that carry the user and administrative data are exchanged within a shared space, so they're encrypted using a symmetric key that's stored in the shared space and thus is available to the group, as we'll see later. (If you're confused already about the various keys, you can refer regularly to the final section of this chapter, "Taxonomy of Groove keys.")

Why each Groove identity requires two public and private keys

It's possible to combine the signature/verification and encryption/decryption functions into a single key pair. Beyond the obvious divide-and-conquer rationale, here are some subtler reasons why Groove doesn't combine these functions:

Plug-in cryptography

Groove's security subsystem is templatized (fully parameterized) and can work with virtually any public-key algorithms. Because some algorithms can only digitally sign while others can only encrypt, it's necessary to distinguish between the two purposes. Further, algorithms differ in the required length and properties of their keys.

External PKI policy

Groove plans (tentatively) to import key pairs from PKI sources. Such key pairs sometimes come with policy (as opposed to technical) constraints—for instance, signing might be allowed, but no encryption or key agreement. In order to comply with such policies, Groove must keep these key usages separate.

When members are joining or leaving a shared space, the identity's signing/verification key pair is used to authenticate the messages that invite (or uninvite) members. The encryption/decryption key pair is used to encrypt/decrypt the symmetric key, which is in turn used to encrypt/decrypt the invitation messages. We'll explore the invitation protocol later. But for now, let's complete the description of the security model when the shared space is in normal use.

Microsoft Exhibit 1019

When Bob types a line of text into his transceiver's chat window, the data flows in two directions—down to the disk, where it's written to the encrypted object store, and over the wire, in encrypted form, to the other members' transceivers and object stores. If Bob owns another Groove device, the data goes there too. The integrity of the message carrying Bob's chat line is protected by a Message Authentication Code (MAC).*

The encryption of each disk file is handled by a per-member, per-shared-space symmetric key. This key is in turn protected by a master symmetric key, stored in the account Bob created when he first installed Groove.† Why not just directly encrypt all on-disk shared spaces with the master key? The extra level of indirection isolates each shared-space file into its own security domain.

The encryption of Groove delta messages sent over the wire is handled by a symmetric key—which we'll call the group key—that's stored in the shared space and accessible to all members. In fact, there are two such keys—one MAC key, called L_G, for data integrity and authentication (the symmetric-key analogue of signing and verification), and one cipher key, called K_G, for encryption and decryption. Here, G denotes the set of group (shared space) members.

The default algorithm used for the MAC is HMAC-SHA1. SHA1 (FIPS 180–1) is used to produce a hash of the header and body of the message. HMAC (RFC 2104) provides authentication and integrity protection of the resulting hash.

The key to mutual trust

Our term "mutual trust" concerns how closely you can trace messages to senders. Upon receiving a message within a shared space, a member of the shared space can prove it was not tampered with by recomputing the MAC using the L_G key and comparing the resulting MAC with the transmitted MAC.

The message's sender is authenticated by the same means. Since the group key, L_G, was exchanged in an authenticated way (that is, via the invitation protocol or piggy-backed on a standard Groove delta message, as later described), only group members will have it. If the MACs match, the message must have been sent by a group member. But the exact member who sent the message cannot be verified,

* All Groove data that is encrypted is of course also integrity-protected. But to simplify the exposition, we sometimes abuse terminology and only explicitly mention the encryption aspect of data protection.

† The account itself is protected by a key derived from the passphrase chosen by the user when the account was created. Just as the master key is used to decrypt each shared-space-storage key, this passphrase-derived key is used to decrypt the account's storage key—and that's the only thing the passphrase is used for.

Microsoft Exhibit 1019

because L_G is common to all members of the shared space. That's why we call this a mutually-trusting shared space. Members are not prevented from spoofing each others' messages; they merely agree to trust one another not to do so.

To see why this mutual trust might not suffice, imagine a transcorporate shared space in which members engage in a high-stakes negotiation. It's not enough to know that a message came from an authenticated member of the group. It's crucial to know that a message came from an authenticated individual who's part of a particular negotiating team. In such cases, you'll need to form a mutually-suspicious shared space.

Why not always be suspicious? All things being equal, you'd rather bind message authenticity to individuals rather than just to the group. But all things aren't equal; it's more costly to authenticate individuals, as we'll see. And for many of the group activities that Groove can support, group-level authentication works just fine. When we were collaborating on this chapter, we cared about three things:

Confidentiality
: We didn't want anybody else reading our stuff.

Authenticity
: We didn't want anybody else impersonating a group member.

Integrity
: We didn't want any data corrupted in transit.

We got these assurances in a mutually-trusting shared space. And we got them just by installing and using Groove. When impersonation is a real risk, though, you'll want to create a mutually-suspicious shared space. Let's look at how that works.*

Mutually-suspicious shared spaces

It's time to introduce some more keys. Each member in each shared space has a Diffie-Hellman public/private key pair. These Diffie-Hellman keys (which are authenticated via the identity key pairs mentioned previously) are used to establish pairwise symmetric keys—that is, keys shared between each pair of members within a shared space. Through the magic of Diffie-Hellman, the pairwise keys aren't sent over the wire. Instead, they're independently computed by

* In the preview version of Groove available at the time this chapter was written, the ability to configure a shared space to run in trusting or suspicious mode was not exposed to the user. By default, shared spaces ran in trusting mode.

Microsoft Exhibit 1019

each pair of members. Bob computes a Bob/Carol pairwise key from his Diffie-Hellman private key and Carol's Diffie-Hellman public key. Carol computes the same pairwise key from her private key and Bob's public key.

There are two kinds of pairwise keys between members M_i and M_j. A cipher pairwise key, K_{ij}, encrypts the group keys (K_G, L_G) for distribution. A MAC pairwise key, L_{ij}, assures the data origin authenticity/integrity of messages in a suspicious shared space.

Recall that in the trusting case, a MAC is attached to each message. It's a MAC of the header and body of the message, protected in the group key: $\{X\}L_G$. Rather than a group-level MAC, suspicious mode uses a set of individual MACs denoted as $\{X\}L_{ij}$, one for each pair of members. Each of these uses HMAC-SHA1 to authenticate a message using the pairwise key shared between a pair of members. The resulting MACs are called authenticators (or multiauthenticators). These are symmetric-key analogues of public-key signatures.

Figure 18-2 shows what the authenticator looks like for a message in the trusting case. The size is the same for groups of any number. For each message, there's just a header, a body encrypted in the group key K_G, and an authenticator in the group key L_G. In a suspicious group, however, as shown in Figure 18-2, the multiauthenticator grows linearly with group number. For each message, there's a header, a body encrypted in the group key K_G, and one MAC per member in the pairwise keys L_{ij}.

Groove could have used public-key signatures instead of multiauthenticators. The advantage is that the size of such signatures and the time required to compute them remain constant. But public-key signatures are big, and they are very slow. For the small groups that are the focus of the initial release of Groove, multiauthenticators work best.

There are two different reasons to bundle authenticators into multiauthenticators: to support message fanout and to support the recovery of lost messages.

Message fanout

The Groove system supports pure peer-to-peer communication. In the simplest case, two Groove-equipped PCs connected by a null 10BaseT cable can communicate happily. But what if a member is firewalled, or offline, or connected by way of a very slow link? In these cases, the Groove software can use a relay server to enable or optimize peer communication.*

* Other companies may also offer relay services to support the Groove software.

Microsoft Exhibit 1019

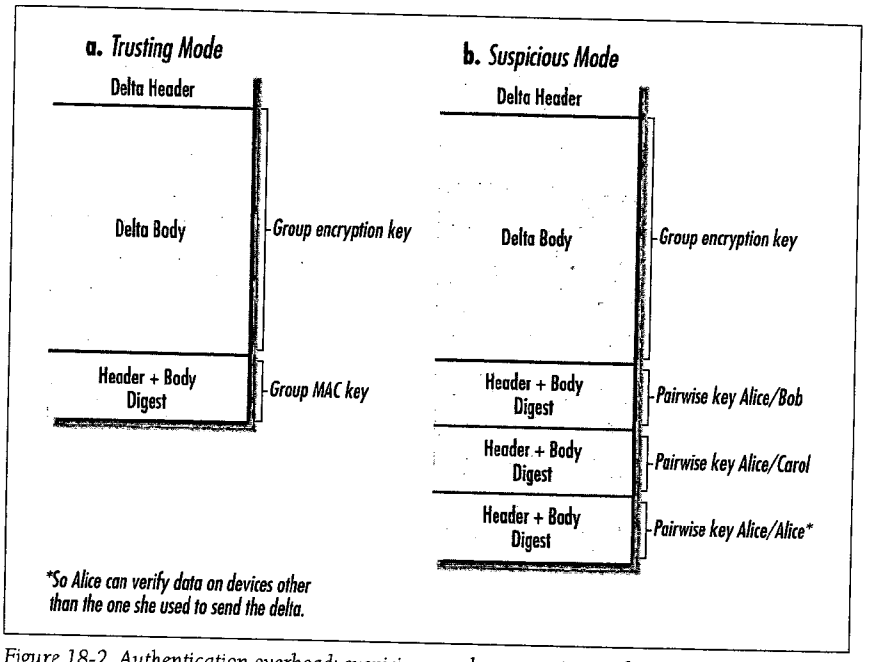

Figure 18-2. Authentication overhead: suspicious mode vs. trusting mode

A relay server is a system that understands Groove communication protocols and can route messages accordingly. The relay doesn't know anything about members, only about shared spaces and devices. Suppose Bob is on a modem link and sends a message to Carol and Alice. Rather than send two messages over that slow link, he'd like to send just one message to the relay and have the message fan out to everybody on its fast link. (The relay can also store and forward the message to someone who is offline.) The single multiauthenticator enables the sender to create a single delta message and push it to the relay server.

Consider the alternative: multiple deltas, each equipped with a single-authenticator. In that case, the sender would have to transmit *n* times the data through the pipe. Or the relay would have to interpret application layer data and then tag on the appropriate authenticator for the appropriate member, rather than just blindly relaying application data without parsing or interpreting it.

A complete description of Groove's communication protocols is beyond the scope of this chapter. Briefly, the Device Presence Protocol (DPP) solves the naming and awareness issues for devices. The Simple Symmetric Transport Protocol (SSTP) connects clients to clients, clients to relays, and relays to relays. It's SSTP that propagates information about a delta message, including its target endpoint.

Microsoft Exhibit 1019

For each target endpoint, there are three possible routes, listed here in order of preference:

1. Send directly (peer-to-peer) to the endpoint. This is impossible if target or sender is offline.
2. Send to the endpoint's preferred relay (currently hardcoded, but eventually user-customizable). This is impossible if either the relay or sender is offline.
3. Store it on the sender until it can be sent to the target or relay. This is always possible, as a last resort.

The preferred choice depends not only on who's online, but also on a heuristic calculation—by the sender—of the relative efficiency of the three choices. The sender considers line speed and message size to determine whether multicast (1) or fanout (2) would be better. The sender also needs to consider the firewall situation. Once the target's IP address is resolved through DPP, the source client will try to communicate directly to the target client. A direct peer-to-peer connection is always preferred over a relayed connection. However, sometimes it's necessary to use the relay as an intermediary even if both clients are online. Table 18-1 summarizes the possibilities.

Table 18-1. Relaying vs. direct connection

Source/client A	Target/client B	Connection
Public[a]	Public	Direct
Firewall/NAT/Proxy W/ 2492[b]	Public	Direct
Firewall/NAT/Proxy NO 2492[c]	Public	Via relay
Firewall/NAT/Proxy	Firewall/NAT/Proxy	Via relay
Public	Firewall/NAT/Proxy NO 2492	Via relay
Public	Firewall/NAT/Proxy W/ 2492	Initially via relay[d]

[a] Public: A device with a public IP address to which a direct connection is possible. This includes devices with DHCP-assigned public addresses.

[b] W/ 2492: A configuration where port 2492 (the network port used by Groove's proprietary SSTP protocol) is allowed inbound (if it appears in the "source" column) or outbound (if it appears in the "target" column).

[c] NO 2492: A configuration where Groove is forced (since other paths have failed) to use a path (such as an HTTP proxy) that blocks port 2492 inbound (if it appears in the "source" column) or outbound (if it appears in the "target" column). In this case, Groove must encapsulate all SSTP messages within HTTP messages and send them through port 80 (the HTTP port). This configuration also implies that the connection must go through the relay server since a Groove client cannot accept inbound connections on port 80 at this time. Groove does not want to conflict with other applications (such as a web server) that may be running on the client.

[d] The connection is initially established via the relay server. However, it is possible that the connection can transition into a direct connection if client B decides to change roles and become not only a listener from client A, but also a sender to client A. If so, client B creates a new connection to A that takes the properties of the second scenario (in the second row above), a direct connection. Since Groove favors direct connections over relayed connections, the original relayed connection from A to B terminates and is replaced by the direct connection from B to A.

Microsoft Exhibit 1019

Fetching lost messages

Suppose Alice never receives Bob's message. Because messages are sequenced, Alice's Groove software will discover that the message is missing and try to fetch it. Ideally, she'll fetch it from Bob. But what if Bob has, meanwhile, gone offline? In this case, Alice will try to fetch the lost message from Carol. Carol then encrypts the message in her Carol/Alice encryption key but attaches the Bob/Alice MAC (which Carol memorized earlier for this purpose) so Alice can correctly authenticate the message as one from Bob.

Shared space formation and trusted authentication

The person who sends an invitation to join a shared space is called the "chair"—in the sense of "chairperson of a meeting." If all of the members of a would-be shared space are already running Groove, the invitation protocol can begin with a Groove instant message from the chair to the invitees. Alternatively, this invitation can travel as email. The email alternative is especially important because it's a bootstrapping mechanism that brings non–Groove users into the Groove realm. In this case, the message's subject says "Please join our shared space." Its body describes the purpose of the shared space and offers a link to download and install the Groove software in case the recipient isn't already running it. Finally, the email message attaches the first in the sequence of Groove messages that comprise the Groove invitation protocol. Once the Groove software is installed, double-clicking that attachment "injects" the message into Groove and kicks off the invitation protocol.

Note that in this scenario, Groove uses email only as an unsecured carrier. This raises the specter of end-entity authentication: how do the chair and the invitee convince themselves that they're talking to the "right" person, i.e., that the invitation protocol hasn't been hijacked by an imposter? It's possible, though unlikely, that the sender and recipient will use S/MIME or PGP to authenticate and/or encrypt the invitation message.

In the case of S/MIME, trust resides in a PKI-based system. If an S/MIME signature were attached to the chair's invitation message, the invitee could examine the certificate bound to that message. The certificate would in turn be signed by a certification authority (CA). The CA assures the invitee that the chair's certificate—on the invitation—binds to a specified real-world identity.

In the case of PGP, trust resides in a more decentralized web of trust system. Rather than depending on a PKI hierarchy, PGP models trust in a more collegial

Microsoft Exhibit 1019

way. Certificates are signed, not by specialized CAs, but by other people. PGP users sometimes expand their webs of trust by holding "signing parties" where people can meet face-to-face and cross-certify their keyrings.

It's plainly evident that neither the hierarchical nor the web of trust approach has taken the world by storm. For most people, PGP and S/MIME implementations are far too complex and hard to use. The result is that the percentage of signed and/or encrypted email on today's Internet is vanishingly small.

Groove aligns itself more closely with PGP than with S/MIME. And it's influenced by recent initiatives to simplify PKI: Simple Public Key Infrastructure (SPKI, RFC 2692), and Simple Distributed Security Infrastructure (SDSI, *http://theory.lcs.mit.edu/~rivest/sdsi1.html*). These initiatives, now merged, aim to make direct user-to-user cross-certification easy enough so that users can actually do it for themselves, in a natural way. Central to this approach is SDSI's notion that identities need not be represented by globally unique names, nor characterized by a fixed set of attributes. SDSI stresses that it is human judgment that must decide, given a set of attributes, whether to accept a claimed identity as valid. To that end, it should be easy to create and read certificates containing attributes that are meaningful to people: Phone numbers, photos, and free-form text. Groove, though not an implementation of SDSI, subscribes to these ideas.

In Groove, the public persona of an identity is called a contact. It contains a self-chosen free-form name,* and the public halves of the identity's two self-chosen key pairs.† In an enterprise version of Groove, it might also contain a student ID or an X.500 name (such as O="Big Corporation Ltd"/OU="Finance"/CN="John Doe").

When Bob creates his identity, he may decide to add more information to the contact—for example, his phone number. Here's how Alice can use that information to authenticate Bob's invitation message. The message is signed by Bob. It comes with the public key that is purportedly Bob's.‡ Alice's Groove software first verifies the signature using the transported public key (thereby guaranteeing that the message was really signed by the private key corresponding to the

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* The self-asserted name will typically be the user's real-world name, but it could also be an email name, or indeed any display name or alias the user chooses.

† The key pairs are currently "self-chosen" in the sense that Groove automatically generates them (locally, on the user's machine) and assigns them to the identity when it is created (there is a different set of generated key pairs for each identity). There is no central "identity authority." In the future, Groove will almost certainly allow users to import their key pairs from external sources, such as PKI certificates.

‡ To simplify the narrative, we sometimes pretend the two Groove key pairs are just one key pair, used for signature/verification and encryption/decryption/key establishment.

Microsoft Exhibit 1019

transported public key) and then locally computes the fingerprint (hash of the public key, a string of hex digits). She can then phone Bob and ask Bob to report his genuine fingerprint to her. If the fingerprints match, the message could only have come from (i.e., been signed by) Bob—assuming that the private key for Bob's Groove identity, protected by Bob's passphrase, remains uncompromised under Bob's control.

Alice knows Bob, along with his phone number and his voice, through trusted, out-of-band, real-world channels. For example, Bob might have printed his fingerprint on his business card and given the card to Alice in a face-to-face meeting. Later, recalling that meeting, Alice matches the fingerprint on the business card to the fingerprint computed from a message claiming to come from Bob. Authentication doesn't depend on third-party certification of Bob's public key. Groove depends on relationships that are rooted in real-world collaboration, and it extends those relationships into the realm of shared spaces.

Admittedly, this scheme places the burden of establishing trust on the user. But the truth is that the burden always rests with the user. Technologies such as S/MIME and SSL purport to take matters out of users' hands. But they, too, ultimately rely on fingerprints that people should in theory check before investing trust, but in practice almost never do. From the user's perspective, what sets Groove apart from other systems is its all-crypto, all-the-time approach. You never have to worry about whether messages are confidential, because they always are. Messages are likewise always authentic—to precisely the degree that you care about authenticity. In a shared space that supports casual discussion, you may not care about the risk of impersonation. What's more, some spaces may be explicitly pseudonymous, with no use of real-world identities. In these cases, you may not bother to check fingerprints. In a shared space that supports highly confidential business activities, however, you should worry about impersonation, and you should check fingerprints.

Note that while Groove does not implement or require PKI, neither is it incompatible with PKI. Recall that Groove is flexible about the kinds of information stored in an identity's contact. If an enterprise has assigned X.500 names to employees, these names can be included in Groove identities. In this way Groove can ride on preexisting directory and naming structures. Further, its authentication protocols can be extended to handle PKI-style certificate validation. When an enterprise runs its own CA, for example, an enterprise version of Groove might be configured to trust that CA, just as browsers today accept SSL and S/MIME certificates signed by VeriSign, Entrust, and others. Look for PKI integration in post-preview editions of Groove.

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Inviting people into shared spaces

When Bob invites Alice into a shared space that already includes Bob and Carol, the following things will have to happen:

- Bob and Alice must complete the invitation protocol.
- New pairwise keys (Alice/Bob, Alice/Carol) will need to be established.
- The group key will need to be given to Alice.

Note that the first step of the invitation may not be confidential. Alice isn't a member yet, and the key that will be used by the new group doesn't exist. What's more, Alice may not even be a Groove user yet, in which case she has no Groove account, identity, or public key that Bob can use to encrypt the first invitation message. In this case, the invitation is not confidential—unless Bob and Alice have established confidentiality in another way, for example, using S/MIME to encrypt an email version of the invitation message. But the invitation is always digitally signed by the Groove software with Bob's signature private key, regardless of the channel used to send it.

Alternatively, Bob and Alice may already be Groove users who have communicated before. In this case, the invitation—if sent as a Groove instant message—is confidential.

Either way, the invitation includes all the information needed to bring the new member into the shared space:

- A cryptographic context, which defines the algorithms to be used in that shared space. It includes, among other things, the Diffie-Hellman parameters used within the shared space. Alice's Groove software will use this information to generate her Diffie-Hellman key pair. Once Bob's Groove software has distributed Alice's Diffie-Hellman public key to everyone, she will be able to use the key to compute pairwise keys with other members.
- The public keys for Bob's identity.

On receiving the message, Alice can verify that it's intended for her (by checking the name on the invitation), authenticate Bob's contact (by calling the phone number listed there and checking the fingerprint), and consider what kind of shared space she's been asked to join. In practice, when the two parties have previously communicated in Groove, the authenticity of the message is not in question. It's only a question of whether to join for the stated purpose (for instance, to collaborate on a project), given the stated mode (for instance, trusting vs. suspicious).

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To accept the invitation, Alice will have to install Groove if she hasn't already done so. If Alice then agrees to join the group, her Groove software sends Bob's software a message containing the following information:

- A one-time key encrypted in Bob's public key.
- Alice's Diffie-Hellman public key for the shared space, signed by the private half of the identity key in her account so that each member can verify its authenticity.

On receiving Alice's acceptance, Bob's Groove software decrypts the one-time key, decrypts Alice's reply with the one-time key, and verifies the validity of Alice's acceptance. Bob himself, after a fingerprint authentication of Alice, can then manually confirm (by clicking an OK button) that Alice should be added to the group. That confirmation triggers the following events:

- A New-Member-Added message is sent to the group.
- Alice receives a copy of the shared-space data, which includes the group keys, K_G and L_G. The shared-space data is encrypted in a one-time key which is, in turn, encrypted in Alice's public key.

The New-Member-Added delta message

The message that tells everybody a new member has joined is called, appropriately, the New-Member-Added delta. This message, sent from Bob to the preexisting members (in this case, only Carol), announces that Alice has joined the group. It includes Alice's Diffie-Hellman public key, which Carol stores in her member list and uses to establish her pairwise key with Alice.

Arguably, the New-Member-Added message should also trigger the establishment of a new group key. It doesn't. Only when a member is uninvited is a new group key established. When Alice is invited, she gets the same group key that Bob and Carol were already using. Therefore, she can read all previous messages stored in the shared space. The policy might instead be that Alice is not necessarily privy to the prior activities of the group. Resetting the group key on invitation would prevent her from reading messages (had she recorded them) that were exchanged among preexisting members before she joined. An early implementation of Groove in fact worked just this way. But this model conflicted with the way most people want to use Groove. When you join a shared space, you don't typically want to receive only future messages exchanged within it. You want to see the whole history of the shared space. Access to that prior transcript is, in many cases, the prime motivation to join. So when Alice joins the group, her shared space synchronizes fully with the group's.

Microsoft Exhibit 1019

Groove's security subsystem retains latent support for reestablishing the group key on invitation, and this feature may be reenabled, at least as an option, in post-preview editions of Groove. Of course, users can themselves achieve a similar effect. If Bob and Carol had been collaborating in a shared space and then wanted to invite Alice without giving away the contents of that shared space, they could simply create a new shared space and invite Alice into it.

If Carol uninvites someone, her Groove software sends a rekey delta message, which includes the changed group membership information plus "piggy-backed" rekey information. A rekey delta can include any combination of group rekeys. In the case of a simple member removal, it's used to establish a new group key. The new group key is transmitted by separately encrypting a copy of it, per member, in the pairwise key (K_{ij}) shared with that member. This applies in both modes: trusting and suspicious.

Groove adds rekey information onto the delta message for two reasons:

- So security metadata (including the rekey information and MACs) can be transparent to the higher-level application. It can MAC the delta without knowing that rekey information was added to it. On the receiving end, the security layer strips off the rekeys before handing the delta back to the application layer.
- So the new key sent in the piggy-back rekey information can be immediately used to encrypt the delta that it is being piggy-backed onto. If the rekey information were embedded inside the payload of the delta, then on the receiving side, the delta would first need to be decrypted in a previous key. Because the rekey information is piggy-backed outside the encrypted delta payload, that delta itself can be encrypted in the new key.

The uninvitation protocol cannot, as yet, be controlled administratively using credentials and permissions. Currently, any member can kick another member out of a shared space. An authorization architecture, for controlling access to this and other actions, is planned for a near-future release. In the meantime, tool writers can use the authentication machinery to implement tool-specific authorization rules, though we recommend waiting (if you can) for explicit authorization support in Groove.

Key versioning and key dependencies

Groove messages may arrive in any order. In particular, Carol might receive a message from Alice encrypted in the new group key, K_2, before she receives the

Microsoft Exhibit 1019

message from Bob that bears that new key. This can happen, for example, if her link to Alice is faster than her link to Bob.

The solution is for Alice's message to state its dependence on a version of the group key. The message says, in effect, "I depend on a key-bearing message from Bob," and it includes the sequence number of that message. If Carol's Groove software hasn't yet seen the message bearing K_2, it will defer handling of Alice's K_2-encrypted message until the key arrives. Fortunately for Groove's security system, message sequencing is a fundamental property of the underlying communication layer.

In fact, things are a bit more complex. It's not just that Carol is waiting for a message from Bob in order to be able to decrypt a message from Alice. More accurately, Carol at her desktop PC is waiting for a message from Bob at his notebook PC in order to decrypt a message from Alice at her desktop PC. Sequence numbers are, in short, device-specific. Bob himself may have sent K_2 from his notebook PC but not yet received it at his desktop PC at work. Like all Groove messages, a rekey message has to propagate to all endpoints.

At some point, Carol's Groove software will want to delete the old group key, K_1. Can it do this once it knows that everyone has received K_2? No. It must also know that every message depending on K_1 has been received by all user/device pairs. To disseminate this knowledge, all Groove endpoints periodically broadcast their current state to the shared space members, and describe their latest dependency sequences.

The old and new keys, K_1 and K_2, represent major versions of the group key. There's also a need to further differentiate keys using minor versions. Here's why. Consider two companies, C1 and C2, participating in a shared space by way of their respective Internet connections, using group key K_3. Figure 18-3 shows the configuration of members in the shared space..

Now C1's T1 line goes down temporarily, isolating the C1 members from the C2 members. During this period of separation, M1 (in C2) uninvites M2, and M4 (in C1) uninvites M5. Figure 18-4 shows the result.

Each uninvitation independently produces a new group key, K_4. Now the T1 line comes back up. How does the system distinguish between the two K_4s? To handle this case, the major version of a key is qualified by a minor version, which is actually the sequence number of the message that transmitted the key.

Microsoft Exhibit 1019

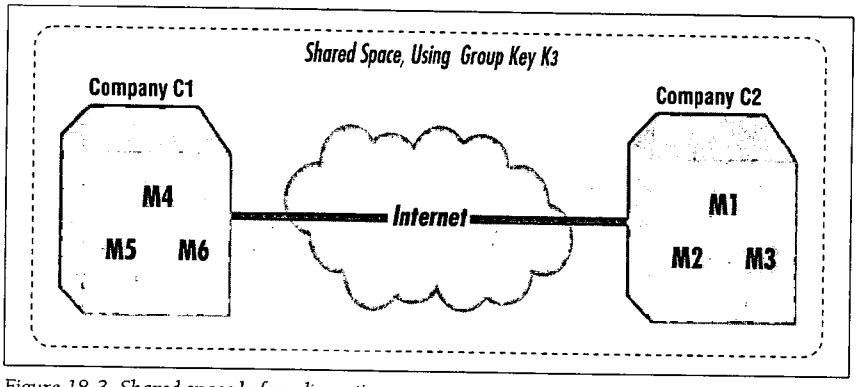

Figure 18-3. Shared space before disruption

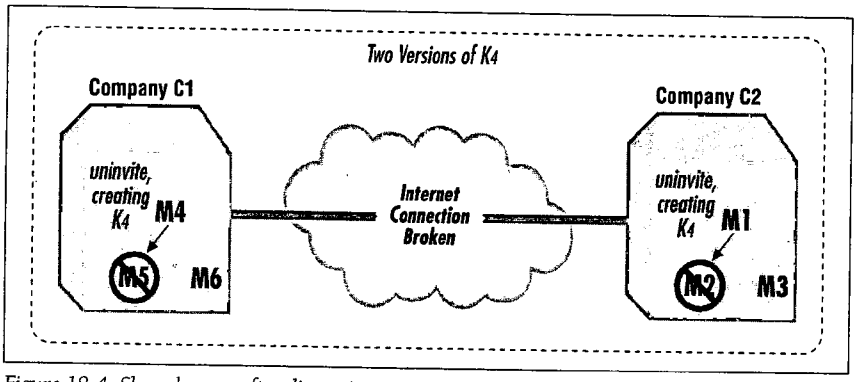

Figure 18-4. Shared space after disruption

Central control and local autonomy

Assuring that Groove would work in a fully decentralized, peer-to-peer mode was a major challenge. "It wasn't just for amusement that we undertook to do this," says Ray Ozzie, founder and CEO of Groove Networks, and before that the creator of Lotus Notes. He notes that the current trend toward hosting critical business software at ASP (application service provider) sites introduces worrisome points of failure. Groove de-emphasizes (but does not abolish) centralization. If a relay service provided by Groove Networks (or another provider) fails, Groove users can in general continue to communicate, and even execute such complex protocols as concurrent uninvitations involving disconnected subgroups.

At the same time, there are crucial aspects of security that should be at least logically centralized. For enterprise IT, a hybrid approach is better than a pure peer model. You want to centrally determine a policy for your network of peers and

Microsoft Exhibit 1019

then distribute that policy to the individual desktops. Such a policy might control the following:

- The required length, complexity, and change frequency of the account passphrase
- Whether a user's PC can memorize the account passphrase
- Whether users are required to authenticate one another, and if so, how
- Whether everyone, or only designated shared-space administrators, can invite and uninvite members
- Who, outside of the enterprise, can join a group, and on what terms

The Groove product was designed to be as easy to use as a PalmPilot, because Ozzie's prior experience with Notes showed that the vast majority of security leaks were caused by human (that is, user and administrative) error. The goal was therefore to create a product that delivers high-grade security by default, requiring as little of the user (or administrator) as possible. Users, in particular, see no security-relevant configuration choices, and as a result their secure use of the system is as failsafe as possible. Administrators, likewise, will be constrained to a minimal set of security choices.

One such choice deserves special note. A user can ask the Groove software to memorize the account passphrase, so that it need not be retyped once per session. If the user must first authenticate to the operating system (Windows NT, 2000), this won't be a problem so long as there is a passphrase. (It's possible, but inadvisable, to create an account with no passphrase at all. In that case, nothing is securely protected on disk.) But if the user doesn't authenticate to the operating system (as Windows 95 and 98 allow), the machine boots up automatically into Groove and only physical security governs the Groove data. Because the goal is for the Groove software to become as ubiquitous as the browser, it was decided to allow this feature.* Clearly this is the kind of policy that you might want to reverse in an enterprise deployment of Groove, in order to ensure that users do the right thing.

There's another important sense in which Groove's central services help people do the right thing with respect to security—or, rather, discourage them from doing the wrong thing. The Groove relay server will proxy connections between

* There's more to the story. Memorized passphrases are, locally, the moral equivalent of null passphrases. But remotely they are superior. That's true because Groove accounts are mobile. A user can transfer an account from one device to an Internet-based service, and thence to another device. Memorized passphrases encourage long/strong passphrases, so are accounts stored on the Internet strongly protected.

Microsoft Exhibit 1019

peers when one or both are behind firewalls or NATs. Although Groove prefers direct connections between endpoints, it will always work out of the box for users who cannot establish such connections. That means people need not try to make complex, and thus error-prone, modifications to their perimeter security.

Practical security for real-world collaboration

At the end of the day, all software is rife with vulnerabilities. No one pretends that Groove is immune to this law. Although Groove encrypts to hard disk, it can't encrypt the virtual memory pages swapped out by the operating system. It can't save you from rogue software components that you trustingly install and use or Trojan horse programs that install themselves without your knowledge. It can't hide the data on your screen from a microcamera hovering over your shoulder or from a Van Eck device down the street.

Security is never wholly attainable, for a long and depressing list of reasons. But you should still take every reasonable precaution. Today, few people bother. And in truth, the popular decision to shun existing ways to secure our routine collaboration is not an irrational one. It's hard to take reasonable precautions. Doing so interferes with the spontaneity we require. The procedures are complex and therefore error-prone. Even when we try to secure our communications, we often fail to do so.

Groove aims to make reasonable precautions automatic and failsafe. It envisions a world in which peer-enabled groupware is as easy and spontaneous as email, yet as secure as anything transpiring on the Net today.

Taxonomy of Groove keys

The cryptographic keys used in Groove are described in the following list:

One passphrase per account
: Not shared with anybody else. Stored in the user's brain (unless the user requests Groove to memorize it on a machine, which is convenient but discouraged). The Unicode passphrase can have any length and take any form. It can be changed wherever you want, but if you have multiple copies of an account on multiple machines, you have to change the passphrase yourself on each machine.

Microsoft Exhibit 1019

One symmetric key per account

This key protects the account itself (actually, the storage key for the account file, described later in this list). By default, it is a MARC4 key generated from the user's passphrase using the PBKDF2 algorithm, with a salt and an iteration count. In this case, MARC4 uses 256-byte keys, rather than the Groove default 24-byte keys. The length of the salt is 20 bytes, and the iteration count is currently set at 256 (i.e., approximately one-tenth of a second on current typical CPUs). These unusual precautions ensure that the passphrase itself, not the cryptography surrounding it, is the weakest link—as it should be.

One asymmetric key pair per identity for signature/verification

The default algorithm is ElGamal. This key pair is used for several signature/verification purposes: to authenticate invitations, instant messages, and Diffie-Hellman public keys. It's stored in the account and cannot be changed (except by creating a new identity). The public half of the key pair is also stored in the identity's contact, which is shared with other users.

*Another asymmetric key pair per identity for encrypting/decrypting symmetric keys**

The default algorithm is ElGamal. This key pair is used to encrypt/decrypt the symmetric (by default, MARC4) keys that in turn protect the invitation protocol and instant messages. It's also stored in the account and cannot be changed (except by creating a new identity). The public half of the key pair is also stored in the identity's contact, which is shared with other users.

One digital fingerprint per identity

This is a hash of the public halves of the above two identity key pairs, as verifiably calculated by Groove on a per-use basis. It enables Groove users to authenticate one another.

One Diffie-Hellman key pair per member, per shared space

It's deterministically generated from the private halves of the previous two identity key pairs and the GUID (24-byte random globally unique ID) of the shared space. This key is constant for the duration of the member's participation in the shared space and is used to establish pairwise keys with other members of that shared space.

* As noted previously, this key pair and the preceding one are sometimes conflated in informal discussion. When we say "the key pair of the identity," we mean both the signature and the encryption key pairs.

Microsoft Exhibit 1019

One pairwise key (K_{ij}), per pair of members, per shared space, used for key distribution

The algorithm for this symmetric key is, by default, MARC4. The key is computed for each pair of members from the Diffie-Hellman key pairs, by means of the classical (authenticated) Diffie-Hellman key agreement algorithm. It's cached in each member's copy of the shared space and used to distribute the group key when it changes.

Another pairwise key (L_{ij}), per pair of members, per shared space, for message authentication

The associated algorithm is, by default, HMAC-SHA1. The key is again computed for each pair of members from the Diffie-Hellman key pairs, and it's used to assure message authenticity and integrity in mutually-suspicious mode.

One group key per shared space (K_G), for confidentiality

This key (by default, MARC4) is used for shared space confidentiality. It's stored in the shared space and is reestablished whenever a member leaves the shared space.

Another group key per shared space (L_G), for message integrity

This key (by default, HMAC-SHA1) is used for shared-space integrity in trusting mode. It's stored in the shared space, and is reestablished whenever a member leaves the shared space.

Master key

One per account, stored in the account (and hence protected by the passphrase). This key (MARC4 by default) is used to protect storage keys (next item).

Storage keys

One per shared-space file, including the account file. These keys encrypt/decrypt on-disk data. The storage keys for (non-account) shared spaces are protected by the master key and stored with the shared-space files. The storage key for the account is protected by the account's passphrase and stored with the account file.

Microsoft Exhibit 1019

chapter nineteen

Interoperability Through Gateways

Brandon Wiley, Freenet

In my travails as a Freenet developer, I often hear a vision of a file-sharing Utopia. They say, "Let's combine all of the best features of Freenet, Gnutella, Free Haven, Mojo Nation, Publius, Jabber, Blocks, Jungle Monkey, IRC, FTP, HTTP, and POP3. We can use XML and create an ÜberNetwork which will do everything. Then we can IPO and rule the world."

When I hear this vision, I shake my head sadly and walk slowly away. I have a different vision for solving the world's file-sharing problems. I envision a heterogeneous mish-mash of existing peer-to-peer applications creating one network with a thousand faces—what you might call an OmniNetwork.

Why unification?

Every network has its flaws. As a Freenet developer, I never miss an opportunity to give Gnutella's scalability and anonymity a good-natured ribbing. At the same time, Freenet is constantly criticized because (unlike with Gnutella) you have to donate your personal hard drive space to a bunch of strangers that may very well use it to host content that you disapprove of.

The obvious solution is to use only the network that best suits your needs. If you want anonymity, Freenet is a good choice. If you also want to be absolutely sure that you are not assisting the forces of evil (can you ever really be absolutely sure?) use Gnutella.

Ah, but what if you want Freenet's "smart routing" and yet you also want Gnutella's fast integration of new nodes into the network?

The answer is obvious: build an ÜberNetwork with anonymity and smart routing and fast node integration *and* a micropayment system *and* artist compensation *and*

Microsoft Exhibit 1019

scalability to a large number of nodes *and* anti-spam safeguards *and* instant messaging capability, etc. It is ideas such as this that make me want to cast off the life of a peer-to-peer network developer in exchange for the gentle ways of a Shao-lin monk.

Why not an ÜberNetwork?

The problem with an ÜberNetwork is simple: it's impossible. The differences in file-sharing networks are not merely which combinations of features are included in each particular one. While many file-sharing networks differ only in choice of features, there are also distinct and mutually exclusive categories of systems. Several optimization decisions are made during the design of a network that cause it to fall into one of these categories. You can't optimize for everything simultaneously. An ÜberNetwork can't exist because there are always trade-offs.

Why not an ÜberClient?

The idea of an ÜberClient is similar to that of an ÜberNetwork: To create a single application that does everything. An example of such an application in the client/server world is the ubiquitous web browser. These days, web browsers can be used for much more than just browsing the Web. They are integrated web, news, email, and FTP clients. The majority of your client/server needs can be serviced by a single application. Unlike the ÜberNetwork, the ÜberClient need not force everyone to convert to a new system. An ÜberClient would be compatible with all of the current systems, allowing you to pick which networks you wanted to retrieve information from.

The problem with the ÜberClient is that it is a client, and clients belong in the client/server world, not the world of peer-to-peer. Furthermore, the ÜberClient that already exists—the web browser—can serve as a kind of gateway to peer-to-peer applications. Many file-sharing networks either act as miniature web servers or are developing browser plugins. Someday you will probably be able to access all of the file-sharing networks from your web browser.

However, there is a catch: you will have to be running a node on each file-sharing network that you want to access. To do otherwise would not be peer-to-peer, but client/server. Also, the advantages of files crossing over between networks are lost. Files on Free Haven will still take a long time to load and unpopular files on Freenet will still disappear.

Microsoft Exhibit 1019

Why not just use XML?

The next most popular solution after the creation of an ÜberClient is to link all of the existing networks together using an interoperable protocol, such as something based on XML, like XML-RPC or SOAP. The problem with this approach is that is doesn't solve the right problem. The beauty of XML is that it's a single syntax that can be used for many different purposes. It's a step in the right direction for cross-platform, language-independent object serialization and a universal syntax for configuration files. However, the problem of interoperability between file-sharing networks is not the lack of a shared syntax. The syntax parsers for all existing file-sharing networks are minor components of the code base. A message parser for any existing system could be written in a weekend if a clear specification was available.

The problem of interoperability is one of semantics. The protocols are not interoperable because they carry different information. You wouldn't expect Eliza to be a very good chess player or Deep Blue to be a good conversationalist even if they both used an XML-based protocol for communicating with you. Similarly, you should not expect a free system such as Gnutella to understand a micropayment transaction or an anonymous system such as Freenet to understand user trust ratings.

One network with a thousand faces

The solution to the problem, then, is not an ÜberNetwork, but the integration of all the different types of networks into a single, interoperable OmniNetwork. This has some advantages over the current state of many non-interoperable networks. Each person could use his or her network of choice and still get content from all of the other networks. This means that everyone gets to choose in what way they want to participate, but the data itself reflects the cumulative benefits of all systems.

To clarify this statement, I will describe how a gateway might work between Freenet and Free Haven. What are the advantages of a gateway? They pertain to the relative strengths and weaknesses of the systems. Data on Freenet can be retrieved quickly, whereas speed is recognized as a problem on Free Haven. However, data can disappear at unpredictable times on Freenet, whereas the person who publishes data on Free Haven specifies when it expires. Combine the two systems and you have readily available, potentially permanent data.

Suppose a user can insert information into either Free Haven or Freenet, depending on her preference. Then a second user can request the same information

Microsoft Exhibit 1019

from either Free Haven or Freenet, depending on his preference. If the users are on the same network, the normal protocols are used for that network. What we're interested in here are the two possibilities left: either the information is on Free Haven and the requester is on Freenet, or the information is on Freenet and the requester is on Free Haven. In either case, the information should still be retrievable:

When Free Haven data is requested through Freenet

Requesting data through Freenet guarantees the anonymity of the requester even if he distrusts the Free Haven node. Additionally, every request of the file through Freenet causes the information to migrate to Freenet, lending the caching ability of Freenet to future requests. While the first request has to go all the way to Free Haven to fetch the information, subsequent requests need only traverse Freenet and will therefore be faster. If the information expires from Freenet, a copy still exists in Free Haven.

When Freenet data is requested through Free Haven

In this case, the information is retrieved from Freenet and cached in Free Haven. Since the information was fetched from a Freenet node, the anonymity of the requester is guaranteed even if he mistrusts the Freenet node. Additionally, requesting the data from Free Haven will cause it to be cached in Free Haven, so a copy with a guaranteed lifetime will now exist. If it should expire from Freenet, a copy still exists in Free Haven.

This is just one example of the ways that the synergy of systems with opposing designs can create a richer whole. Each of the major types of file-sharing systems adds its own benefits to the network and has its own deficiencies that are compensated for. We'll look at some details in the next section.

Well-known networks and their roles

In this section I'll list the characteristics that distinguish each of five popular networks—Freenet, Gnutella, Mojo Nation, Free Haven, and Publius—so we can evaluate the strengths each would offer to an all-encompassing OmniNetwork.

While the world of peer-to-peer is already large at quite a young age, I've chosen to focus here just on file storage and distribution systems. That's because they already have related goals, so comparisons are easy. There are also several such systems that have matured far enough to be good subjects for examination.

Freenet

Freenet adds several things to the OmniNetwork. Its niche is in the efficient and anonymous distribution of files. It is designed to find a file in the minimum

Microsoft Exhibit 1019

number of node-to-node transactions. Additionally, it is designed to protect the privacy of the publisher of the information, the requester of the information, and all intervening nodes through which the information passes.

However, because of these design goals, Freenet is deficient in some other aspects. Since it is designed for file distribution and not fixed storage, it has no way to ensure the availability of a file. If the file is requested, it will stay in the network. If it is not requested, it will be eliminated to make room for other files. Freenet, then, is not an ideal place to store your important data for the rest of eternity.

Second, Freenet does not yet have a search system, because designing a search system which is sufficiently efficient and anonymous is very difficult. That particular part of the system just hasn't been implemented yet.

A final problem with Freenet is that in order to assure that the node operators cannot be held accountable for what is passing through their nodes, the system makes it very difficult for a node operator to determine what is being stored on his hard drive. For some this is fine, but some people want to know exactly what is being stored on their computers at all times.

Gnutella

Gnutella offers an interesting counterpoint to Freenet. It is also designed for file distribution. However, each node holds only what the node operator desires it to hold. Everything being served by a Gnutella node was either put there by the node operator or else has been requested from the network by the node operator. The node operator has complete control over what she serves to the network.

Additionally, this provides for a form of permanent storage. The Gnutella request propagation model allows that if a single node wants to act as a permanent storage facility for some data, it need do nothing more than keep the files it is serving. Requests with a high enough time-to-live (TTL) will eventually search the entire network, finding the information that they are looking for. Also, Gnutella provides searching and updating of files.

However, the Gnutella design, too, has some deficiencies. For instance, it does not provide support for any sort of verification of information to avoid tampering, spamming, squatting, or general maliciousness from evil nodes and users. It also does not have optimized routing or caching to correct load imbalances. In short, it does not scale as well as Freenet. Nor does it provide much anonymity or deniability for publishers, requesters, or node operators. By linking Freenet and Gnutella, those who wish to remain anonymous and those who wish to retain control over their computers can share information.

Microsoft Exhibit 1019

Mojo Nation

What Mojo Nation adds to the peer-to-peer file-sharing world is a micropayment system, and a rich and complex one at that. A micropayment system adds the following advantages to the OmniNetwork: Reciprocity of contribution of resources, compensation for the producer of content, and monetary commerce.

Reciprocity of contribution simply means that somebody has to give something in order to get something. Both Freenet and Gnutella must deal with a lack of reciprocity, an instance of the archetypal problem called the tragedy of the commons. Actually, this has been a problem for file-sharing systems throughout the ages, including BBSs, anonymous FTP sites, and Hotline servers. Now, in the peer-to-peer age, there is no centralized administrator to kick out the leeches. In an anonymous system, it's impossible even to tell who the leeches are (unless the providers of content want to voluntarily give up their anonymity, which they generally don't).

Micropayments solve the reciprocity of contribution problem by enforcing a general karmic balance. You might not give me a file for every file you get from me (after all, I might not want your files), but all in all you will have to upload a byte to someone for every byte you download. Otherwise, you will run out of electronic currency. This is indeed a boon for those who fear the network will be overrun by leeches and collapse under its own weight.*

Solving reciprocity is particularly important for controlling spam and denial of service attacks. For every piece of junk someone asks you to serve to the network, you receive some currency. If you are flooded with requests from a single host, you receive currency for each request. The attacker may be able to monopolize all of your time, effectively rendering your node inoperable to the rest of the network, but he will have to pay a high price in order to do so. With micropayments, you are, in effect, being paid to be attacked. Also, the attackers must have some way of generating the currency for the attack, which limits the attackers to those with enough motivation and resources.

There are other uses for micropayments besides reciprocity, particularly the ability to engage in actual commerce through a file-sharing network. If you want to trade other people's content for them in order to gain some currency, the handy tip button, a feature of the Mojo Nation interface, allows people to send some currency to the producers of content as well as the servers.

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* See the Jargon File "Imminent Death of the Net Predicted!", *http://www.tuxedo.org/~esr/jargon/jargon.html*.

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Also, the system could someday perhaps be used to exchange electronic currency for not just information, but things like food and rent. I can already see the kids dreaming of supporting themselves through savvy day trading of the latest underground indie tunes (the artists being supported by the tip button). Hipness can metamorphose from something that gets you ops on an IRC channel to a way to make mad cash.

However, not everyone wants to exchange currency for information. Even exchanges are certainly one mode of interaction, but it is very different from the Freenet/Gnutella philosophy of sharing information with everyone. Freenet and Gnutella serve a useful role in the Mojo Nation framework when a single node does not have the resources to make a transaction. If you don't have any resources (you have a slow machine, slow connection, and small hard drive) it is hard to get currency, since you get currency by contributing resources. Without currency you can't request anything. However, if a lot of low-resource nodes decided to get together and act as a single, pooled node, they would have significant resources. This is exactly how Freenet and Gnutella work. One node is the same as a whole network as far as your node is concerned. Thus, low-resource Mojo Nation nodes can form "syndicates" so that they will not be excluded from having a presence on the Mojo Nation network.

By combining the two types of networks, the free and communal networks of Freenet and Gnutella with the commercial network of Mojo Nation, people can choose whether to share freely or charge for resources as they see fit. Different people will choose differently on the matter, but they can still share content with each other.

Free Haven and Publius

Free Haven and Publius are in an entirely different category from other file-sharing networks. While the other networks concentrate on the distribution of content people want (reader-centric systems), these systems concentrate on anonymously preserving information (publisher-centric systems). The flaw people point out most often in Freenet is that data disappears if no one requests it for a long enough time. Luckily, Free Haven and Publius are optimized to provide for just that eventuality. They are conceptually derived from the mythical "Eternity Service" in which once you add a file it will be there forever. While it may be possible to delete a file from a Free Haven or Publius node, these networks are specifically designed to be resistant to the removal of content.

File storage networks have problems when viewed as file distribution networks. They are generally much slower to retrieve content from (because they are optimized for storage and not distribution). Additionally, they do not deal

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well with nodes fluttering on and off the network rapidly. To lose a node in a file storage network is problematic. To lose a node in a good file distribution network is unnoticeable.

There are great possibilities with the combination of reader-centric distribution networks with publisher-centric storage networks. It would be ideal to know that your information will always be available to everyone using a file-sharing network anywhere. People can choose to share, trade, buy, and sell your information, anonymously or non-anonymously, with all the benefits of distributed caching and a locationless namespace, and with no maintenance or popularity required to survive. Once the information is inserted in the network, it will live on without the publisher needing to provide a server to store it. Unfortunately, making gateways between networks actually work is somewhat problematic.

Problems creating gateways

The problem with creating gateways is finding a path. Each piece of information is inserted into a single network. From there it must either find its way into every connected network, or else a request originating in another network must find its way to the information. Both of these are very difficult. In short, the problem is that an insert or request must find its way to a node that serves as a gateway to the separate network where the information is stored.

Problems with inserts

The problem with finding a path to another network during an insert is that the paths of inserts are generally very short and directed. Each network routes its inserts using a different method:

- Freenet takes the "best" path to the "epicenter" for a given key. The length of the path is specified by the user. A longer path means that there is a greater chance for an insert to happen upon a gateway. However, longer paths also mean that you have to wait longer for the insert to complete.
- Gnutella doesn't have inserts.
- Mojo Nation splits a file into multiple parts and inserts each part into a node. The nodes are chosen by comparing the file part's hash to the range of hash values that a node advertises as serving.
- Free Haven splits up the file using *k*-of-*n* file splitting and inserts each part to a node. The nodes are chosen by asking trusted nodes if they want to trade their own data for that particular file part.
- Publius sends a file to a static lists of nodes and gives each node part of the key.

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Some of these techniques could be extended to put material on a gateway node (for instance, Free Haven and Publius choose which nodes to use), but techniques that depend on randomizing the use of nodes are inimical to using gateways.

Problems with requests

The problem with finding a path on a request is that the networks do not take into account the presence of gateways when routing a request message. Therefore, it is unlikely that a request message will randomly happen upon a gateway. The easy solution, of course, is to have everyone running a node on any network also run a gateway to all of the other networks. It's an ideal solution, but probably infeasible.

The following sections describe the routing techniques used by each of the systems we're looking at.

Freenet

Freenet requests are routed just like inserts, using the "best" path to the "epicenter." The length of the path is set by the user if the information is in Freenet (and if it is, we don't need a gateway), but longer paths take longer to fail if the key is not in the network. You could, of course, find a gateway by searching *all* of Freenet, assuming that the number of nodes in the network is less than or equal to the maximum path length. That would almost certainly take longer than you would care to wait. Freenet is designed so that if the file is in the network, the path to the file is usually short. Consequently, Freenet is not optimized for long paths. Long paths are therefore very slow.

Gnutella

Gnutella messages are broadcast to all nodes within a certain time-to-live, so choosing a path is not an issue. You can't choose a path even if you want to. The issue with Gnutella is that a gateway has to be within the maximum path radius, which is usually seven hops away. Fortunately, Gnutella is generally a very shallow network in which your node knows of a whole lot of other nodes. Generally, a gateway out of Gnutella to another system would have a high probability of being reached, since every request will potentially search a large percentage of the network. If there is a gateway node anywhere in the reachable network, it will be found. This is good if you want to access the whole world through Gnutella. Of course, it doesn't help at all if you want to gateway into Gnutella from another system.

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Mojo Nation

Mojo Nation requests are somewhat complicated. First, you must find the content you want on a content tracker that keeps a list of content and who has a copy of it. From the content tracker, you retrieve the address of a node that has a copy of the file part that you want. Then, you request the file part from the node. You do this until you have all of the parts needed to reconstruct the file.

This process actually lends itself quite well to gatewaying. As long as the gateways know what files are in Freenet, they can advertise for those keys. Unfortunately, gateways can't know what files are in Freenet. A gateway can only know what files have passed through it, which is only a fraction of the total content of the network.

However, if gateways also act as content trackers, they can translate requests for unknown keys into Freenet requests and place any keys found into the Mojo Nation content tracker index. In this way, you can access content from Freenet as long as you are willing to use a content tracker that is also a Freenet gateway. While it would be nice just to ask the network in general for a key and have it be found in Freenet (if appropriate), that is not how Mojo Nation works. In Mojo Nation, you ask a particular content tracker for content.

One way to integrate gatewayed and non-gatewayed content trackers in Mojo Nation would be to have a proxy node that acts as a Freenet gateway. Using that, any content tracker that functions as a gateway and a proxy could be used. The content tracker would be searched first, and if it failed, the gateway could be searched.

Publius

Gatewaying Publius is an interesting problem. Each file is split into a number of parts, each of which is sent to a different server. In order to reconstruct the file, you need a certain number of parts. It is therefore necessary for at least that number of parts to make it into gateways.

The length of the path for each part of the file is only 1 because the file goes directly to a single node and then stops. That means that if you need k parts of the file, k of the nodes contacted must be gateways in order for the file to be able to be reconstructed in the other network. The only solution, therefore, is to make most Publius nodes gateways.

Free Haven

Making a gateway out of Free Haven is not quite as difficult as making one out of Publius, because parts of files get routinely traded between nodes. Every time

Microsoft Exhibit 1019

a trade is made, the file part could potentially find a gateway, thus reaching the other network. However, when and how often files are traded is unknown and unpredictable. Thus, file trading cannot be counted on to propagate files, although it certainly will increase the probability of propagation by a nontrivial amount.

Gateway implementation

There is much theoretical work to be done in the area of making actual, working gateways between the different networks. However, even once that has been worked out, there is still the issue of implementation specifics.

There are a couple of ways that this could be approached. The first is to make each gateway between networks X and Y a hybrid X-Y node, speaking the protocols of both networks. This is undesirable because it leads to a combinatorial explosion of customized nodes, each of which has to be updated if the protocol changes for one of the networks.

A preferable solution would be to define a simple and universal interface that one node can use to query another for a file. Then, a gateway would consist merely of a cluster of nodes running on different networks and speaking different protocols, but talking to each other via a common interface mechanism. Using a common interface mechanism, gateway nodes would not even have to know what foreign networks they were talking to.

There are different possible interface mechanisms: CORBA,* RMI,† XML-RPC,‡ SOAP,§ etc. The mechanism that I would recommend is HTTP. It is a standard protocol for requesting a file from a particular location (in this case a particular node, which represents a particular network). Also, some file-sharing networks already have support for HTTP. Freenet and Gnutella support interfacing through HTTP, for instance.

Modification of the code base for each system to make a normal node into a gateway would be minor. The node need merely keep a list of gateways and, upon the failure of the network to find a requested file, query the gateways. If a file is found on a gateway, it is transferred by HTTP to the local node and then treated exactly as if it was found in the node's local data storage.

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* *http://www.corba.org*

† *http://java.sun.com/docs/books/tutorial/rmi*

‡ *http://www.xmlrpc.com*

§ *http://www.w3.org/TR/SOAP*

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Existing projects

Despite the desire for interoperability among networks, little has been done to facilitate this. Network designers are largely consumed by the difficult implementation details of their individual networks.

The only gatewaying project currently underway, to my knowledge, is the World Free Web (WFW) project, which aims to combine Freenet and the World Wide Web. While the Web may not at first seem like a file-sharing network as much as a publication medium, now that we find web sites offering remote hosting of vacation photographs and business documents, the two uses are merging into one.

Freenet and the Web complement each other nicely. Freenet is adaptive, temporary, and locationless, whereas the Web is static, semipermanent, and location-based. The point of the WFW project is to ease the load of popular content on web servers by integrating Freenet into web browsers. A WFW-enabled web browser will first check Freenet for the requested file. If the browser can't find the file, it will fetch the file from the Web and insert it into Freenet. The net effect is that popular web sites will load faster and the web servers will not crash under the load. This project, like many open source projects existing today, really needs only developers. The concepts are sound and merely call for experts on the various browsers to integrate them.

Conclusion

The peer-to-peer file-sharing developer community is not large. While the peer-to-peer world is expected to explode in popularity, those who have code in the here and now are few. There has been much discussion of interoperability among the various projects, so it may well happen. The technical challenges of routing requests to gateways are difficult ones, but certainly no more difficult than the challenges involved in anonymity, scalability, performance, and security that network designers have already had to face.

Acknowledgments

I'd like to thank Christine and Steve for contributing greatly to my second draft, and Isolde for making me write when I'd rather go to the park.

Microsoft Exhibit 1019

Afterword

Andy Oram, O'Reilly & Associates, Inc.

Like many new ideas with substantial "disruptive" potential (that is, ideas whose impacts can fundamentally change the roles and relationships of people and institutions), peer-to-peer has been surrounded by a good amount of fear. In particular, it has been closely associated in the public mind with the legal difficulties faced by Napster over claims that the company engaged in copyright infringement. The association is ironic, because Napster depends heavily on a central server where users register information. It is precisely the existence of the central server that makes it technically possible for a court to shut down the service.

However, Napster does demonstrate important peer-to-peer aspects. Files are stored on users' individual systems, and each download creates a peer-to-peer Internet connection between the source and destination systems. Furthermore, each system must furnish metadata information about the title and artist of the song. The legal questions Napster raises naturally attach themselves to some of the other peer-to-peer technologies, notably Gnutella and Freenet.

Precedents and parries

The Napster case in itself may not be dangerous to other peer-to-peer technologies. Its particular business model, its dependence on the preexisting popularity of exchanging MP3 files that are unauthorized copies of copyrighted material, and the many precedents for the concepts invoked by both sides (fair use, vicarious and contributory copyright infringement, substantial non-infringing uses) make the case unique.

But there are several indications that large copyright holders wield their legal weapons too widely for the comfort of technological innovators. For instance, during the Napster case, the band Metallica conducted a search for Metallica

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MP3s and created a list of 335,000 Napster users that it forced Napster to ban temporarily from the system. This raises the possibility that a determined plaintiff could try to prosecute all the individuals that form an entire community of peer-to-peer systems, such as Gnutella, Freenet, or Publius.

Users of those systems could then face the dilemma of being condemned for providing computer resources to a system that has social value, simply because one user of that system (perhaps a malicious user) provided material that raised the ire of a powerful commercial or political force. It would be interesting to see whether users would then try to invoke a kind of "ISP exemption," where they claim they are simply providing communications channels and have no control over content.

This legal status for ISPs is pretty well established in some countries. In the United States, numerous courts have refused to prosecute ISPs for Internet content. Still, a section of the enormous Digital Millennium Copyright Act, passed by the U.S. Congress in 1998, requires sites hosting content to take it down at the request of a copyright holder. Canada also protects ISPs from liability.

The status of ISPs and hosting sites is much shakier in other countries. In Britain, an ISP was successfully sued over defamatory content posted by an outsider to a newsgroup. The German parliament has shrouded the issue in ambiguity, stating that ISPs are responsible for blocking illegal content when it would be "technically feasible" to do so. Of course, some countries such as China and Saudi Arabia monitor all ISP traffic and severely restrict it.

France exempts ISPs from liability for content, but they have to remove access to illegal content when ordered to by a court, and maintain data that can be used to identify content providers in case of a court request. The latter clause would seem to make a system like Freenet, Publius, or Free Haven automatically illegal. The November 2000 French decision forcing Yahoo! to block the display of Nazi memorabilia auction sites sets a precedent that peer-to-peer users cannot ignore. It has already been echoed by a ruling in Germany's high court declaring that German laws apply to web sites outside the country. The trend will undoubtedly lead to a flood of specialized legal injunctions in other countries that try to control whether particular domain names and IP addresses can reach other domain names and IP addresses.

Further threats to technological development are represented by companies' invocation of copyrights and trade secrets to punish people who crack controls on software content filters or video playback devices. The latter happened in the much publicized DeCCS case, where the court went so far as to force web sites unrelated to the defendants to delete source code. In 1998, Congress acceded to

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the wishes of large content vendors and put clauses in the extensive Digital Millennium Copyright Act that criminalize technological development, like some types of encryption cracking and reverse engineering.

It would be irresponsible of me to suggest that copyright is obsolete (after all, this book is under copyright, as are most O'Reilly publications), but it is perfectly reasonable to suggest that new movements in society and technology should make governments reexamine previous guidelines and compromises. Copyright is just such a compromise, where government is trying to balance incentives to creative artists with benefits to the public.

Napster showed above all that there is now a new social context for music listening, as well as new technological possibilities. The courts, perhaps, cannot redefine fair use or other concepts invoked by both sides in the Napster case, but the U.S. Congress and the governing bodies of other countries can ask what balance is appropriate for this era.

Who gets to innovate?

Peer-to-peer, like all technologies, embodies certain assumptions about people and future directions for technology. It so happens that peer-to-peer is moving the compass of information use in a direction that directly contradicts the carefully mapped-out plans drawn by some large corporate and government players.

The question now posed is between two views of how to use technology and information. One common view gives consumers and users the maximum amount of control over the application of technology and information. One example will suffice to show how powerful this principle can be.

Despite Tim Berners-Lee's hope that the World Wide Web would be a two-way (or even multiperson to multiperson) medium, early browsers were pretty much glorified file transfer programs with some minimal GUI elements for displaying text and graphics together. The addition of CGI and forms allowed users to talk back, but did not in itself change the notion of the Web as an information transfer service. What caused the Web to take on new roles was the crazy idea invented by some visionary folks to use the available web tools for selling things. An innovative use of existing technology resulted in an economic and social upheaval.

Putting tools in the hands of users has an impact on business models, though. People might no longer buy a technical manual from O'Reilly & Associates; they might download it from a peer instead—or more creatively, extract and combine pieces of it along with other material from many peers. And peer-to-peer,

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of course, is just a recent option that joins many other trends currently weakening copyright.

When a revenue stream that information providers have counted on for over 2000 years threatens to dry up, powerful reactions emerge. Copyright holders have joined with a wide range of other companies to introduce legal changes that revolve around a single (often unstated) notion: that the entity providing information or technology should control all uses of it. The manufacturer of a disk decides what devices can display it. A compiler of information decides how much a person can use at any one time, and for how long. The owner of a famous name controls where that name can appear.

Trying to plug serious holes in the traditional web of information control—copyrights, trade secrets, patents, trademarks—information owners are extending that control into areas where they have previously been excluded. In their view, new ideas like selling over the Web would have to come from the company who provides the media or the service, not from people using the service.

So where do we look for the future uses of information and technology? The two answers to this question—users versus corporate owners—are likely to struggle for some time before either a winner or a workable compromise appears. But the thrust of peer-to-peer implicitly throws its weight behind the first answer: trust the users. The technological innovations of peer-to-peer assume that users have something to offer, and some peer-to-peer projects (notably Jabber in its community-building, and Gnutella in its search model) actually encourage or even provoke users to contribute something new and different.

A clean sweep?

Some people ask whether peer-to-peer will replace the client/server model entirely. Don't worry, it emphatically will not. Client/server remains extremely useful for many purposes, particularly where one site is recognized as the authoritative source for information and wants to maintain some control over that information.

Client/server is also a much simpler model than peer-to-peer, and we should never abandon simplicity for complexity without a clear benefit. Client/server rarely presents administrative problems except where the amount of traffic exceeds the server's capacity.

Peer-to-peer is useful where the goods you're trying to get at lie at many endpoints; in other words, where the value of the information lies in the contribu-

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tions of many users rather than the authority of one. Peer-to-peer systems can also be a possible solution to bandwidth problems, when designed carefully. (Of course, they can also cause bandwidth problems, either because their design adds too much overhead or because people just want a lot of stuff without paying for the bandwidth that can accommodate it.)

In short, peer-to-peer and client/server will coexist. Many systems will partake of both models. In fact, I have avoided using the phrase "peer-to-peer model" in this book because such a variety of systems exist and so few can be considered pure peer-to-peer. The ones that are completely decentralized—Gnutella, Freenet, and Free Haven—are extremely valuable for research purposes in addition to the direct goals they were designed to meet. Whether or not other systems move in their direction, the viability of the most decentralized systems will help us judge the viability of peer-to-peer technology as a whole.

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appendix

Directory of Peer-to-Peer Projects

This is a partial list of interesting projects, companies, and standards that could reasonably be considered examples of peer-to-peer technology. It is not meant to be exhaustive, and we apologize for any worthy projects that were not included. The field, of course, expands constantly.

Agents as Peers

- Infobot
- Sandia National Laboratories
- WebV2

Collaboration

- Engenia Software, Inc.
- eZ
- Interbind

Development Frameworks

- Mithral Communications & Design, Inc.
- WorldOS Corporation

Devices as Peers

- Bluetooth
- Brazil Project
- dHTTP (Distributed HTTP)
- Endeavors Technology, Inc.
- Jini

Distributed Computation

- 2AM
- Applied MetaComputing
- Centrata
- Datasynapse

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Distributed.net
DistributedScience
Entropia
Parabon Computation
Popular Power
Porivo Technologies, Inc.
SETI@home: The Search for Extraterrestrial Intelligence
Ubero
United Devices, Inc.: Individuals Accelerating Science

Distributed Search Engines

gonesilent.com (aka InfraSearch)
OpenCOLA
Plebio
WebV2

File Sharing

CuteMX.Com (GlobalScape, Inc.)
File Navigator
Free Haven
Freenet
Gnutella
Hotline Communications, Ltd.
Jungle Monkey
Mojo Nation
Napster
Ohaha
OnSystems, Inc.
OpenNap
Pointera
Publius
Spinfrenzy.com

Gaming

2AM
CenterSpan

Internet Operating System

Applied MetaComputing
Globus
ROKU
Static

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Licensed Media Distribution
- eMikolo
- Flycode
- Kalepa Networks, Inc.

Messaging Frameworks
- AIMster
- BXXP
- CenterSpan
- IMXP
- Jabber

Metadata
- RDF
- RSS
- XNS (eXtensible Name Service)

Servers/Services as Peers
- .NET
- BXXP
- Meerkat: An Open Wire Service
- Simple Object Access Protocol (SOAP)
- Universal Description, Discovery and Integration (UDDI)
- XML-RPC

Superdistribution
- 2AM
- 3Path
- Freenet
- vTrails

The Writable Web
- Amaya Web Editor/Browser
- Blogger
- Brazil Project
- Endeavors Technology, Inc.
- Manila
- Radio Userland
- WebDAV
- Wiki Wiki Web

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Contributors

NELSON MINAR, CTO and cofounder of distributed computing leader Popular Power, has an extensive history researching Internet systems. While at the MIT Media Lab, he built a mobile agent based peer-to-peer computing platform called Hive, and previously studied agent-based modeling at the Sante Fe Institute. Minar sees the Internet as a place, a world with its own rules and behaviors.

MARC HEDLUND is the chief executive officer and cofounder of Popular Power, the first distributed computing company to launch commercial software. He previously served as the founder and director of Lucasfilm Ltd.'s Internet division and director of engineering at Organic Online. His Internet experience dates to early 1994, when he worked on several IETF committees and built early e-commerce applications while CTO of a Web start-up.

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TIM O'REILLY is founder and president of O'Reilly & Associates, Inc. Tim's goal is to enable change by capturing and transmitting the knowledge of innovators and innovative communities via books, conferences, and web sites.

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DANIEL BRICKLIN, a software designer, is best known as the cocreator of VisiCalc, the first electronic spreadsheet. In addition to the spreadsheet, he helped develop one of the first word processing systems in the mid-1970's, programmed the most popular prototyping tool of the MSDOS world, and helped introduce the world to the capabilities of electronic ink on pen computers. Mr. Bricklin has served on the boards of the Software Publishers Association and the Boston Computer Society and has received many honors for his contributions to the computer industry, including the IEEE Computer Society's Computer Entrepreneur Award and Lifetime Achievement Award from the SPA. Most recently he is the founder and CTO of Trellix Corporation which creates web site building systems.

DAVID P. ANDERSON is the director of the SETI@home project. He cofounded Tunes.com and is currently CTO of United Devices. From 1985 to 1991 he was on the faculty of the U.C. Berkeley Computer Science Department.

JEREMIE MILLER has been developing Internet-related and Open Source projects since 1993, having been involved with the early Web standards and projects such as Apache and Linux. In 1997 he started following the DHTML and XML standards very closely, and in 1998 founded Jabber, an Open Source movement designed to create a new, standard, distributed XML-based platform for instant messaging and presence applications. Today he continues developing Jabber, helping to advance the new XML infrastructure available on the Internet.

ADAM LANGLEY is student in England and a free software programmer in his free time. Interests range from software to politics/freedom, typesetting, and theater. His current project is an implementation of Freenet in C++, with which he could really do with some help.

GENE KAN was among the first to produce an open source version (under the GNU General Public License) of Gnutella software after Gnutella was released by Justin Frankel and Tom Pepper of Gnullsoft. Soon, Mr. Kan became one of Gnutella's key spokesmen. Previously an SGML/XML consultant and kernel network engineer at Check Point Software, Mr. Kan is now CEO of InfraSearch. In his spare moments he enjoys racing his cars, LTLENDN and BIGENDN.

ALAN BROWN is currently the assistant director of a human rightsorganization. He has served on the executive committee of an ACLU state affiliate and taught mathematical logic at several midwest universities. He will launch a new

Microsoft Exhibit 1019

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MARC WALDMAN is a Ph.D. candidate in Computer Science at New York University. He is one of the co-developers of the Publius censorship-resistant publishing system. His research interests include privacy-enhancing technologies and computer security. Marc received a BA and MS in Computer Science from New York University.

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ROGER DINGLEDINE graduated from MIT in May 2000 (B.Sc. computer science, B.Sc. mathematics, M.Eng. computer science and electrical engineering), where his Master's research in anonymous distributed publishing systems was supervised by Ronald Rivest. He is project leader for both the Simple End-User Linux project (seul.org) and the Free Haven project (freehaven.net). Currently he works as the Security Philosopher for Reputation Technologies, Inc. (reputation.com).

MICHAEL J. FREEDMAN is a graduate student in computer science at MIT. His research interests focus on cryptography and computer/network security, especially in the realm of distributed systems. He is a principal researcher of the Free Haven project, and has worked at Zero-Knowledge Systems implementing an electronic cash architecture.In his spare time, Michael enjoys climbing, mountaineering, and other outdoor pursuits, much to the concern of family and friends.

Microsoft Exhibit 1019

DAVID MOLNAR began using PGP in 1993. He became interested (obsessed?) with figuring out "why it worked" and has been studying cryptography ever since. Now an undergraduate at Harvard University, he keeps up with security issues by attending courses, reading newsgroups, mailing lists, and conference papers, and attending DEF CON in his home city of Las Vegas. David is an ACM Student Member and a member of the International Association for Cryptologic Research.

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Microsoft Exhibit 1019

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Microsoft Exhibit 1019

Index

A

Abe, Masayuki, 328
access control lists, POWs not needed, 288
account passphrase, 364
  cryptographic keys, 378
  security concerns and, 377
accountability, 11, 20, 271–340
  buddy system and, 173
  centralized control over resources, 272
  changing the level of, 305
  complicated by revoking documents, 170
  difficulty of, 274–283
  engineering polite behavior, 20
  Free Haven case study, 329–339
  fungible micropayments and, 293
  junk mail and, 279, 288
  micropayment schemes and, 286–305
  minimizing a server's threat, 277–279
  parallel solutions and, 290
  peer-to-peer and impact on, 276
  real world vs. Internet, 274
  vs. anonymity, 329
accountability slider, 303–305
active caching
  adversaries and, 285
  maintaining data availability with, 285
  protection against Slashdot effect, 285
active mirroring
  Akamai Technologies, 284
  maintaining data availability with, 284
  protection against Slashdot effect, 285
active-server document-anonymity, 164
ActiveState, 45
Adams, Rick, 50
Adar, Eytan, 206, 292
addresses, dynamic IP (see dynamic IP addresses)
addressing, protocol-centric, 30–32
ADSL (Asymmetric Digital Subscriber Line), 14, 36
adversarial approach to scoring systems, 324
Advogato, 311–313
  how trust is determined, 311
  resisting pseudospoofing, 312
  trust metric vs. Slashdot moderation system, 311
Aiken, Alexander, 348
AIM (AOL Instant Messenger)
  attacking by shilling, 316
  protocol-centric addressing, 30–32
AIMster, 52
Akamai Technologies, 284
Albert, Réka, 222, 225
all points payment model, 332
  node-specific tickets and, 334
"all you can eat" business models, 33
alt newsgroups, 6
America Online (AOL) and Gnutella, 95
amortized pairwise payment model, 331
Anderson, David, xv, 67–76, 404
Anderson, Ross, 170, 262
anonymity
  analysis of, 181–184
  attacks on, 180
  computational, 181–184
  Crowds software and, 258
  Freenet and, 125
  on the Internet, 251, 275

Microsoft Exhibit 1019

anonymity (*continued*)
Napster users and, 264
partial, 165
peer-to-peer, 117–120, 160
perfect forward, 181–184
reliability and, 161–163
types of, 163–165
vs. accountability, 329
why it is needed, 251
Anonymizer.com, 144, 252
anonymizing proxies, 251, 256
anonymous
digital cash (see digital cash)
Gnutella chat, 120
macropayment digital cash (see macropayment digital cash schemes)
publishing systems, 261
Eternity Service, 262
Freenet, 123–132, 265
problems with, 266
Publius, 145–158
searching technologies, 266–269
remailers, 89–93
servers, 277
storage (Free Haven), 163–165
anti-censorship strategy, 133–144
AOL (America Online) and Gnutella, 95
application service provider (ASP), 29
application-to-application conversations, 78
Arecibo radio telescope, 68
Arnold, Ken, 39
ARPANET, 4
ASP (application service provider), 29
Asthagiri, Nimisha, xvii, 354–380, 406
asymmetric algorithm, default, 361
asymmetric bandwidth, 14, 36
asymmetric keys, 379
authenticators (MAC), 366
fetching lost messages, 369
supporting message fanout, 366–368
author-anonymity, 163, 182
automatic "pass-along" participation and Napster, 49
avatars, selling, 283

B

Back, Adam, 182, 185, 187, 263, 289
Bailin, J., 203
Bakos, Y., 302
bandwidth, 397
asymmetric, 14, 36
Gnutella queries, used by, 239
junk mail and, 279
managing congestion to prevent attacks, 303
replenishable resource, 32
system capabilities and, 206
use by Napster and Gnutella, 271
using wisely, 19
verifying allocation, 338
(see also performance)
Barabási, Albert-László, 222, 225
Beam, Dan and Meredith, 40
Bellare, Mihir, 338
benchmark sources, 351
Berners-Lee, Tim, 53
Berthold, Oliver, 334
best effort packet delivery, 11
blind signatures, 297, 334
normal vs. restrictive protocols, 298
blinded proto-coins, 297
blinded receipts used for transactions, 326
blocking web sites, 133–135
using IP addresses, 139
Blogger, 54
bootstrapping
centralized vs. decentralized systems, 319
obstacle to Reputation Server, 351
reputation systems, 319
Born, Susan, 187, 340
Box, Don, 39
Boyd, Colin, 338
Brainard, John, 289
Brands, Stefan, 298, 340
digital cash scheme
offline checking of double-spending for fraud tracing, 298
payment in, 299
restrictive blind signature protocol vs. Chaum's protocol, 298
Bray, Tim, 82
bread pudding protocol, 296
Brickley, Dan, 191–202, 406
Bricklin, Dan, xv, 49, 54, 59–63, 404
broadcasts
reducing (Gnutella), 115–117
TCP-based, 104
Brown, Alan, xv, 133–144, 404
Brown, Lawrie, 310

Microsoft Exhibit 1019

browsing
  Jabber, 87
  web (see web browsers)
Bruce, Matt, 241
Brynjolfsson, E., 302
buddy lists (instant messaging services), 86
buddy notifications, 174
buddy system (Free Haven), 335
  accountability and, 173
  attacks on, 179
bulletin boards and upload/download ratios, 307
Burbeck, Steve, 39
business models
  "all you can eat", 33
  Beam, Inc., 40

C

cable modems, 14, 36
caching (see active caching)
Camarda, Tom, 109
Canada and legal status of ISPs, 394
Canter and Seigel "green card spam", 10
CAs (certifying authorities) and digital certificates, 246–248, 369
categories of transactions, weighting by, 323
CDDB, a case study, 61
CD-ROMs vs. HTML, 26
CDs and metadata queries, 199–201
cellular telephony and Gnutella, 110
censorship
  legal attacks as, 177, 261
  physical threats as, 177, 261
  in Publius, 253–261
  Red Rover's fight against, 133–144
  search engines as route to, 266–269
  social pressure as, 177
  using Publius to resist, 145–158
censorship-resistant publishing systems
  trust in, 249–256
  trust issues in, 269
  why they are important, 146
centralization (see decentralization)
centrally coordinated peer-to-peer systems, 204
certificates, digital (see digital certificates)
certifying authorities (CAs) and digital certificates, 246–248, 369
certifying keys, 308
chains of paywords (hash values), 294
characteristic pathlengths, 210
  calculating, 218
  measuring optimal path between nodes, 219
  scaling logarithmically with network size
    Freenet, 229
    Gnutella, 239
  shortening via small-world model, 215
  of World Wide Web, 222
Charfman, J.J., 203
Chaum, David, 256, 325, 338
  development of electronic cash system, 297
  normal blind signature protocol vs. Brands's protocol, 298
China and legal status of ISPs, 394
CHKs (Content Hash Keys), 129
Christenson, Clayton, 45, 352
cipher keys, 364
cipher pairwise keys, 366
ciphertext messages, 149
Clark, Keith, 241
Clarke, Ian, 39, 123, 241
client puzzles, 289
  preventing bandwidth attacks, 303
  sliding scale for payment, 304
client software, Publius (see Publius, client software)
clients (Red Rover), 137
  enticements to running, 143
  governmental cloning of, 141
  life cycle of, 139
client/server model
  switch from network model to, 9
  ÜberClient and, 382
  vs. Gnutella, 97–99
  vs. peer-to-peer, 4, 26, 35, 75, 396
  Web problems traced to, 123
Clip2.com, Inc., 50, 112, 116
  Gnutella's scalability barrier, 206
clustering coefficient of vertices, 212–215
  calculating, 218
Coase, Ronald, 352
Cocconi, Giuseppe, 68
coins, verifying, 300
collaboration and security problems, 357
collecting ratings, 318
collisions
  full, 295
  partial hash, 289

Microsoft Exhibit 1019

commercialization of the Internet, 8–15
communications channel, Free Haven, 166, 174, 333
computational anonymity, 181–184
computations, SETI@home, 69
confidence values
of Free Haven servers, 337
for scores, 320
congestion management
micropayments and, 302
preventing bandwidth attacks, 303
congestion on the network, 11–13
connected graphs, 209
Freenet case study, 216
Gnutella case study, 232
connection depletion attacks and client puzzles, 290
connections, TCP, 11–12
connectivity
Internet (pre-1994), 23
peer-to-peer and, 22–26
transient, 25
consumers and information providers, split between, 35–37
contacts (Groove), 370
Content Hash Keys (CHKs), 129
content syndication formats, 55
content trackers, 164
content-at-the-center data model, 29
conversational technologies, 77–88
conversations
future possibilities for, 80
Internet, unifying, 82
managing, 88
types on the Internet, 78
cooperation, breakdown of, 10–13
copyright
decentralized peer-to-peer and, 122
threats to technology, 393–396
Cormen, Thomas, 312
cornucopia of the commons, 59–63
cost/benefit analysis for participants in peer-to-peer, 303
Cottrell, Lance, 89, 144, 174, 276
Cox@Home vs. Napster, 35
CPU-based payment schemes, 260
Cranor, Lorrie Faith, xvi, 145–158, 242–270, 405
"credentials without identity", 299
credibility
of Free Haven servers, 337
of raters, used to weight ratings, 321
of sources (Reputation Server), 350
Crowds software, 258
Crypto++ and Publius proxy, 255
CRYPTO conference, 300
cryptographic algorithms for shared spaces, 361
cryptographic keys (Groove), 378–380
cryptographically secure hash functions, 245
cryptography
client puzzles and connection depletion attacks, 290
digital cash schemes and, 300
fundamentals (Publius), 149
micropayments vs. macropayments, 296
public key, 246
(see also encrypting)
cryptosystems, recipient-hiding, 333
currency
exchanging for information, 386
real-world vs. digital cash, 293
currency system, digital, 262
Cypherpunks remailer network, 167, 175

D

data, caching and mirroring, 284
data synchronization, 355, 360
databases
organized manual, 60–62
organized mechanical, 60
shared, 60–63
volunteer, 60–63
data-clustering techniques and scoring algorithms, 323
data-culling method (Freenet), 128
DataRequest (Freenet), 125
Dawkins, Richard, 40
DDoS attacks (see distributed denial of service attacks)
decentralization, 16
breaking the IT bottleneck, 58
Freenet and, 123–132
Gnutella and, 97, 121
Napster and, 28

Microsoft Exhibit 1019

peer-to-peer systems, 204
performance issues, 205–240
using many hops to send messages, 205
Usenet and, 5–7, 50
decentralizing scoring systems, 326–328
deconstruction process, 352
decrypting
files in Publius, 153
keys in Groove, 362–364
messages, 149
degradation (see performance)
Delete operation (Publius), 155
delta messages (see Groove, delta messages)
denial of service (DoS) attacks
common methods for dealing with, 284–286
foiled by fungible micropayments, 293
Free Haven and, 178
micropayments can be targets of, 300
nonfungible micropayments not a deterrent to, 292
POWs can protect against, 286
protecting peer-to-peer from, 273
reciprocity of contribution and, 386
using Publius to resist, 259–261
Detweiler, L., 280
Device Presence Protocol (DPP), 367
Diffie-Hellman public/private key pair, 365, 372, 379
digest functions, 245
digests, calculating, 289
DigiCash, 293, 297
digital cash, 286
anonymous vs. identified, 287
Brands's scheme (see Brands, Stefan)
cryptography and, 300
double-spending a problem with, 298
how it works, 297
macropayment (see macropayment digital cash schemes)
MicroMint, 294
PayWord, 294
preventing DoS attacks, 261
private and real-world currencies, 293
sliding scale for payment, 305
using reputation to reduce risk in transactions, 343
digital certificates, 246–248
anonymous publishing systems and, 266
conveying reputation information, 243
creating, 246
PayWord and, 294
verifying public keys, 246
digital currency system (Mojo), 262
digital IDs, 359
Digital Millennium Copyright Act, 394
Digital Rights Management (DRM), 33
digital signatures, 246–248
anonymous publishing systems and, 266
blind signatures, extension of, 297
verifying, 247
Dingledine, Roger, xvi, 144, 159–187, 271–340, 405
directed graphs, 209
Freenet case study, 217
trust relationships in Advogato, 311–313
disruptive technologies, 45
distributed computation, 51
managing metadata and, 57
SETI@home and, 67–76
distributed denial of service (DDoS) attacks
parallel solutions overwhelmed by, 290
resource allocation and, 340
using Publius to resist, 145, 158
distributed indexes, 267
distributed intelligence system (Gnutella), 100–103
Distributed Search Solutions (DSS), 50
DMCI (Dublin Core Metadata Initiative), 192
DNS (Domain Name System), 7
addresses vs. peer-to-peer addresses, 24
bypassing, 24
protocol-centric addressing and, 31
document-anonymity, 164
Gnutella does not provide, 182
Mojo Nation provides, 183
Publius provides, 183
Domain Name System (see DNS)
domains, reputation, 343–345
weak vs. strong entities, 345
donating resources, 292
dormant servers, 176
Dornfest, Rael, xvi, 39, 191–202, 406
DoS attacks (see denial of service attacks)
DoubleClick, Inc., 124
double-spending
hash cash protects against, 289
MicroMint protects against, 295
offline checking for fraud tracing, 298
online verification to prevent, 298

Microsoft Exhibit 1019

double-spending (*continued*)
paywords, 294
preventing, 287, 333
download accelerator programs, 12
downloaded software
risks and solutions, 245, 266
trust issues and, 244–249
DPP (Device Presence Protocol), 367
Drake, Frank, 68
DRM (Digital Rights Management), 33
DSS (Distributed Search Solutions), 50
Dublin Core Metadata Element Set, 194
Dublin Core Metadata Initiative (DMCI), 192
Dupuit, Jules, 301
Dwork, Cynthia, 288
dynamic IP addresses, 13, 54
denial of service attacks and, 260
encrypted (Red Rover), 136–139
IPv6 and, 19
Mosaic demand and, 23
protocol-centric addressing, 30–32
dynamic participation
Free Haven and, 277
on the Internet, 275
dynamic routing, Gnutella, 106

E

Eazel, 45
eBay
feedback system, 310
fraudulent behavior on, 313
reputation scores, problems with, 316
eCash, 293
e-cash payment schemes (see digital cash)
Ecash Technologies, 297
EditThisPage.Com, 54
eGroups and creating shared spaces, 357
electronic cash system, 297
ElGamal algorithm, 361, 379
elliptic curve cryptography, 296
email
as a form of conversation, 79
collaboration issues with, 357
encrypting messages, 90–93, 256–258
frustrating traffic analysis, 92
information leakage easily occurs, 359
mixing centralization and decentralization, 51
remailers and, 89–93
security risks in, 356
shared spaces, weaknesses of, 358–360
vs. Groove, 356–358
why it is not peer-to-peer, 25
encrypting, 149
delta messages, 362
documents with Publius, 147–149
email messages, 90–93, 256–258
Freenet documents, 131
IP addresses (Red Rover), 136–139
micropayments, 333
on the Web, 359
using PGP (Pretty Good Privacy), 248
(see also cryptography)
encryption/decryption key pairs, 363, 379
end-to-end payment model, 330
Enterprise Resource Planning (ERP) systems, 346
entities
in reputation domains, 343–345
Reputation Server and, 345
ERP (Enterprise Resource Planning) systems, 346
Eternity Service, 387
trust issues and, 262
Eternity Usenet, 263
anonymity properties of, 181
Ethernet and Gnutella network, 111
EUROCRYPT conference, 300
Evans, Philip, 352
expiration dates of shares (Free Haven), 169
extraterrestrial signals (SETI@home), 67, 71

F

fanout, message, 366–368
Fast Fourier Transform (FFT) algorithm, 68
fault tolerance, 203
simulating
in Freenet, 224–227
in Gnutella, 234–238
Faybishenko, Yaroslav, 101
Federrath, Hannes, 334
feedback
soliciting from parties in transactions, 318, 342–353
system on eBay, 310
trusting sources of, 350
FFT (Fast Fourier Transform) algorithm, 68

Microsoft Exhibit 1019

file sharing
between Free Haven and Freenet, 384
DNS and, 7
Freenet and, 123–132
next-generation peer-to-peer technologies, 121
file tampering, detecting, 245
file-sharing systems
searching technologies for, 266–269
trust issues and, 263–265
unifying with an OmniNetwork, 381
Financial Cryptography conference, 300
firewalls, 368
abuse of port 80, 17
as obstacles to peer-to-peer, 13
making smarter, 18
no good if internal network is compromised, 357
no longer a guarantee of protection, 357
Fishburn, P.C., 302
flat-fee methods vs. pay-per-use methods, 302
floating-point operations and SETI@home, 74
flooding attacks, 178
common methods for dealing with, 284–286
foiled by fungible micropayments, 293
protecting against, 285
protecting peer-to-peer from, 273
Ford-Fulkerson algorithm and Advogato, 312
forgery
macropayment techniques for protecting against, 296
preventing with micropayment schemes, 287
thwarted by MicroMint, 295
France and legal status of ISPs, 394
Frankel, Justin, 95
Free Haven, 144, 159–187
accountability, 159
buddy system and, 173
case study, 329–339
impact on, 277
solving problem of, 279, 304
anonymity, 159–165, 181–184
attacks on, 180
properties of, 181
attacks on, 176–181
buddy system (see buddy system (Free Haven))
choosing good algorithms, 336
communications channel, 166, 174, 333
design of, 165–176
documents
attacks on, 177
not possible to revoke, 169
retrieving, 168
storing new, 167
efficiency problems, 185
flexibility, 159
freeloader problem, 293, 303
goals of, 159
introducers, 166, 176, 337
micropayments in, 333–335
nonfungible micropayments work best with, 334
OmniNetwork, what it can add to, 387
persistence, 159
privacy in data-sharing systems, 160–163
problems in design of, 184–186
pseudonyms on, 162
public keys, 167–169
publication system, 166–168
receipts, 171–173
attacks on, 180
reply blocks, 167
reputation referrals, 336
reputation (see reputation, Free Haven)
routing techniques, 390
scores and ratings, 336
sending broadcasts in batches, 333
servers
adding/removing, 176
as introducers, 166, 176, 337
broadcasting referrals, 336
contracts formed by, 165
credibility/confidence values, 337
punishing misbehaving, 335
servnet, 165
attacks on, 177
dynamic nature of, 167
introducing files into, 167
shares
buddy system, 173
expiration dates of, 169
hoarding, 178
receipts and, 171–173, 180
replicating not allowed, 174
trading, 166, 170–172

Microsoft Exhibit 1019

Free Haven (*continued*)
trust issues and, 262, 338
vs. Freenet, 160, 169
vs. Gnutella, 160
vs. Mojo Nation, 160, 169
vs. Publius, 160
when Free Haven data is requested through Freenet, 384
Free Haven Protocol, 337
free riders
impact on Freenet, 229
impact on Gnutella, 238
performance degradation and, 206
vs. malicious nodes, 229
free software
meme map, 41–43
summit, 38, 43
Free Software Foundation (FSF), 41
Freedman, Daniel, 340
Freedman, Michael, xvi, 159–187, 271–340, 405
Freedom (Zero-Knowledge Systems), 175, 258
freeloading
congestion caused by, 303
resource allocation and, 292
Freenet, x, 123–132
accountability problem, solving, 279
active caching, 285
algorithm for forwarding queries, 216
anonymity and, 125
anonymity properties, 181, 183, 217
caching approach to storage space problem, 304
censorship attempts, 131
combined with World Wide Web, 392
creating highly connected nodes, 228
data-culling method, 128
design deficiencies, 385
distributed indexes, 267
free riding, impact on, 229
future of, 132
hash functions, 129
introductory nodes
adding new nodes using, 223
reducing dependence on, 216
keys, 128–132
Content Hash Keys (CHKs), 129
Keyword Signed Keys (KSKs), 129
redirects and, 131
Signature Verification Keys (SVKs), 130
link distribution in, 228
node failure
due to random removal, 225
due to targeted attack, 226
vs. Gnutella, 236–238
OmniNetwork, what it can add to, 384
passive-server document-anonymity, 164
performance case study, 215–231
pseudonyms on, 130
request and insert mechanisms, 216
requests, 124–128
checking the data store, 126–128
details of, 125
dropping, 124
routing, 126–128
routing techniques, 389
scalability and, 229–231
searching problems, 196
simulating
behavior over time, 217–222
fault tolerance, 224–227
growth in, 222–224
stack model, 126–128
trust issues and, 265
URIs (Uniform Resource Indicators), 129
vs. Free Haven, 160, 169
vs. Gnutella, 103, 127
vs. Publius, 147
when Freenet data is requested through Free Haven, 384
frequent correspondent lists, POWs not required for, 288
FSF (Free Software Foundation), 41
full collisions, 295
fungible micropayments, 286, 291–300
amortized pairwise payment model, 332
as accountability measure, 293
pairwise payment model and, 330
redeemable for real-world currencies, 293

G

Garay, Juan, 338
gateways
advantages of, 383
between Freenet and Free Haven, 383
creating, problems with, 388–391

Microsoft Exhibit 1019

existing projects, 392
implementing, 391
inserts, problems with, 388
interoperability through, 381–392
requests, problems with, 389–391
Gaussians (signals), 71
gcc (GNU C Compiler), 41
Gedye, David, 67
GeoCities, 54
Germany and legal status of ISPs, 394
GhostScript, 41
GNU C Compiler (gcc), 41
GNU Emacs, 41
GNU General Public License (GPL), 41
Gnutella Version 0.56, 95
GNU tools, 43
GnuCache, 113
Gnut software for Unix, 113
Gnutella, x, 94–122
accountability problem, solving, 279
analogues to, 109–112
anonymity properties of, 181
anonymous chat, 120
bandwidth, using too much, 271
blending client, server, and network, 100
cellular telephony and, 110
creating an ad hoc backbone, 108
design deficiencies, 385
distributed intelligence system, 100–103
distributed search engines and, 268
dynamic routing, 106
Ethernet analogous to, 111
exchanging micropayments among peers, 330
free riding, impact on, 238
freeloading on, 292
horizon concept, 110
host caches and, 113–115
how it began, 95
impact on accountability, 276
InfraSearch and, 101, 196
legal issues, 117
link distribution in, 234
message-based routing, 104
node failure
due to random removal, 234–238
due to targeted attack, 234–238
vs. Freenet, 236–238
OmniNetwork, what it can add to, 385
performance case study, 231–240
placing nodes on the network, 108
private networks, 115
pseudoanonymity and, 118–120
querying the network, 100–103, 232–234
reducing broadcasts causes big impact, 115–117
Reflectors, 50, 112, 116
routing techniques, 389
saved by open source developers, 96
scalability and, 239
simulating
behavior over time, 232–234
fault tolerance, 234–238
super peers and, 50, 240
TCP-based broadcasts, 104
traffic problems with, 112–117
transmission loss over TCP, 107
trust issues and, 264
TTL (time-to-live) numbers, 105
vs. client/server model, 97–99
vs. Free Haven, 160
vs. Freenet, 103, 127
vs. Napster, 97–99
vs. Publius, 147
#gnutella (IRC channel), 96
Goldberg, Ian, 175
Goldreich, Oded, 185
Goldschlag, D.M., 175
Gottbrath, C., 203
GPL (General Public License), 41
Gnutella Version 0.56, 95
Gracenote, 61
graph theory, 207–211
characteristic pathlengths, 210
connectedness, 209, 216
directed graphs, 209
random graphs, 213
regular graphs, 212–214
small-world graphs, 214
social networks and, 207
transitive paths, 210
Great Britain and legal status of ISPs, 394
"green card spam", 10
Greenlaw, Raymond, 291
Groove, 354–380
centralizing crucial aspects of security, 376–378
communication protocols and routing messages, 367

Microsoft Exhibit 1019

Groove (*continued*)
contacts, 370
cryptographic algorithms, 361
cryptographic keys used in, 378–380
delta messages, 360
adding rekey information, 374
authenticity of, 362
encryption handled by a symmetric key, 363
New-Member-Added, 373
encryption/decryption key pairs, 363, 379
identities
creating, 370
defined by public/private key pairs, 362
in mutually-trusting shared spaces, 362–364
instant messages, 363
invitation protocol, 369, 372
message sequencing, 375
mutually-suspicious shared spaces, 365–369
mutually-trusting shared spaces, 362–365
operating in a hostile environment, 355
passphrase, 364
cryptographic keys, 378
memorizing, 377
PGP vs. S/MIME, 370
PKIs and, 369–371
rekey delta messages, 374
using a relay server for peer communication, 366–368
relaying vs. direct communication, 368
SDSI/SPKI and, 370
security guarantees made by, 356
shared spaces, 354, 360
key benefits of, 355
signature/verification key pairs, 363, 379
transceiver, 360
uninvitation protocol, 374–376
vs. email, 356–358
vs. S/MIME, 359
group keys (Groove)
deleting old, 375
encrypting delta messages, 364
given to new members, 372
reestablishing the, 374
trusting vs. suspicious groups, 366
groupware system (Groove), 354

H

Hale, Thomas, 99
Hall-Beyer, Ian, 96, 109
Halting Problem, 249
Hardin, Garrett, 49, 63, 272
hash cash, 289
sliding scale for payment, 304
hash functions
cryptographically secure, 245
Freenet, 129
Publius, 150
Hedlund, Marc, 3–20, 39, 403
Helsingius, Johan, 292
Hertzfeld, Andy, 39
hierarchical model (DNS), 7
hierarchical peer-to-peer systems, 204, 240
HMAC-SHA1 (default algorithm for MAC), 364, 366, 380
hoarding shares in Free Haven, 178
Hong, Theodore, xvi, 187, 203–241, 340, 406
Hoover, H. James, 291
hops-to-live count (Freenet), 124, 131
Hopwood, David, 333
Hosein, Gus, 144
host caches and Gnutella, 113–115
host catchers (Gnutella), 114, 231
host lists vs. host caches (Gnutella), 114
HTML
Dublin Core metadata in documents, 194
why it is successful, 27
HTTP
abuse of port 80, 17
interface mechanism for gateways, 391
Publius implemented over protocol, 147
HTTP POST operation, 156
HTTP proxy (Publius), 146, 148
implementation, 156
local vs. remote proxies, 255
Retrieve operation and, 152–154
hub (Red Rover), 136
Huberman, Bernardo, 206, 292
hyperlinks, web, 53

I

ICQ
launch of, 24
why it is peer-to-peer, 25
identification flooding attacks, 326

Microsoft Exhibit 1019

identities
as elements of reputation, 345
Groove (see Groove, identities)
identity-based payment policies, 301
information dispersal algorithm (IDA), 168
information providers and consumers, split between, 35–37
InfraSearch, 101, 196
InsertRequest (Freenet), 125
inserts, problems with, 388
instant messaging, 52
buddy lists/rosters, 86
Groove, 363
hybrid approach to decentralization, 17
Jabber's approach to, 83–88
managing metadata and, 57
pros/cons of traditional services, 80
intermediate peers in transactions, 329–335
Internet
abuse of port 80, 18
anonymity on, 251, 275
breakdown of cooperation, 10–13
changing trust model for downloading software, 244
commercialization of, 8–15
conversational technologies, 77–88
DNS in early days of, 7
dynamic participation on, 275
messaging protocols, 83
network model of, 8–15
peer-to-peer through history of, 3–20
protocol-centric addressing and, 30–32
publishing on, 15
success of HTML, 27
types of conversations on, 78
Internet Relay Chat (IRC), evaluating as a form of conversation, 79
Internet Service Providers (see ISPs)
interoperability
semantics is a deterrent to, 383
through gateways, 381–392
introducers (Free Haven servers), 166, 176, 337
introductory nodes
adding new nodes using, 223
reducing dependence on, 216
Invisible Worlds, 57
IP addresses
dynamic (see dynamic IP addresses)
quota systems based on, 260
IP lists, difficulty of governmental interception of, 142
IPv6, 19, 36
IRC (Internet Relay Chat), evaluating as a form of conversation, 79
ISPs (Internet Service Providers)
asymmetric bandwidth and, 15
blocking by countries, 140
legal status in various countries, 394
logging software downloads, 244
monitoring messages, 256
Usenet and, 6, 51

J

Jabber, xi, 77–88
architecture, 86–88
browsing with, 87
buddy lists/rosters, 86
closed commercial services vs. an open medium, 83
conversation management, 88
infrastructure, 84–86
instant messaging platform, 52
naming components, 85
new approach to instant messaging, 83–88
presence information and, 85
protocol-agnostic design of, 87
supporting namespaces, 84
unique identities for resources, 85
using XML as design basis, 83–88
why it is peer-to-peer, 25
Jakobsson, Markus, 296
Jargon File, 386
Jeong, Hawoong, 222, 225
Juels, Ari, 289, 296
junk mail, 10
as a resource allocation problem, 279
reciprocity of contribution and, 386
using POWs to combat, 288

K

Kaashoek, M. Frans, 167
Kan, Gene, xv, 39, 94–122, 404
k-bit partial hash collisions
client puzzles and, 290
hash cash and, 289

Microsoft Exhibit 1019

keys
certifying, 308
cipher, 364
encryption/decryption, 362–364
Freenet, 128–132
group (see group keys (Groove))
pairwise, 366
adding new members, 372
private (see private keys)
public (see public keys)
Publius, 147, 250
encryption/decryption, 149
reconstructing, 153
revoking, 309
signature/verification, 362–364
Keyword Signed Keys (KSKs), 129
Khare, Rohit, 310
Kimball, Spencer, 101
Kirkpatrick, David, 317
Klein, Sammy, 241
Köhntopp, Marit, 334
Komodo development environment, 45
KSKs (Keyword Signed Keys), 129
Kuvshinov, Andrey, 144
k-way hash function collisions (MicroMint), 294

L

Lampson, Butler, 309
Langley, Adam, xv, 89–93, 123–132, 404
LCS35 Time Capsule Crypto-Puzzle, 290
Leach and Salz, 105
leakage problems
in email shared spaces, 359
in scoring systems, 324–326
legal action, Cox vs. Napster, 35
legal attacks as censorship, 177, 261
legal issues for peer-to-peer, 393–395
Leiserson, Charles, 312
Lendl, Otmar, 144
Lethin, Richard, xvi, 341–353, 406
Levien, Raph, 311, 348
link distribution
in Freenet, 228
in Gnutella, 234
local proxies (Publius), 255
local vs. global credibility (ratings weight), 322
locations and pseudonyms, 163
logging
issues when using Napster, 264
web server, 250–252
Long, Richard, 144
Lysyanskaya, Anna, 187

M

MAC (Message Authentication Code), 364
HMAC-SHA1 default algorithm, 364, 366, 380
pairwise keys, 366
(see also multiauthenticators)
macropayment digital cash schemes, 296–300
reputation and, 297
risks involved for peer-to-peer, 305
macropayments vs. micropayments, 296
Mahn, Nathan, 187
Malkin, Tal, 327
manual databases, organized, 60–62
MARC4 algorithm, 361, 379, 380
Mark Twain Bank, 293
Marshall Aid Commemoration Commission, 241
Martin, David, 175
Mathewson, Nick, 340
maxflow-mincut theorem, 312
Maximum Likelihood Estimation (MLE), 348–350
May, Tim, 162, 343
Mayland, Bryan, 96
Mazieres, David, 167
McBurnett, Neal, 309
MD5 hash function (Publius), 150–155
Meakin, C., 203
mechanical databases, organized, 60
Meike, Blake, 340
meme maps, 39–47
free software, 41–43
open source, 43–45
peer-to-peer
current, 45–47
new, 47–56
Message Authentication Code (see MAC)
message broadcasting, Gnutella, 105
message digest functions, 245
message fanout (Groove), 366–368
<meta> tags, 193
meta-address servers, 31

Microsoft Exhibit 1019

metadata, 191–202
describing resources, 193–195
lessons from the Web, 193–197
MP3 file exchange vs. RDF, 199–201
Napster and, 56
properties, 198
UDDI and management of, 55
unique identifiers and, 198
Metallica and Napster, 393
MicroMint, 294
bread pudding protocol for minting coins, 296
extending peer-to-peer applications, 295
sliding scale for payment, 305
micropayment schemes, 286–305
economic impact of, 301–303
MicroMint, 294
PayWord, 294
positive impact on peer-to-peer, 295
protecting against resource allocation attacks, 286
risks involved for peer-to-peer, 305
sliding scale for payment, 304
vs. reputation systems, 306
micropayments, 273
as way to limit junk mail, 280
encrypting, 333
excessive use has negative effect, 302
exchanging among peers, 329–333
in Free Haven, 333–335
fungible (see fungible micropayments)
managing congestion with, 302
means of preventing attacks, 302
needed to provide accountability, 277
nonfungible (see nonfungible micropayments)
problems for mix networks, 333
purpose of, 277–279
reciprocity of contribution and, 386
useful for peer-to-peer, 300–305
varieties of, 287
vs. macropayments, 296
Milgram, Stanley, 207
Miller, Jeremie, xv, 39, 77–88, 404
Miller, Scott, 39
MIME type (Publius), 157
Minar, Nelson, 3–20, 39, 403
mirroring (see active mirroring)
Mitzenmacher, Michael, 186
mix networks
messages between Publius client/server, 256–258
micropayments not used to address accountability, 335
solving the micropayments issue, 333
timing attacks on, 325
verifying bandwidth allocation, 338
Mixmaster remailers, 89–93, 174
amortized pairwise payment model, 332
exchanging micropayments among peers, 329
impact on accountability, 276
MLE (Maximum Likelihood Estimation), 348–350
moderation system for postings on Slashdot, 310
Moechel, Erich, 144
Moinvaziri, Nathan, 96, 109
Mojo Nation
accountability problem, solving, 279
anonymity properties of, 181, 183
combined with Freenet and Gnutella, 387
distributed computation and file sharing, 52
document-anonymity, 164
freeloaders, tackling problem of, 293
micropayments in, 386
Mojo based on eCash technology, 293
OmniNetwork, what it can add to, 386
routing techniques, 390
trust issues and, 262
vs. Free Haven, 160, 169
Molnar, David, xvi, 144, 159–187, 241, 271–340, 406
Morrison, Phil, 68
Mosaic, 23
Mozilla, 45
MP3 file exchange, 199–201
trust issues and, 264
working around unique identifier problem, 200
multiauthenticators (MAC), 366
fetching lost messages, 369
supporting message fanout, 366–368
multiplayer games, transferring ownership in, 283
music industry and Napster, 32–35, 50, 57

Microsoft Exhibit 1019

mutually-suspicious shared spaces, 365–369
mutually-trusting shared spaces, 362–365

N

namespaces
  of DNS names, 8
  peer-to-peer applications and, 57
  XML
    Jabber support of, 84
    metadata and, 194
Naor, Moni, 288
Napster, x, 21–37, 47–52
  accountability problem and, 278
  automatic "pass-along" participation, 49
  bandwidth
    asymmetric, 16
    using too much, 271
  Cox@Home vs., 35
  disabling via server shut-down, 267
  distributed computation and, 52
  instant messaging and, 52
  legal issues, 117, 393–395
  leveraging the power of PCs, 29
  logging issues and, 264
  metadata and, 56
  Metallica and, 393
  mixing centralization and decentralization, 17, 28, 47
  music industry and, 32–35, 50, 57
  normal use increases database value, 62
  positive externalities and, 33
  predictor of future trends, 27
  protocol-centric addressing, 30–32
  searching issues, 196
  super peers and, 50
  trust issues and, 264
  vs. Gnutella, 97–99
  why it came about, 28, 59
  why it is peer-to-peer, 25
NAT (Network Address Translation), 368
  IPv6 and, 19
Nautilus file manager, 45
negative shilling, 316
Nerdherd web site, 96
.NET, xi, 80, 82
Netscape's parallel download strategy, 12
Network Address Translation (see NAT)
network congestion, 11–13
network model of Internet, 8–15
Network News Transport Protocol (NNTP) and Usenet, 6
networks, private (Gnutella), 115
neural networks and scoring algorithms, 323
New, Darren, 57
Newmarch, Jan, 310
New-Member-Added delta message, 373
news.admin Usenet group, 6, 20
newsgroups (see Usenet)
next-generation Web, evaluating it as a form of conversation, 80
NNTP (Network News Transport Protocol) and Usenet, 6
node failure
  due to random removal
    in Freenet, 225
    in Gnutella, 234–238
  due to targeted attack
    in Freenet, 226
    in Gnutella, 234–238
nodes, remailer (see remailers)
node-specific tickets, 334
nonfungible micropayments, 288–291
  amortized pairwise payment model, 331
  extended types of, 289
  Free Haven works best with, 334
  limitations of, 292
  pairwise payment model and, 331
  POWs and, 286
nonparallelizable work functions, 290
notifications, buddy, 174
Nullsoft, 95
Nutella and Gnutella, 96

O

Odlyzko, A.M., 302
Olson, Mancur, 272
OmniNetwork, 381
  integrating all network types into, 383
  networks and their roles in, 384–388
Onion Routing (mix network), 175, 258
open source
  meme map, 43–45
  peer-to-peer and, 49
  projects and "trust metric", 311–313
  saving Gnutella, 96
  software and trust issues, 248
  summit, 38, 43

Microsoft Exhibit 1019

Oram, Andy, ix–xvii, 393–397
O'Reilly Network's P2P directory, 40
O'Reilly, Tim, xv, 38–58, 403
organized manual databases, 60–62
organized mechanical databases, 60
Ozzie, Ray, 39, 53, 376

P

P in P2P is People, 49
P vs. NC problem, 291
P vs. NP problem, 291
P2P directory (O'Reilly Network), 40
packet delivery, best effort, 11
pairwise keys, 366
  adding new members, 372
pairwise payment model, 330
parallel download strategy (Netscape), 12
parallelizable schemes, 290
partial anonymity, 165
partial hash collisions, 289
  minting MicroMint coins, 296
passive-server document-anonymity, 164
  provided by
    Eternity Usenet, 182
    Free Haven, 182
    Freenet, 183
passwords and Publius Update operations, 155
pathlengths
  characteristic (see characteristic pathlengths)
  query (Gnutella), 233
Pavlovski, Chris, 338
payment models in peer-to-peer systems, 330–333
payment schemes
  anonymous e-cash, 261
  CPU-based, 260
pay-per-use web sites and PayWord, 294
PayWord, 294
  extending peer-to-peer applications, 295
  sliding scale for payment, 305
payword chains (hash values), 294
PCs
  as promiscuous computers, 30
  in the early days of the Internet, 23
  Napster leverages the power of, 29
  underused resources of, 24
peer-to-peer
  accountability problem, solving, 279
  anonymity and, 117–120, 160
  asymmetric bandwidth and, 14
  business battlegrounds for, 57
  certification system, 248
  connectivity issues and, 22–26
  copyright issues and, 393–396
  cost/benefit analysis of, 302
  current applications of, 15–18
  decentralization and, 16–17, 28, 58
  devices and, 55
  directory of projects, 399–401
  distributed computation, 51
  early examples of, xi
  fault tolerance and, 203, 224, 234
  Free Haven's contribution to, 186
  future of, 18–20, 56–58, 121
  governmental threats to privacy, 133–135
  instant messaging and, 52
  managing congestion to prevent bandwidth attacks, 303
  managing metadata and, 57
  meme map
    current, 45–47
    new, 47–56
  micropayments
    economic impact of, 301–303
    exchanging, 329–333
    identity-based, 301
    risks involved, 305
    usefulness of, 300–305
  models, 3–20, 276
  next-generation file sharing technologies, 121
  open source and, 49
  problems with, 3, 28, 274
  protocol-centric addressing, 30–32
  remailer difficulties, 93
  reputation system, 353
  revisionist history of, 4–8
  scalability and, 204
  security provided by Groove, 354–380
  SETI@home and, 75
  social and technical solutions to problems, 18
  social impact of, xii
  summit, 38–41
    meme map developed at, 39, 47–56

Microsoft Exhibit 1019

peer-to-peer (*continued*)
trust issues, 242–270
users donating resources, 292
using accountability to expand, 273
using port 80, 18
using RDF for applications, 199
vs. client/server model, 4, 26, 35, 75, 396
vs. World Wide Web, 191
web site for, **xvii**
what is and what is not, 25
Pepper, Tom, 95
perfect forward anonymity, 181–184
performance, 203–241
decentralized systems sensitive to, 205
degradation
caused by free riders, 206
Gnutella's traffic problems, 112–117
Freenet vs. Gnutella, 240
(see also bandwidth)
Personal Web Server (PWS), 54
person-to-person conversations, 78
PGP (Pretty Good Privacy), 248, 308–310
authenticating invitations via, 369
key certification problem in, 308
key revocation problem, 309
not routinely used to secure email, 358
physical threats as censorship, 177, 261
PICS (Platform for Internet Content Selection), 197
Pieper, Josh, 113
piggy-back rekey information, 374
PING packet type, 115
Pippenger, Nicholas, 291
PKIs (Public Key Infrastructures), 297
alternative to web of trust, 309
Groove and, 369–371
lack of, 270
pseudospoofing and, 281
plaintext messages, 149
Platform for Internet Content Selection (PICS), 197
pooling approach for ratings, 325
Popular Power, 4, 25
port 80, abuse of, 17
portable reputations, 351
positive externalities and Napster, 33
positive shilling, 316
POWs (proofs of work)
combatting junk mail with, 288
exchanging for other resources, 295
handling resource allocation requests, 288
hash cash and, 289
preventing communication DoS attacks, 286
with trap doors, 289
preferential attachment and nodes, 229
presence information and Jabber, 85
Pretty Good Privacy (see PGP)
"pricing via processing" and resource allocation requests, 288
privacy
in Brands's digital cash scheme, 299
distributed systems and, 275
in data-sharing systems, 160–163
protecting, 251
scoring systems and, 324–326
private Gnutella networks, 115
private keys, 246
defining Groove identities, 362–364
identifying authors of files, 246
mix networks and, 257
reputation systems and, 308
Signature Verification Keys (SVKs), 130
trustworthiness, 248
probabilistic checking, 297
product-of-two-primes problem, 291
Project Anon, 334
proof of transaction and fraud prevention, 317
"proofs of time", 290
proofs of work (see POWs)
properties and metadata, 198
proto-coins, 297
protocol-centric addressing, 30–32
protocols
friendly vs. unfriendly, 11–13
Internet messaging, 83
Jabber's transparent use of, 87
proxies
anonymizing, 251, 256
local/remote, 255
pseudoanonymity and Gnutella, 118–120
pseudonyms, 275
consequences of using, 280–283
digital cash and, 287
Free Haven use of, 162, 166
on Freenet, 130
pseudospoofing and, 280–282
sold or stolen, 282

Microsoft Exhibit 1019

pseudospoofing
possible defenses against, 281
problems with, 280
resistance by Advogato, 312
public key cryptography, 246
PGP and, 308–310
Public Key Infrastructures (see PKIs)
public keys, 246
defining Groove identities, 362–364
Free Haven servers and, 167–169
Keyword Signed Keys (KSKs) and, 130
macropayment digital cash schemes, 296
mix networks and, 257
remailers and, 90
Signature Verification Keys (SVKs), 130
verifying using digital certificates, 246
public/private key pairs, 362
Diffie-Hellman, 365, 372, 379
Publish operation (Publius), 150–152
publisher-anonymity, 163
attacks on, 181
enhanced by trading, 170
Eternity Usenet and, 182
Free Haven and, 182
Freenet and, 183
Gnutella and, 182
Mojo Nation and, 183
Publius and, 183
publisher-centric systems, 387
publishing on the Internet, 15
publishing systems
anonymous (see anonymous, publishing systems)
trust in censorship-resistant, 249–256
Publius, 145–158, 249–261
accountability problem, solving, 279
anonymity properties of, 181, 183
censorship in, 253–261
censorship resistance in, 145–158, 249
CGI script, 156
client software, 146, 150, 250
implementation, 156
local vs. remote proxies, 255
publishing documents, 147
cryptography fundamentals, 149
Delete operation, 155
denial of service attacks, resisting, 259–261
distributed denial of service attacks, resisting, 145, 158
distributed search engines and, 268
documents
deleting/updating, 147
encrypting using keys, 147
protecting against censorship, 253–255
revoking, 169
encrypted files jeopardized by search engines, 268
hash functions, 150
how it works, 147–149
HTTP proxy (see HTTP proxy)
implemented over HTTP, 147
MIME type, 157
OmniNetwork, what it can add to, 387
passwords and Update operations, 155
Publish operation, 150–152
Retrieve operation, 152–154
routing techniques, 390
search engines, problems with, 267
secret sharing algorithms, 149
server software, 146, 150, 156
shares, 148, 250
protecting against update attacks, 254
retrieving, 153
splitting the key into, 150–152
trust issues in, 249–261
Update operation, 154
censoring documents, 253–255
user interface, 156
vs. Free Haven, 160
vs. Freenet, 147
vs. Gnutella, 147
web site for, 158
why it is important, 146
Publius Server List, 147
Publish operation and, 150–152
Retrieve operation and, 152
Publius Servers, 146, 250
Publius URL, 148, 250
format of, 151
protecting against update attacks, 254
retrieving a, 152–154
tamper-check mechanism, 147, 153
updating, 154
pulses (signals), 71
PUSH REQUEST packet type, 115
puzzles
client (see client puzzles)
time-lock (see time-lock puzzles)

Microsoft Exhibit 1019

PWS (Personal Web Server), 54
Pyra Labs, 54

Q

QUERY packet type, 115
query pathlengths in Gnutella, 233
query-anonymity, 164
Quintessenz, 144
quota systems based on IP addresses, 260

R

Rabin, Michael O., 168
Rabin, Tal, 338
radio frequency interference (RFI), 71
radio SETI, 68
random graphs, 213
ratings
  collecting, 318
  of eBay trades, 311
  fairness of, 319
  identification flooding attacks and, 326
  incentives for contributing, 319
  local vs. global credibility, 322
  pooling approach for, 325
  proof of transaction validates, 317
  protecting against timing attacks, 325
  pruning bad, 321
  of Slashdot postings, 310
  submitting fake or misleading, 315
  submitting to mutually agreeable scorers, 327
  weighting, using credibility of raters, 321
Raymond, Jean-François, 187, 340
RDF (Resource Description Framework), 197–199
  unique identifiers and, 198
  vs. MP3 file exchange, 199–201
reader-anonymity, 164
  attacks on, 180
  Eternity Usenet and, 182
  Free Haven and, 182
  Freenet and, 183
  Gnutella and, 182
  Mojo Nation and, 183
  Publius and, 183
reader-centric systems, 387
receipts and trading shares with Free Haven, 171–173
  attacks on, 180
recipient-hiding cryptosystems, 333
reciprocity of contribution, 386
Red Hat Linux, 43
Red Rover, 133–144
  architecture of, 135–139
  clients, 137
  enticements to running a client, 143
  hub, 136
    tracking subscriber information, 137
  making use of Internet liabilities, 135
  origin of name, 143
  putting security weaknesses into perspective, 140–143
  subscribers, 138
  why those who most need it can least use it, 133
redeemable payments, 292
redirects and keys (Freenet), 131
Reed, M.G., 175
Reflectors (Clip2.com, Inc.), 50, 112, 116
regular graphs, 212–214
  Freenet case study, 217
Reiter, Michael, 348
Reive, Catherine, 241
rekey delta messages, 374
relay servers for peer communication, 366–368
reliability with anonymity, 161–163
remailers, 89–93, 174
  impact on accountability, 276
remote proxies (Publius), 255
replenishable resources, 32
replicable resources, 32
reply blocks, Free Haven, 167
reputation, 274, 306–328, 341–353
  automatic handling via "web of trust", 308–310
  calculating using a Bayesian approach, 348–350
  can be sold or stolen, 282
  codifying with "web of trust", 307–310
  default score is important, 320
  Free Haven, 175, 335–337
    attacks on the system, 179
    broadcasting referrals, 336
    pseudonyms and, 162
  good vs. bad, determining, 318

Microsoft Exhibit 1019

identity as an element of, 345
key certification problem, 308
macropayment digital cash schemes and, 297
meta-reputation problem, 327
metrics, 347–350
portable, 351
real world vs. Internet, 341
reducing risk in transactions, 343
referrals, 336
risking by trying out new nodes, 319
separating into categories to defend against shilling, 323
tracking mechanisms, 20
trust and, 242
using statistical models of behavior to calculate, 348–350
reputation domains, 343–345
weak vs. strong entities, 345
Reputation Server, 342–353
auction sites, supporting, 351
benchmark sources, 351
bootstrapping, obstacle to, 351
buyers and sellers benefit from, 342
centralized vs. distributed, 353
communicating with the marketplace, 345
credibility measures for sources, 350
domains, 343–345
interdomain sharing, 351
needs to know identity of entities, 345
reducing risk with, 343
references to bootstrap reputations, 352
reputation metrics, 347–350
scoring algorithms, 350
scoring system, 347
identity properties influence, 345
soliciting feedback from parties in transactions, 342, 346
reputation systems
bootstrapping, 319
collecting ratings, 318
creating new identities, 320
early online, 307–313
evaluating the security of, 315
Free Haven, 335–337
needed to provide accountability, 277
observing transaction flow, 318
open source development, 311–313
partially-automated (Slashdot), 310
personalizing reputation searches, 320
problems with, 313
pseudospoofing, bad loophole in, 280
purpose of, 277–279
scores and ratings, 314–328
usefulness, 306
vs. micropayment schemes, 306
Reputation Technologies, Inc.
potential adversaries of, 315
providing bootstrap information, 319
requests
Free Haven, 390
Freenet, 124–128, 389
Gnutella, 389
Mojo Nation, 390
Publius, 390
resource allocation
bandwidth usage and, 271
centralized control and, 272
freeloading and, 292
junk mail and, 279
limiting congestion using micropayments, 302
problems and peer-to-peer, 340
protecting against attacks, 286
using POWs to handle requests, 288
Resource Description Framework (see RDF)
resources
describing using RDF, 197–199
peer-to-peer taking advantage of, 22
properties of, 198
replicable/replenishable, 32
restricting access, 273
restrictive blind signature protocols, 298
Retrieve operation (Publius), 152–154
retrieving documents in Free Haven, 168
revoking
documents (Free Haven), 169
keys (web of trust), 309
Rewebber.de, 252
RFI (radio frequency interference), 71
Rich Site Summary (RSS) and content syndication, 55
Rifkin, Adam, 310
Rijndael (Advanced Encryption Standard), 149
Rivest, Ronald, 186, 291, 294, 309, 312
Rose, Marshall, 39
rosters (instant messaging services), 86
routing, message-based (Gnutella), 104

Microsoft Exhibit 1019

routing system (Freenet), 126–128
RSA Labs, 289
RSA/Blowfish algorithm, 361
RSS (Rich Site Summary) and content syndication, 55
Rubin, Avi, xvi, 145–158, 242–270, 405
Ruzzo, Walter, 291

S

sandboxing, 249
Saudi Arabia and legal status of ISPs, 394
scalability, 204
  Freenet and, 229–231
  Gnutella and, 239
  web of trust and, 309
scale-free link distributions, 228
Schmidt, Bob, 109, 113
Schmidt, Eric, 49
Schneier, Bruce, 149, 357
scoring algorithms, 321–324
  factoring credibility into, 350
  Free Haven, 336
  incorporating credibility into, 322
  Reputation Server, 347–350
  treating reputations as probabilities, 323
  vs. privacy of transaction data, 325
scoring postings/posters on Slashdot, 310
scoring systems, 314–328
  adversarial approach to, 324
  aspects of, 316
  attacks against, 315
  bootstrapping, 319
  collecting ratings, 318
  confidence values and, 320
  decentralizing, 326–328
  default reputation score, 320
  detecting suspicious behavior, 321
  importance of user interface in, 320
  meta-reputation problem, 327
  privacy/information leaks, 324–326
  qualitative vs. quantitative information, 321
  qualities of good, 314
  Reputation Server, 345, 347
  separating performance and credibility, 322
  timing attacks against privacy of, 325
  true decentralization, 327
  using multiple trusted parties, 326
  using transactions to calculate ratings, 316
  weighting a score, 337
Scott, Tracy, 101
SDMI (Secure Digital Music Initiative), 34
SDSI/SPKI (Simple Distributed Security Infrastructure/Simple Public Key Infrastructure), 309, 370
search algorithms and centralization, 17
search engines
  centralized vs. distributed, 266–268
  problems with, 196
  trust issues and, 266–269
secret sharing algorithms, 164
  Publius, 149
Secure Digital Music Initiative (SDMI), 34
Secure Sockets Layer (see SSL)
security, 354–380
  centralizing crucial aspects of, 376–378
  collaboration and, 357
  email and, 356
  granular, distributed at workgroup level, 357
  guarantees made by Groove, 356
  Red Rover weaknesses and, 140–143
  of reputation systems, 315
selecting favored users, 274
Semantic Web, 197
SERENDIP project, 68
server deniability, 164
server software, Publius, 146, 150, 156
server volunteers, 259, 262, 265
server-anonymity, 164
  attacks on, 180
  Eternity Usenet and, 182
  Free Haven and, 182
  Freenet and, 183
  Gnutella and, 182
  Publius and, 183
servers
  dormant, 176
  web (see web servers)
servnet, Free Haven (see Free Haven, servnet)
SETI@home, x, 51, 67–76
  client program of, 70
  computations, 69
  data distribution server, 70
  detecting signals, 71
  difficulties and challenges, 72

Microsoft Exhibit 1019

drift rates and, 71
floating-point operations and, 74
how it began, 67–69
how it works, 69–72
peer-to-peer paradigm and, 75
processor-specific optimizations, 73
security challenges, 73
SERENDIP and, 69
server performance problems, 72
user interest in, 73
world's most powerful computer, 74
SHA-1 hash function (Publius), 150
Shamir, Adi, 291, 294
Shamir's secret sharing algorithm (Publius), 149, 327
shared databases, 60–63
shared spaces
cryptographic algorithms, 361
email, 356
weaknesses of, 358–360
forming, 369–371
Groove, 354, 360
inviting people into, 372
joining or leaving, 363
mutually-suspicious, 365–369
mutually-trusting, 362–365
projecting identities into, 362
security characteristics of, 360–362
using a relay server for peer communication, 367
shares
Free Haven
expiration dates of, 169
splitting files into, 167
trading, 166, 170–172
Publius, 148, 250
protecting against update attacks, 254
retrieving, 153
splitting the key into, 150–152
shilling, 315
categorizing reputations to defend against, 323
tying feedback to transactions to avoid, 318
Shirky, Clay, xiv, 21–37, 403
Shostack, Adam, 175
signals, detecting (SETI@home), 71
Signature Verification Keys (SVKs), 130
signatures, digital (see digital signatures)
signature/verification key pairs, 363, 379
Simon and Rackoff, 185
Simple Object Access Protocol (SOAP) and web services, 54
Simple Symmetric Transport Protocol (SSTP), 367
simulating
behavior over time
Freenet, 217–222
Gnutella, 232–234
fault tolerance
in a Freenet network, 224–227
in Gnutella, 234–238
growth, in a Freenet network, 222–224
Sipser, Michael, 249
Slashdot effect, 123
protecting against, 285
resource allocation and, 340
Slashdot moderation system, 310
vs. Advogato trust metric, 311
small-world effect, 207
achieving performance gains, 218
finding short routes between vertices, 221
fundamental to Freenet, 215
random node removal and, 226
web of trust and, 309
small-world graphs, 214
scaling properties of, 229–231
S/MIME
authenticating invitations via, 369
vs. Groove, 359
Smirnov, Sergei, 144
Smith, Adam, 187
Sniffen, Brian, 179, 187
sniffers and compromised internal networks, 358
SOAP (Simple Object Access Protocol) and web services, 54
social inefficiencies
data loss due to insufficient short-term storage, 304
due to excessive micropayments, 302
social networks
computer networks and, 208
graph theory and, 207
understanding link distribution in, 211
World Wide Web and, 208
social pressure as censorship, 177
SOCKS protocol, 19
Sokol-Margolis, Joseph, 187

Microsoft Exhibit 1019

spam (see junk mail)
spent-coin database, 288
spikes (signals), 71
SSL (Secure Sockets Layer)
  encrypting traffic between Publius client/server, 256
  hub/client relationship in Red Rover, 140
  protecting online shopping, 359
SSTP (Simple Symmetric Transport Protocol), 367
stack model (Freenet), 126–128
statistical models of behavior, using to calculate reputation, 348–350
storage flooding attacks (see flooding attacks)
streaming media applications and network congestion, 12
Strogatz, Steven, 211–215
Stubblebine, Stuart, 348
Stutz, Dave, 39
subpuzzles, 290, 304
subscribers (Red Rover), 138
  risks involved in becoming, 141
Sullivan, Woody, 67
Sun Microsystems and revocation of browser certificates, 309
super peers
  Gnutella and, 50, 240
  Napster and, 50
SVKs (Signature Verification Keys), 130
symmetric algorithm, default, 361
symmetric ciphers, 149
symmetric keys
  encrypting
    disk files, 364
    instant messages, 363, 379
  one per account, 379
  pairwise, 365
synchronizing data, 355, 360
Syverson, P.F., 175

T

tamper-check mechanism (Publius URL), 147, 153, 250
tampering with files, 245
TCP rate equation, 11–13
TCP-based broadcasts (Gnutella), 104
technology
  copyright infringement issues, 393–396
  giving users maximum control, 395
Thawte's Freemail Web of Trust, 309
Thompson, T., 203
Tiemann, Michael, 39
Time Capsule Crypto-Puzzle (LCS35), 290
time-lock puzzles, 291
  sliding scale for payment, 305
timing attacks against privacy of scoring systems, 325
<title> element, 193
traceroutes, performing, 210
trading
  handshakes (Free Haven), 171
  shares (Free Haven), 166, 170–172
traffic analysis
  of email messages, 92
  of Freenet network, 125
  using mix networks to thwart, 257
tragedy of the commons, 49
  accountability and, 272
  junk mail as example of, 280
  performance degradation and, 206
  reverse of, 307
  vs. cornucopia of the commons, 63
transactions
  between the marketplace and the Reputation Server, 346
  blinded receipts used for, 326
  calculating ratings using, 316
  hiding transaction data for, 325
  intermediate peers in, 329–335
  monitored by Reputation Server, 342–352
  reducing vulnerability by tying feedback to, 318
  using reputation to reduce risk, 343
  verifying, as fraud prevention, 317
  weighting categories of, 323
transceiver, Groove, 360
transient connectivity, 25
transitive paths in graphs, 210
  Freenet case study, 216
transmission loss over TCP (Gnutella), 107
Trellix, 54
triplets (signals), 71
true names vs. pseudonyms, 162
trust, 242–270
  bottlenecks, 313
  in the physical world, 242–244
  reputation and, 242, 311–313
  risk assessment and, 243
  ways to increase, 242–244

Microsoft Exhibit 1019

trust issues
censorship-resistant publishing systems and, 249–256
downloaded software and, 244–249
Eternity Service and, 262
file-sharing systems and, 263–265
Free Haven and, 262, 338
Freenet and, 265
Gnutella and, 264
Mojo Nation and, 262
Napster and, 264
open source software and, 248
Publius and, 249–256
search engines and, 266–269
trust levels and PGP, 308
trusted third parties, 246
TTL (time-to-live) numbers (Gnutella), 105
Tuvell, Walter, xvii, 354–380, 407
Two-Way Web, 80
Tygar, J.D., 185
Type 1 remailers, 89
weaknesses of, 92
Type 2 remailers, 89–93
vs. Type 1 remailers, 92

U

ÜberClient, pros and cons of, 382
ÜberNetwork
problems with, 382
unifying file-sharing systems, 381
Udell, Jon, xvii, 55, 354–380, 406
Ultima Online game identities, selling, 283
Uniform Resource Indicators (see URIs)
unique identifiers and metadata, 198
United States and legal status of ISPs, 394
Universal Description, Discovery, and Integration (UDDI), 55
unsolicited email (see junk mail)
Update operation (Publius), 154
censoring documents with, 253–255
upload/download ratios on bulletin boards, 307
URIs (Uniform Resource Indicators)
Freenet, 129
metadata and, 198
URL, Publius (see Publius URL)
Usenet, 5–7
centralization and decentralization, 50
Eternity Usenet and, 263
evolution of decentralized system, 16
integrating local and global control, 20
remailers and, 89–93
users
adding value to systems, 59–63
adopting peer-to-peer applications, 26
filling shared databases, 60–63
intense interest in SETI@home, 73
maximizing contributions from, 59–63
preferring HTML over CD-ROMs, 27
refusing to accept per-unit prices, 34
UUCP (Unix-to-Unix-copy protocol), 5
UUCPnet, 50
UUIDs of messages (Gnutella), 105
UUnet, 51

V

verifying
bandwidth allocation, 338
coins, 300
digital signatures, 247
downloaded files, 245
transactions, 317
VeriSign (certificate authority), 309, 359
Vinge, Vernor, 162
VIP Reference (U.S.), 144
virtual private networks (VPNs), extending security off-site, 358
volunteer databases, 60–63
volunteers, server, 259, 262, 265

W

W3C (World Wide Web Consortium) and metadata vision, 197
Wagner, David, 291
Waldman, Marc, xvi, 145–158, 242–270, 340, 405
Wall of Shame (Zeropaid.com), 118, 265
Watts, Duncan, 211–215
web access, Publius (see HTTP proxy)
web browsers
as ÜberClient, 382
asymmetry between web servers and, 53
caching pages, 284
protocol-centric namespaces and, 31
web of trust, 248, 308–310
key revocation problem, 309
overtaken by hierarchical PKIs, 313

Microsoft Exhibit 1019

web of trust (*continued*)
PKIs as alternative to, 309
scalability question, 309
web servers
Akamai, 284
asymmetry between web browsers and, 54
Publius Servers, 146, 250
risks involved in logging, 250–252
web services, 54
web sites
blocking, 133–135
using IP addresses, 139
for Free Software Foundation (FSF), 41
for peer-to-peer, xvii
for Publius, 158
Website web server, 54
Werbach, Kevin, 63
Werthimer, Dan, 67
Whitten, Alma, 185
Wiki, 54
Wiley, Brandon, xvii, 381–392, 407
Winamp, 95
Winer, Dave, 49, 80
EditThisPage.Com, 54
Woolsridge, Brett, 187
World Free Web (WFW) project, 392
World Wide Web
abuse of port 80, 17
anonymizing proxies and, 251
as a form of conversation, 79
as small-world domain, 221
brief history of, 81
combined with Freenet, 392
conversational technologies, 77–88
data models for, 29
early days of, 395
finding data on, 192
hyperlinks on, 53
lessons for metadata, 193–197
problems with searching, 196
social networks and, 208
vs. peer-to-peer, 191
World Wide Web Consortium (W3C) and metadata vision, 197
wrappers (limiting behavior of CGI scripts), 249
writable Web, 53
Wurster, Thomas, 352

X

XML
creating interoperability between existing networks, 383
Dublin Core metadata in documents, 195
namespaces supported by Jabber, 84
used by Jabber as design basis, 83–88

Y

Yang, Yinan, 310
Young, Bob, 43

Z

Zeithammer, Robert, 340
Zephyr chat, 121
Zero-Knowledge Systems, Inc., 298
Zeropaid.com, 118
Zimmermann, Phil, 308

Microsoft Exhibit 1019

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Andy Oram has been writing and editing books on programming and networking for O'Reilly & Associates since 1992. He created the O'Reilly Linux series long before Linux became mainstream and has spent considerable time researching the social impacts of computing and networking. He and his family live near Boston, Massachusetts.

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January 1996

RTP: A Transport Protocol for Real-Time Applications

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

Abstract

This memorandum describes RTP, the real-time transport protocol. RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data, over multicast or unicast network services. RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (RTCP) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and mixers.

Table of Contents

1. Introduction 3
2. RTP Use Scenarios 5
2.1 Simple Multicast Audio Conference 5
2.2 Audio and Video Conference 6
2.3 Mixers and Translators 6
3. Definitions 7
4. Byte Order, Alignment, and Time Format 9
5. RTP Data Transfer Protocol 10
5.1 RTP Fixed Header Fields 10
5.2 Multiplexing RTP Sessions 13

Microsoft Exhibit 1019

5.3 Profile-Specific Modifications to the RTP Header..... 14
5.3.1 RTP Header Extension 14
6. RTP Control Protocol -- RTCP 15
6.1 RTCP Packet Format 17
6.2 RTCP Transmission Interval 19
6.2.1 Maintaining the number of session members 21
6.2.2 Allocation of source description bandwidth 21
6.3 Sender and Receiver Reports 22
6.3.1 SR: Sender report RTCP packet 23
6.3.2 RR: Receiver report RTCP packet 28
6.3.3 Extending the sender and receiver reports 29
6.3.4 Analyzing sender and receiver reports 29
6.4 SDES: Source description RTCP packet 31
6.4.1 CNAME: Canonical end-point identifier SDES item 32
6.4.2 NAME: User name SDES item 34
6.4.3 EMAIL: Electronic mail address SDES item 34
6.4.4 PHONE: Phone number SDES item 34
6.4.5 LOC: Geographic user location SDES item 35
6.4.6 TOOL: Application or tool name SDES item 35
6.4.7 NOTE: Notice/status SDES item 35
6.4.8 PRIV: Private extensions SDES item 36
6.5 BYE: Goodbye RTCP packet 37
6.6 APP: Application-defined RTCP packet 38
7. RTP Translators and Mixers 39
7.1 General Description 39
7.2 RTCP Processing in Translators 41
7.3 RTCP Processing in Mixers 43
7.4 Cascaded Mixers 44
8. SSRC Identifier Allocation and Use 44
8.1 Probability of Collision 44
8.2 Collision Resolution and Loop Detection 45
9. Security 49
9.1 Confidentiality 49
9.2 Authentication and Message Integrity 50
10. RTP over Network and Transport Protocols 51
11. Summary of Protocol Constants 51
11.1 RTCP packet types 52
11.2 SDES types 52
12. RTP Profiles and Payload Format Specifications 53
A. Algorithms 56
A.1 RTP Data Header Validity Checks 59
A.2 RTCP Header Validity Checks 63
A.3 Determining the Number of RTP Packets Expected and Lost 63
A.4 Generating SDES RTCP Packets 64
A.5 Parsing RTCP SDES Packets 65
A.6 Generating a Random 32-bit Identifier 66
A.7 Computing the RTCP Transmission Interval 68

Microsoft Exhibit 1019

A.8 Estimating the Interarrival Jitter 71
B. Security Considerations 72
C. Addresses of Authors 72
D. Bibliography 73

1. Introduction

This memorandum specifies the real-time transport protocol (RTP), which provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video. Those services include payload type identification, sequence numbering, timestamping and delivery monitoring. Applications typically run RTP on top of UDP to make use of its multiplexing and checksum services; both protocols contribute parts of the transport protocol functionality. However, RTP may be used with other suitable underlying network or transport protocols (see Section 10). RTP supports data transfer to multiple destinations using multicast distribution if provided by the underlying network.

Note that RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantees, but relies on lower-layer services to do so. It does not guarantee delivery or prevent out-of-order delivery, nor does it assume that the underlying network is reliable and delivers packets in sequence. The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence, but sequence numbers might also be used to determine the proper location of a packet, for example in video decoding, without necessarily decoding packets in sequence.

While RTP is primarily designed to satisfy the needs of multi-participant multimedia conferences, it is not limited to that particular application. Storage of continuous data, interactive distributed simulation, active badge, and control and measurement applications may also find RTP applicable.

This document defines RTP, consisting of two closely-linked parts:

- the real-time transport protocol (RTP), to carry data that has real-time properties.

- the RTP control protocol (RTCP), to monitor the quality of service and to convey information about the participants in an on-going session. The latter aspect of RTCP may be sufficient for "loosely controlled" sessions, i.e., where there is no explicit membership control and set-up, but it is not necessarily intended to support all of an application's control communication requirements. This functionality may be fully or partially subsumed by a separate session control protocol,

Microsoft Exhibit 1019

which is beyond the scope of this document.

RTP represents a new style of protocol following the principles of application level framing and integrated layer processing proposed by Clark and Tennenhouse [1]. That is, RTP is intended to be malleable to provide the information required by a particular application and will often be integrated into the application processing rather than being implemented as a separate layer. RTP is a protocol framework that is deliberately not complete. This document specifies those functions expected to be common across all the applications for which RTP would be appropriate. Unlike conventional protocols in which additional functions might be accommodated by making the protocol more general or by adding an option mechanism that would require parsing, RTP is intended to be tailored through modifications and/or additions to the headers as needed. Examples are given in Sections 5.3 and 6.3.3.

Therefore, in addition to this document, a complete specification of RTP for a particular application will require one or more companion documents (see Section 12):

- a profile specification document, which defines a set of payload type codes and their mapping to payload formats (e.g., media encodings). A profile may also define extensions or modifications to RTP that are specific to a particular class of applications. Typically an application will operate under only one profile. A profile for audio and video data may be found in the companion RFC TBD.

- payload format specification documents, which define how a particular payload, such as an audio or video encoding, is to be carried in RTP.

A discussion of real-time services and algorithms for their implementation as well as background discussion on some of the RTP design decisions can be found in [2].

Several RTP applications, both experimental and commercial, have already been implemented from draft specifications. These applications include audio and video tools along with diagnostic tools such as traffic monitors. Users of these tools number in the thousands. However, the current Internet cannot yet support the full potential demand for real-time services. High-bandwidth services using RTP, such as video, can potentially seriously degrade the quality of service of other network services. Thus, implementors should take appropriate precautions to limit accidental bandwidth usage. Application documentation should clearly outline the limitations and possible operational impact of high-bandwidth real-

Microsoft Exhibit 1019

time services on the Internet and other network services.

2. RTP Use Scenarios

The following sections describe some aspects of the use of RTP. The examples were chosen to illustrate the basic operation of applications using RTP, not to limit what RTP may be used for. In these examples, RTP is carried on top of IP and UDP, and follows the conventions established by the profile for audio and video specified in the companion Internet-Draft draft-ietf-avt-profile

2.1 Simple Multicast Audio Conference

A working group of the IETF meets to discuss the latest protocol draft, using the IP multicast services of the Internet for voice communications. Through some allocation mechanism the working group chair obtains a multicast group address and pair of ports. One port is used for audio data, and the other is used for control (RTCP) packets. This address and port information is distributed to the intended participants. If privacy is desired, the data and control packets may be encrypted as specified in Section 9.1, in which case an encryption key must also be generated and distributed. The exact details of these allocation and distribution mechanisms are beyond the scope of RTP.

The audio conferencing application used by each conference participant sends audio data in small chunks of, say, 20 ms duration. Each chunk of audio data is preceded by an RTP header; RTP header and data are in turn contained in a UDP packet. The RTP header indicates what type of audio encoding (such as PCM, ADPCM or LPC) is contained in each packet so that senders can change the encoding during a conference, for example, to accommodate a new participant that is connected through a low-bandwidth link or react to indications of network congestion.

The Internet, like other packet networks, occasionally loses and reorders packets and delays them by variable amounts of time. To cope with these impairments, the RTP header contains timing information and a sequence number that allow the receivers to reconstruct the timing produced by the source, so that in this example, chunks of audio are contiguously played out the speaker every 20 ms. This timing reconstruction is performed separately for each source of RTP packets in the conference. The sequence number can also be used by the receiver to estimate how many packets are being lost.

Since members of the working group join and leave during the conference, it is useful to know who is participating at any moment and how well they are receiving the audio data. For that purpose,

Microsoft Exhibit 1019

each instance of the audio application in the conference periodically multicasts a reception report plus the name of its user on the RTCP (control) port. The reception report indicates how well the current speaker is being received and may be used to control adaptive encodings. In addition to the user name, other identifying information may also be included subject to control bandwidth limits. A site sends the RTCP BYE packet (Section 6.5) when it leaves the conference.

2.2 Audio and Video Conference

If both audio and video media are used in a conference, they are transmitted as separate RTP sessions RTCP packets are transmitted for each medium using two different UDP port pairs and/or multicast addresses. There is no direct coupling at the RTP level between the audio and video sessions, except that a user participating in both sessions should use the same distinguished (canonical) name in the RTCP packets for both so that the sessions can be associated.

One motivation for this separation is to allow some participants in the conference to receive only one medium if they choose. Further explanation is given in Section 5.2. Despite the separation, synchronized playback of a source’s audio and video can be achieved using timing information carried in the RTCP packets for both sessions.

2.3 Mixers and Translators

So far, we have assumed that all sites want to receive media data in the same format. However, this may not always be appropriate. Consider the case where participants in one area are connected through a low-speed link to the majority of the conference participants who enjoy high-speed network access. Instead of forcing everyone to use a lower-bandwidth, reduced-quality audio encoding, an RTP-level relay called a mixer may be placed near the low-bandwidth area. This mixer resynchronizes incoming audio packets to reconstruct the constant 20 ms spacing generated by the sender, mixes these reconstructed audio streams into a single stream, translates the audio encoding to a lower-bandwidth one and forwards the lower-bandwidth packet stream across the low-speed link. These packets might be unicast to a single recipient or multicast on a different address to multiple recipients. The RTP header includes a means for mixers to identify the sources that contributed to a mixed packet so that correct talker indication can be provided at the receivers.

Some of the intended participants in the audio conference may be connected with high bandwidth links but might not be directly reachable via IP multicast. For example, they might be behind an

Microsoft Exhibit 1019

application-level firewall that will not let any IP packets pass. For these sites, mixing may not be necessary, in which case another type of RTP-level relay called a translator may be used. Two translators are installed, one on either side of the firewall, with the outside one funneling all multicast packets received through a secure connection to the translator inside the firewall. The translator inside the firewall sends them again as multicast packets to a multicast group restricted to the site's internal network.

Mixers and translators may be designed for a variety of purposes. An example is a video mixer that scales the images of individual people in separate video streams and composites them into one video stream to simulate a group scene. Other examples of translation include the connection of a group of hosts speaking only IP/UDP to a group of hosts that understand only ST-II, or the packet-by-packet encoding translation of video streams from individual sources without resynchronization or mixing. Details of the operation of mixers and translators are given in Section 7.

3. Definitions

RTP payload: The data transported by RTP in a packet, for example audio samples or compressed video data. The payload format and interpretation are beyond the scope of this document.

RTP packet: A data packet consisting of the fixed RTP header, a possibly empty list of contributing sources (see below), and the payload data. Some underlying protocols may require an encapsulation of the RTP packet to be defined. Typically one packet of the underlying protocol contains a single RTP packet, but several RTP packets may be contained if permitted by the encapsulation method (see Section 10).

RTCP packet: A control packet consisting of a fixed header part similar to that of RTP data packets, followed by structured elements that vary depending upon the RTCP packet type. The formats are defined in Section 6. Typically, multiple RTCP packets are sent together as a compound RTCP packet in a single packet of the underlying protocol; this is enabled by the length field in the fixed header of each RTCP packet.

Port: The "abstraction that transport protocols use to distinguish among multiple destinations within a given host computer. TCP/IP protocols identify ports using small positive integers." [3] The transport selectors (TSEL) used by the OSI transport layer are equivalent to ports. RTP depends upon the lower-layer protocol to provide some mechanism such as ports to multiplex the RTP and RTCP packets of a session.

Microsoft Exhibit 1019

Transport address: The combination of a network address and port that identifies a transport-level endpoint, for example an IP address and a UDP port. Packets are transmitted from a source transport address to a destination transport address.

RTP session: The association among a set of participants communicating with RTP. For each participant, the session is defined by a particular pair of destination transport addresses (one network address plus a port pair for RTP and RTCP). The destination transport address pair may be common for all participants, as in the case of IP multicast, or may be different for each, as in the case of individual unicast network addresses plus a common port pair. In a multimedia session, each medium is carried in a separate RTP session with its own RTCP packets. The multiple RTP sessions are distinguished by different port number pairs and/or different multicast addresses.

Synchronization source (SSRC): The source of a stream of RTP packets, identified by a 32-bit numeric SSRC identifier carried in the RTP header so as not to be dependent upon the network address. All packets from a synchronization source form part of the same timing and sequence number space, so a receiver groups packets by synchronization source for playback. Examples of synchronization sources include the sender of a stream of packets derived from a signal source such as a microphone or a camera, or an RTP mixer (see below). A synchronization source may change its data format, e.g., audio encoding, over time. The SSRC identifier is a randomly chosen value meant to be globally unique within a particular RTP session (see Section 8). A participant need not use the same SSRC identifier for all the RTP sessions in a multimedia session; the binding of the SSRC identifiers is provided through RTCP (see Section 6.4.1). If a participant generates multiple streams in one RTP session, for example from separate video cameras, each must be identified as a different SSRC.

Contributing source (CSRC): A source of a stream of RTP packets that has contributed to the combined stream produced by an RTP mixer (see below). The mixer inserts a list of the SSRC identifiers of the sources that contributed to the generation of a particular packet into the RTP header of that packet. This list is called the CSRC list. An example application is audio conferencing where a mixer indicates all the talkers whose speech was combined to produce the outgoing packet, allowing the receiver to indicate the current talker, even though all the audio packets contain the same SSRC identifier (that of the mixer).

Microsoft Exhibit 1019

End system: An application that generates the content to be sent in RTP packets and/or consumes the content of received RTP packets. An end system can act as one or more synchronization sources in a particular RTP session, but typically only one.

Mixer: An intermediate system that receives RTP packets from one or more sources, possibly changes the data format, combines the packets in some manner and then forwards a new RTP packet. Since the timing among multiple input sources will not generally be synchronized, the mixer will make timing adjustments among the streams and generate its own timing for the combined stream. Thus, all data packets originating from a mixer will be identified as having the mixer as their synchronization source.

Translator: An intermediate system that forwards RTP packets with their synchronization source identifier intact. Examples of translators include devices that convert encodings without mixing, replicators from multicast to unicast, and application-level filters in firewalls.

Monitor: An application that receives RTCP packets sent by participants in an RTP session, in particular the reception reports, and estimates the current quality of service for distribution monitoring, fault diagnosis and long-term statistics. The monitor function is likely to be built into the application(s) participating in the session, but may also be a separate application that does not otherwise participate and does not send or receive the RTP data packets. These are called third party monitors.

Non-RTP means: Protocols and mechanisms that may be needed in addition to RTP to provide a usable service. In particular, for multimedia conferences, a conference control application may distribute multicast addresses and keys for encryption, negotiate the encryption algorithm to be used, and define dynamic mappings between RTP payload type values and the payload formats they represent for formats that do not have a predefined payload type value. For simple applications, electronic mail or a conference database may also be used. The specification of such protocols and mechanisms is outside the scope of this document.

4. Byte Order, Alignment, and Time Format

All integer fields are carried in network byte order, that is, most significant byte (octet) first. This byte order is commonly known as big-endian. The transmission order is described in detail in [4]. Unless otherwise noted, numeric constants are in decimal (base 10).

Microsoft Exhibit 1019

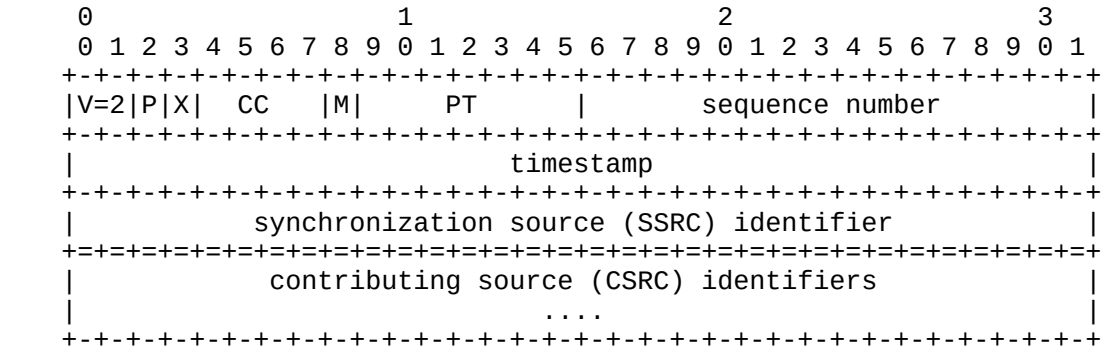
All header data is aligned to its natural length, i.e., 16-bit fields are aligned on even offsets, 32-bit fields are aligned at offsets divisible by four, etc. Octets designated as padding have the value zero.

Wallclock time (absolute time) is represented using the timestamp format of the Network Time Protocol (NTP), which is in seconds relative to 0h UTC on 1 January 1900 [5]. The full resolution NTP timestamp is a 64-bit unsigned fixed-point number with the integer part in the first 32 bits and the fractional part in the last 32 bits. In some fields where a more compact representation is appropriate, only the middle 32 bits are used; that is, the low 16 bits of the integer part and the high 16 bits of the fractional part. The high 16 bits of the integer part must be determined independently.

5. RTP Data Transfer Protocol

5.1 RTP Fixed Header Fields

The RTP header has the following format:



The first twelve octets are present in every RTP packet, while the list of CSRC identifiers is present only when inserted by a mixer. The fields have the following meaning:

version (V): 2 bits
This field identifies the version of RTP. The version defined by this specification is two (2). (The value 1 is used by the first draft version of RTP and the value 0 is used by the protocol initially implemented in the "vat" audio tool.)

padding (P): 1 bit
If the padding bit is set, the packet contains one or more additional padding octets at the end which are not part of the

Microsoft Exhibit 1019

payload. The last octet of the padding contains a count of how many padding octets should be ignored. Padding may be needed by some encryption algorithms with fixed block sizes or for carrying several RTP packets in a lower-layer protocol data unit.

extension (X): 1 bit
If the extension bit is set, the fixed header is followed by exactly one header extension, with a format defined in Section 5.3.1.

CSRC count (CC): 4 bits
The CSRC count contains the number of CSRC identifiers that follow the fixed header.

marker (M): 1 bit
The interpretation of the marker is defined by a profile. It is intended to allow significant events such as frame boundaries to be marked in the packet stream. A profile may define additional marker bits or specify that there is no marker bit by changing the number of bits in the payload type field (see Section 5.3).

payload type (PT): 7 bits
This field identifies the format of the RTP payload and determines its interpretation by the application. A profile specifies a default static mapping of payload type codes to payload formats. Additional payload type codes may be defined dynamically through non-RTP means (see Section 3). An initial set of default mappings for audio and video is specified in the companion profile Internet-Draft draft-ietf-avt-profile, and may be extended in future editions of the Assigned Numbers RFC [6]. An RTP sender emits a single RTP payload type at any given time; this field is not intended for multiplexing separate media streams (see Section 5.2).

sequence number: 16 bits
The sequence number increments by one for each RTP data packet sent, and may be used by the receiver to detect packet loss and to restore packet sequence. The initial value of the sequence number is random (unpredictable) to make known-plaintext attacks on encryption more difficult, even if the source itself does not encrypt, because the packets may flow through a translator that does. Techniques for choosing unpredictable numbers are discussed in [7].

timestamp: 32 bits
The timestamp reflects the sampling instant of the first octet in the RTP data packet. The sampling instant must be derived

Microsoft Exhibit 1019

from a clock that increments monotonically and linearly in time to allow synchronization and jitter calculations (see Section 6.3.1). The resolution of the clock must be sufficient for the desired synchronization accuracy and for measuring packet arrival jitter (one tick per video frame is typically not sufficient). The clock frequency is dependent on the format of data carried as payload and is specified statically in the profile or payload format specification that defines the format, or may be specified dynamically for payload formats defined through non-RTP means. If RTP packets are generated periodically, the nominal sampling instant as determined from the sampling clock is to be used, not a reading of the system clock. As an example, for fixed-rate audio the timestamp clock would likely increment by one for each sampling period. If an audio application reads blocks covering 160 sampling periods from the input device, the timestamp would be increased by 160 for each such block, regardless of whether the block is transmitted in a packet or dropped as silent.

The initial value of the timestamp is random, as for the sequence number. Several consecutive RTP packets may have equal timestamps if they are (logically) generated at once, e.g., belong to the same video frame. Consecutive RTP packets may contain timestamps that are not monotonic if the data is not transmitted in the order it was sampled, as in the case of MPEG interpolated video frames. (The sequence numbers of the packets as transmitted will still be monotonic.)

SSRC: 32 bits
The SSRC field identifies the synchronization source. This identifier is chosen randomly, with the intent that no two synchronization sources within the same RTP session will have the same SSRC identifier. An example algorithm for generating a random identifier is presented in Appendix A.6. Although the probability of multiple sources choosing the same identifier is low, all RTP implementations must be prepared to detect and resolve collisions. Section 8 describes the probability of collision along with a mechanism for resolving collisions and detecting RTP-level forwarding loops based on the uniqueness of the SSRC identifier. If a source changes its source transport address, it must also choose a new SSRC identifier to avoid being interpreted as a looped source.

CSRC list: 0 to 15 items, 32 bits each
The CSRC list identifies the contributing sources for the payload contained in this packet. The number of identifiers is given by the CC field. If there are more than 15 contributing sources, only 15 may be identified. CSRC identifiers are

Microsoft Exhibit 1019

inserted by mixers, using the SSRC identifiers of contributing sources. For example, for audio packets the SSRC identifiers of all sources that were mixed together to create a packet are listed, allowing correct talker indication at the receiver.

5.2 Multiplexing RTP Sessions

For efficient protocol processing, the number of multiplexing points should be minimized, as described in the integrated layer processing design principle [1]. In RTP, multiplexing is provided by the destination transport address (network address and port number) which define an RTP session. For example, in a teleconference composed of audio and video media encoded separately, each medium should be carried in a separate RTP session with its own destination transport address. It is not intended that the audio and video be carried in a single RTP session and demultiplexed based on the payload type or SSRC fields. Interleaving packets with different payload types but using the same SSRC would introduce several problems:

1. If one payload type were switched during a session, there would be no general means to identify which of the old values the new one replaced.

2. An SSRC is defined to identify a single timing and sequence number space. Interleaving multiple payload types would require different timing spaces if the media clock rates differ and would require different sequence number spaces to tell which payload type suffered packet loss.

3. The RTCP sender and receiver reports (see Section 6.3) can only describe one timing and sequence number space per SSRC and do not carry a payload type field.

4. An RTP mixer would not be able to combine interleaved streams of incompatible media into one stream.

5. Carrying multiple media in one RTP session precludes: the use of different network paths or network resource allocations if appropriate; reception of a subset of the media if desired, for example just audio if video would exceed the available bandwidth; and receiver implementations that use separate processes for the different media, whereas using separate RTP sessions permits either single- or multiple-process implementations.

Using a different SSRC for each medium but sending them in the same RTP session would avoid the first three problems but not the last two.

Microsoft Exhibit 1019

5.3 Profile-Specific Modifications to the RTP Header

The existing RTP data packet header is believed to be complete for the set of functions required in common across all the application classes that RTP might support. However, in keeping with the ALF design principle, the header may be tailored through modifications or additions defined in a profile specification while still allowing profile-independent monitoring and recording tools to function.

- The marker bit and payload type field carry profile-specific information, but they are allocated in the fixed header since many applications are expected to need them and might otherwise have to add another 32-bit word just to hold them. The octet containing these fields may be redefined by a profile to suit different requirements, for example with a more or fewer marker bits. If there are any marker bits, one should be located in the most significant bit of the octet since profile-independent monitors may be able to observe a correlation between packet loss patterns and the marker bit.

- Additional information that is required for a particular payload format, such as a video encoding, should be carried in the payload section of the packet. This might be in a header that is always present at the start of the payload section, or might be indicated by a reserved value in the data pattern.

- If a particular class of applications needs additional functionality independent of payload format, the profile under which those applications operate should define additional fixed fields to follow immediately after the SSRC field of the existing fixed header. Those applications will be able to quickly and directly access the additional fields while profile-independent monitors or recorders can still process the RTP packets by interpreting only the first twelve octets.

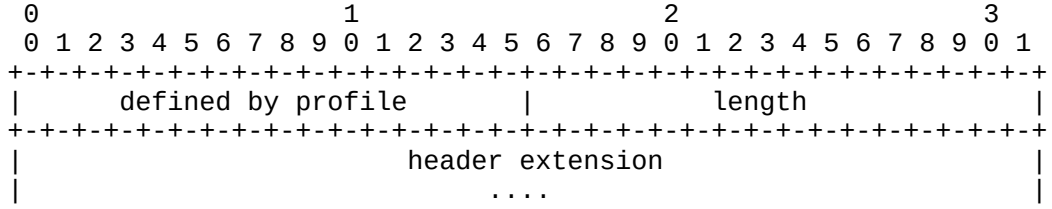
If it turns out that additional functionality is needed in common across all profiles, then a new version of RTP should be defined to make a permanent change to the fixed header.

5.3.1 RTP Header Extension

An extension mechanism is provided to allow individual implementations to experiment with new payload-format-independent functions that require additional information to be carried in the RTP data packet header. This mechanism is designed so that the header extension may be ignored by other interoperating implementations that have not been extended.

Microsoft Exhibit 1019

Note that this header extension is intended only for limited use. Most potential uses of this mechanism would be better done another way, using the methods described in the previous section. For example, a profile-specific extension to the fixed header is less expensive to process because it is not conditional nor in a variable location. Additional information required for a particular payload format should not use this header extension, but should be carried in the payload section of the packet.



If the X bit in the RTP header is one, a variable-length header extension is appended to the RTP header, following the CSRC list if present. The header extension contains a 16-bit length field that counts the number of 32-bit words in the extension, excluding the four-octet extension header (therefore zero is a valid length). Only a single extension may be appended to the RTP data header. To allow multiple interoperating implementations to each experiment independently with different header extensions, or to allow a particular implementation to experiment with more than one type of header extension, the first 16 bits of the header extension are left open for distinguishing identifiers or parameters. The format of these 16 bits is to be defined by the profile specification under which the implementations are operating. This RTP specification does not define any header extensions itself.

6. RTP Control Protocol -- RTCP

The RTP control protocol (RTCP) is based on the periodic transmission of control packets to all participants in the session, using the same distribution mechanism as the data packets. The underlying protocol must provide multiplexing of the data and control packets, for example using separate port numbers with UDP. RTCP performs four functions:

1. The primary function is to provide feedback on the quality of the data distribution. This is an integral part of the RTP's role as a transport protocol and is related to the flow and congestion control functions of other transport protocols. The feedback may be directly useful for control of adaptive encodings [8,9], but experiments with IP

Microsoft Exhibit 1019

multicasting have shown that it is also critical to get feedback from the receivers to diagnose faults in the distribution. Sending reception feedback reports to all participants allows one who is observing problems to evaluate whether those problems are local or global. With a distribution mechanism like IP multicast, it is also possible for an entity such as a network service provider who is not otherwise involved in the session to receive the feedback information and act as a third-party monitor to diagnose network problems. This feedback function is performed by the RTCP sender and receiver reports, described below in Section 6.3.

2. RTCP carries a persistent transport-level identifier for an RTP source called the canonical name or CNAME, Section 6.4.1. Since the SSRC identifier may change if a conflict is discovered or a program is restarted, receivers require the CNAME to keep track of each participant. Receivers also require the CNAME to associate multiple data streams from a given participant in a set of related RTP sessions, for example to synchronize audio and video.

3. The first two functions require that all participants send RTCP packets, therefore the rate must be controlled in order for RTP to scale up to a large number of participants. By having each participant send its control packets to all the others, each can independently observe the number of participants. This number is used to calculate the rate at which the packets are sent, as explained in Section 6.2.

4. A fourth, optional function is to convey minimal session control information, for example participant identification to be displayed in the user interface. This is most likely to be useful in "loosely controlled" sessions where participants enter and leave without membership control or parameter negotiation. RTCP serves as a convenient channel to reach all the participants, but it is not necessarily expected to support all the control communication requirements of an application. A higher-level session control protocol, which is beyond the scope of this document, may be needed.

Functions 1-3 are mandatory when RTP is used in the IP multicast environment, and are recommended for all environments. RTP application designers are advised to avoid mechanisms that can only work in unicast mode and will not scale to larger numbers.

Microsoft Exhibit 1019

6.1 RTCP Packet Format

This specification defines several RTCP packet types to carry a variety of control information:

SR: Sender report, for transmission and reception statistics from participants that are active senders

RR: Receiver report, for reception statistics from participants that are not active senders

SDES: Source description items, including CNAME

BYE: Indicates end of participation

APP: Application specific functions

Each RTCP packet begins with a fixed part similar to that of RTP data packets, followed by structured elements that may be of variable length according to the packet type but always end on a 32-bit boundary. The alignment requirement and a length field in the fixed part are included to make RTCP packets "stackable". Multiple RTCP packets may be concatenated without any intervening separators to form a compound RTCP packet that is sent in a single packet of the lower layer protocol, for example UDP. There is no explicit count of individual RTCP packets in the compound packet since the lower layer protocols are expected to provide an overall length to determine the end of the compound packet.

Each individual RTCP packet in the compound packet may be processed independently with no requirements upon the order or combination of packets. However, in order to perform the functions of the protocol, the following constraints are imposed:

- Reception statistics (in SR or RR) should be sent as often as bandwidth constraints will allow to maximize the resolution of the statistics, therefore each periodically transmitted compound RTCP packet should include a report packet.

- New receivers need to receive the CNAME for a source as soon as possible to identify the source and to begin associating media for purposes such as lip-sync, so each compound RTCP packet should also include the SDES CNAME.

- The number of packet types that may appear first in the compound packet should be limited to increase the number of constant bits in the first word and the probability of successfully validating RTCP packets against misaddressed RTP

Microsoft Exhibit 1019

data packets or other unrelated packets.

Thus, all RTCP packets must be sent in a compound packet of at least two individual packets, with the following format recommended:

Encryption prefix: If and only if the compound packet is to be encrypted, it is prefixed by a random 32-bit quantity redrawn for every compound packet transmitted.

SR or RR: The first RTCP packet in the compound packet must always be a report packet to facilitate header validation as described in Appendix A.2. This is true even if no data has been sent nor received, in which case an empty RR is sent, and even if the only other RTCP packet in the compound packet is a BYE.

Additional RRs: If the number of sources for which reception statistics are being reported exceeds 31, the number that will fit into one SR or RR packet, then additional RR packets should follow the initial report packet.

SDES: An SDES packet containing a CNAME item must be included in each compound RTCP packet. Other source description items may optionally be included if required by a particular application, subject to bandwidth constraints (see Section 6.2.2).

BYE or APP: Other RTCP packet types, including those yet to be defined, may follow in any order, except that BYE should be the last packet sent with a given SSRC/CSRC. Packet types may appear more than once.

It is advisable for translators and mixers to combine individual RTCP packets from the multiple sources they are forwarding into one compound packet whenever feasible in order to amortize the packet overhead (see Section 7). An example RTCP compound packet as might be produced by a mixer is shown in Fig. 1. If the overall length of a compound packet would exceed the maximum transmission unit (MTU) of the network path, it may be segmented into multiple shorter compound packets to be transmitted in separate packets of the underlying protocol. Note that each of the compound packets must begin with an SR or RR packet.

An implementation may ignore incoming RTCP packets with types unknown to it. Additional RTCP packet types may be registered with the Internet Assigned Numbers Authority (IANA).

Microsoft Exhibit 1019

6.2 RTCP Transmission Interval

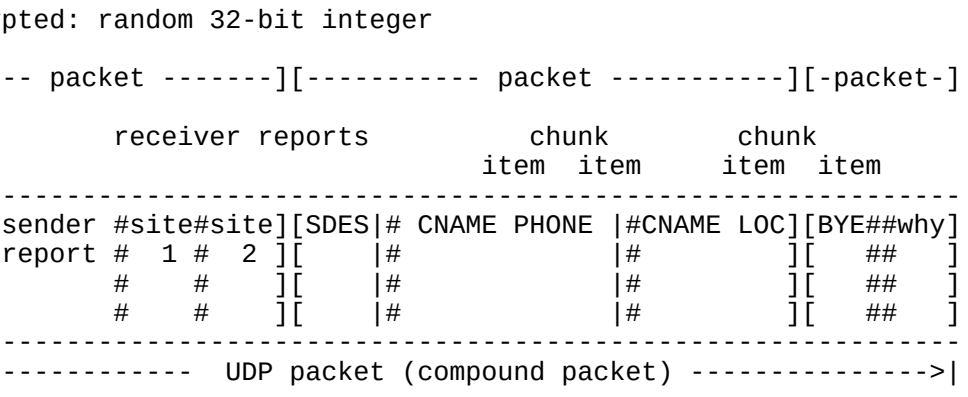


Figure 1: Example of an RTCP compound packet

RTP is designed to allow an application to scale automatically over session sizes ranging from a few participants to thousands. For example, in an audio conference the data traffic is inherently self-limiting because only one or two people will speak at a time, so with multicast distribution the data rate on any given link remains relatively constant independent of the number of participants. However, the control traffic is not self-limiting. If the reception reports from each participant were sent at a constant rate, the control traffic would grow linearly with the number of participants. Therefore, the rate must be scaled down.

For each session, it is assumed that the data traffic is subject to an aggregate limit called the "session bandwidth" to be divided among the participants. This bandwidth might be reserved and the limit enforced by the network, or it might just be a reasonable share. The session bandwidth may be chosen based or some cost or a priori knowledge of the available network bandwidth for the session. It is somewhat independent of the media encoding, but the encoding choice may be limited by the session bandwidth. The session bandwidth parameter is expected to be supplied by a session management application when it invokes a media application, but media applications may also set a default based on the single-sender data bandwidth for the encoding selected for the session. The application may also enforce bandwidth limits based on multicast scope rules or other criteria.

Microsoft Exhibit 1019

Bandwidth calculations for control and data traffic include lower-layer transport and network protocols (e.g., UDP and IP) since that is what the resource reservation system would need to know. The application can also be expected to know which of these protocols are in use. Link level headers are not included in the calculation since the packet will be encapsulated with different link level headers as it travels.

The control traffic should be limited to a small and known fraction of the session bandwidth: small so that the primary function of the transport protocol to carry data is not impaired; known so that the control traffic can be included in the bandwidth specification given to a resource reservation protocol, and so that each participant can independently calculate its share. It is suggested that the fraction of the session bandwidth allocated to RTCP be fixed at 5%. While the value of this and other constants in the interval calculation is not critical, all participants in the session must use the same values so the same interval will be calculated. Therefore, these constants should be fixed for a particular profile.

The algorithm described in Appendix A.7 was designed to meet the goals outlined above. It calculates the interval between sending compound RTCP packets to divide the allowed control traffic bandwidth among the participants. This allows an application to provide fast response for small sessions where, for example, identification of all participants is important, yet automatically adapt to large sessions. The algorithm incorporates the following characteristics:

- Senders are collectively allocated at least 1/4 of the control traffic bandwidth so that in sessions with a large number of receivers but a small number of senders, newly joining participants will more quickly receive the CNAME for the sending sites.

- The calculated interval between RTCP packets is required to be greater than a minimum of 5 seconds to avoid having bursts of RTCP packets exceed the allowed bandwidth when the number of participants is small and the traffic isn’t smoothed according to the law of large numbers.

- The interval between RTCP packets is varied randomly over the range [0.5,1.5] times the calculated interval to avoid unintended synchronization of all participants [10]. The first RTCP packet sent after joining a session is also delayed by a random variation of half the minimum RTCP interval in case the application is started at multiple sites simultaneously, for example as initiated by a session announcement.

Microsoft Exhibit 1019

- o A dynamic estimate of the average compound RTCP packet size is calculated, including all those received and sent, to automatically adapt to changes in the amount of control information carried.

This algorithm may be used for sessions in which all participants are allowed to send. In that case, the session bandwidth parameter is the product of the individual sender's bandwidth times the number of participants, and the RTCP bandwidth is 5% of that.

6.2.1 Maintaining the number of session members

Calculation of the RTCP packet interval depends upon an estimate of the number of sites participating in the session. New sites are added to the count when they are heard, and an entry for each is created in a table indexed by the SSRC or CSRC identifier (see Section 8.2) to keep track of them. New entries may not be considered valid until multiple packets carrying the new SSRC have been received (see Appendix A.1). Entries may be deleted from the table when an RTCP BYE packet with the corresponding SSRC identifier is received.

A participant may mark another site inactive, or delete it if not yet valid, if no RTP or RTCP packet has been received for a small number of RTCP report intervals (5 is suggested). This provides some robustness against packet loss. All sites must calculate roughly the same value for the RTCP report interval in order for this timeout to work properly.

Once a site has been validated, then if it is later marked inactive the state for that site should still be retained and the site should continue to be counted in the total number of sites sharing RTCP bandwidth for a period long enough to span typical network partitions. This is to avoid excessive traffic, when the partition heals, due to an RTCP report interval that is too small. A timeout of 30 minutes is suggested. Note that this is still larger than 5 times the largest value to which the RTCP report interval is expected to usefully scale, about 2 to 5 minutes.

6.2.2 Allocation of source description bandwidth

This specification defines several source description (SDES) items in addition to the mandatory CNAME item, such as NAME (personal name) and EMAIL (email address). It also provides a means to define new application-specific RTCP packet types. Applications should exercise caution in allocating control bandwidth to this additional information because it will slow down the rate at which reception reports and CNAME are sent, thus impairing the performance of the protocol. It is recommended that no more than 20% of the RTCP

Microsoft Exhibit 1019

bandwidth allocated to a single participant be used to carry the additional information. Furthermore, it is not intended that all SDES items should be included in every application. Those that are included should be assigned a fraction of the bandwidth according to their utility. Rather than estimate these fractions dynamically, it is recommended that the percentages be translated statically into report interval counts based on the typical length of an item.

For example, an application may be designed to send only CNAME, NAME and EMAIL and not any others. NAME might be given much higher priority than EMAIL because the NAME would be displayed continuously in the application's user interface, whereas EMAIL would be displayed only when requested. At every RTCP interval, an RR packet and an SDES packet with the CNAME item would be sent. For a small session operating at the minimum interval, that would be every 5 seconds on the average. Every third interval (15 seconds), one extra item would be included in the SDES packet. Seven out of eight times this would be the NAME item, and every eighth time (2 minutes) it would be the EMAIL item.

When multiple applications operate in concert using cross-application binding through a common CNAME for each participant, for example in a multimedia conference composed of an RTP session for each medium, the additional SDES information might be sent in only one RTP session. The other sessions would carry only the CNAME item.

6.3 Sender and Receiver Reports

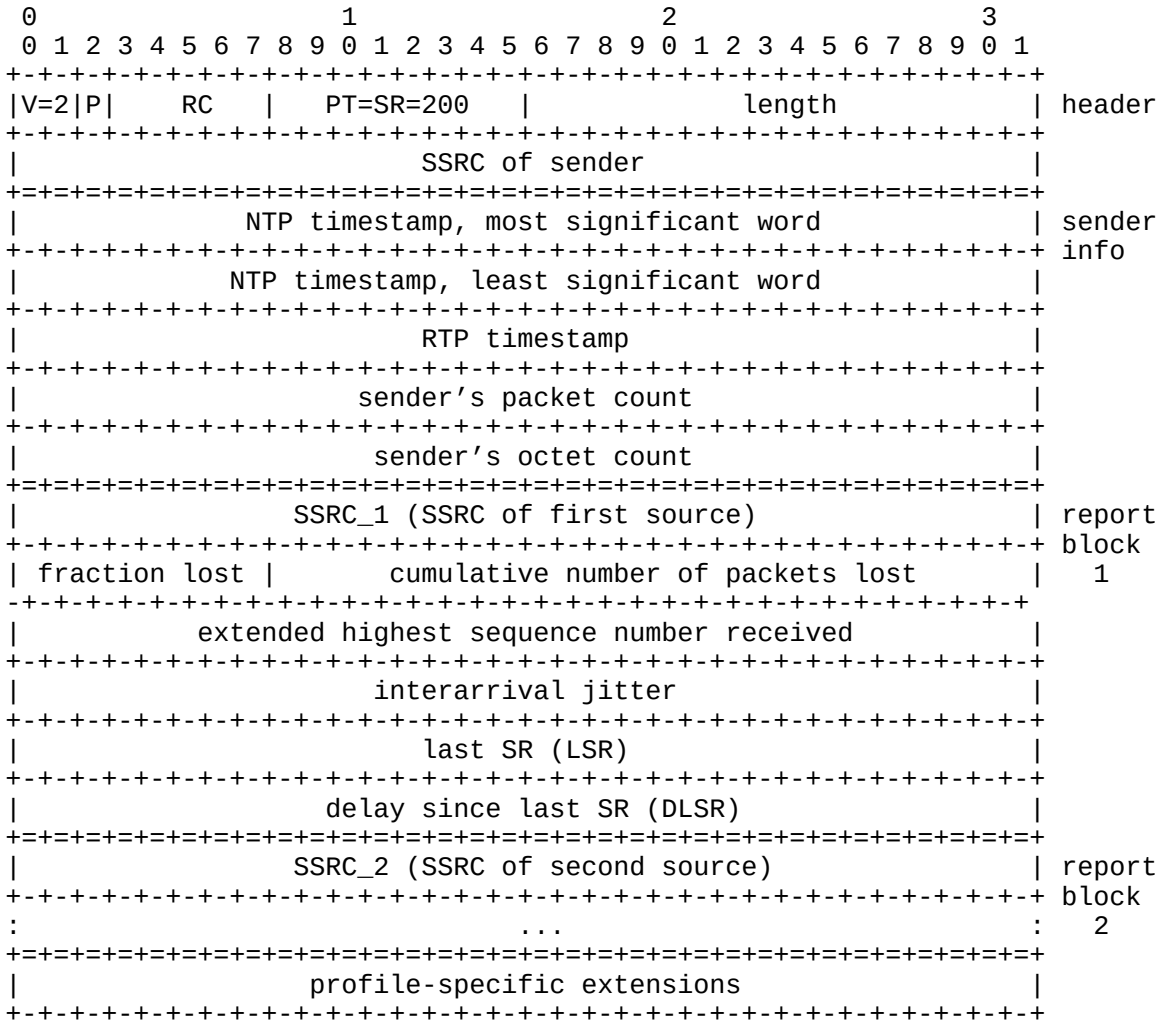
RTP receivers provide reception quality feedback using RTCP report packets which may take one of two forms depending upon whether or not the receiver is also a sender. The only difference between the sender report (SR) and receiver report (RR) forms, besides the packet type code, is that the sender report includes a 20-byte sender information section for use by active senders. The SR is issued if a site has sent any data packets during the interval since issuing the last report or the previous one, otherwise the RR is issued.

Both the SR and RR forms include zero or more reception report blocks, one for each of the synchronization sources from which this receiver has received RTP data packets since the last report. Reports are not issued for contributing sources listed in the CSRC list. Each reception report block provides statistics about the data received from the particular source indicated in that block. Since a maximum of 31 reception report blocks will fit in an SR or RR packet, additional RR packets may be stacked after the initial SR or RR packet as needed to contain the reception reports for all sources heard during the interval since the last report.

Microsoft Exhibit 1019

The next sections define the formats of the two reports, how they may be extended in a profile-specific manner if an application requires additional feedback information, and how the reports may be used. Details of reception reporting by translators and mixers is given in Section 7.

6.3.1 SR: Sender report RTCP packet



The sender report packet consists of three sections, possibly followed by a fourth profile-specific extension section if defined. The first section, the header, is 8 octets long. The fields have the following meaning:

Microsoft Exhibit 1019

version (V): 2 bits
Identifies the version of RTP, which is the same in RTCP packets as in RTP data packets. The version defined by this specification is two (2).

padding (P): 1 bit
If the padding bit is set, this RTCP packet contains some additional padding octets at the end which are not part of the control information. The last octet of the padding is a count of how many padding octets should be ignored. Padding may be needed by some encryption algorithms with fixed block sizes. In a compound RTCP packet, padding should only be required on the last individual packet because the compound packet is encrypted as a whole.

reception report count (RC): 5 bits
The number of reception report blocks contained in this packet. A value of zero is valid.

packet type (PT): 8 bits
Contains the constant 200 to identify this as an RTCP SR packet.

length: 16 bits
The length of this RTCP packet in 32-bit words minus one, including the header and any padding. (The offset of one makes zero a valid length and avoids a possible infinite loop in scanning a compound RTCP packet, while counting 32-bit words avoids a validity check for a multiple of 4.)

SSRC: 32 bits
The synchronization source identifier for the originator of this SR packet.

The second section, the sender information, is 20 octets long and is present in every sender report packet. It summarizes the data transmissions from this sender. The fields have the following meaning:

NTP timestamp: 64 bits
Indicates the wallclock time when this report was sent so that it may be used in combination with timestamps returned in reception reports from other receivers to measure round-trip propagation to those receivers. Receivers should expect that the measurement accuracy of the timestamp may be limited to far less than the resolution of the NTP timestamp. The measurement uncertainty of the timestamp is not indicated as it may not be known. A sender that can keep track of elapsed time but has no notion of wallclock time may use the elapsed time since joining

Microsoft Exhibit 1019

the session instead. This is assumed to be less than 68 years, so the high bit will be zero. It is permissible to use the sampling clock to estimate elapsed wallclock time. A sender that has no notion of wallclock or elapsed time may set the NTP timestamp to zero.

RTP timestamp: 32 bits
Corresponds to the same time as the NTP timestamp (above), but in the same units and with the same random offset as the RTP timestamps in data packets. This correspondence may be used for intra- and inter-media synchronization for sources whose NTP timestamps are synchronized, and may be used by media-independent receivers to estimate the nominal RTP clock frequency. Note that in most cases this timestamp will not be equal to the RTP timestamp in any adjacent data packet. Rather, it is calculated from the corresponding NTP timestamp using the relationship between the RTP timestamp counter and real time as maintained by periodically checking the wallclock time at a sampling instant.

sender's packet count: 32 bits
The total number of RTP data packets transmitted by the sender since starting transmission up until the time this SR packet was generated. The count is reset if the sender changes its SSRC identifier.

sender's octet count: 32 bits
The total number of payload octets (i.e., not including header or padding) transmitted in RTP data packets by the sender since starting transmission up until the time this SR packet was generated. The count is reset if the sender changes its SSRC identifier. This field can be used to estimate the average payload data rate.

The third section contains zero or more reception report blocks depending on the number of other sources heard by this sender since the last report. Each reception report block conveys statistics on the reception of RTP packets from a single synchronization source. Receivers do not carry over statistics when a source changes its SSRC identifier due to a collision. These statistics are:

SSRC_n (source identifier): 32 bits
The SSRC identifier of the source to which the information in this reception report block pertains.

fraction lost: 8 bits
The fraction of RTP data packets from source SSRC_n lost since the previous SR or RR packet was sent, expressed as a fixed

Microsoft Exhibit 1019

point number with the binary point at the left edge of the field. (That is equivalent to taking the integer part after multiplying the loss fraction by 256.) This fraction is defined to be the number of packets lost divided by the number of packets expected, as defined in the next paragraph. An implementation is shown in Appendix A.3. If the loss is negative due to duplicates, the fraction lost is set to zero. Note that a receiver cannot tell whether any packets were lost after the last one received, and that there will be no reception report block issued for a source if all packets from that source sent during the last reporting interval have been lost.

cumulative number of packets lost: 24 bits
The total number of RTP data packets from source SSRC_n that have been lost since the beginning of reception. This number is defined to be the number of packets expected less the number of packets actually received, where the number of packets received includes any which are late or duplicates. Thus packets that arrive late are not counted as lost, and the loss may be negative if there are duplicates. The number of packets expected is defined to be the extended last sequence number received, as defined next, less the initial sequence number received. This may be calculated as shown in Appendix A.3.

extended highest sequence number received: 32 bits
The low 16 bits contain the highest sequence number received in an RTP data packet from source SSRC_n, and the most significant 16 bits extend that sequence number with the corresponding count of sequence number cycles, which may be maintained according to the algorithm in Appendix A.1. Note that different receivers within the same session will generate different extensions to the sequence number if their start times differ significantly.

interarrival jitter: 32 bits
An estimate of the statistical variance of the RTP data packet interarrival time, measured in timestamp units and expressed as an unsigned integer. The interarrival jitter J is defined to be the mean deviation (smoothed absolute value) of the difference D in packet spacing at the receiver compared to the sender for a pair of packets. As shown in the equation below, this is equivalent to the difference in the "relative transit time" for the two packets; the relative transit time is the difference between a packet’s RTP timestamp and the receiver’s clock at the time of arrival, measured in the same units.

Microsoft Exhibit 1019

If Si is the RTP timestamp from packet i, and Ri is the time of arrival in RTP timestamp units for packet i, then for two packets i and j, D may be expressed as

$$D(i,j)=(Rj-Ri)-(Sj-Si)=(Rj-Sj)-(Ri-Si)$$

The interarrival jitter is calculated continuously as each data packet i is received from source SSRC_n, using this difference D for that packet and the previous packet i-1 in order of arrival (not necessarily in sequence), according to the formula

$$J=J+(|D(i-1,i)|-J)/16$$

Whenever a reception report is issued, the current value of J is sampled.

The jitter calculation is prescribed here to allow profile-independent monitors to make valid interpretations of reports coming from different implementations. This algorithm is the optimal first-order estimator and the gain parameter 1/16 gives a good noise reduction ratio while maintaining a reasonable rate of convergence [11]. A sample implementation is shown in Appendix A.8.

last SR timestamp (LSR): 32 bits
: The middle 32 bits out of 64 in the NTP timestamp (as explained in Section 4) received as part of the most recent RTCP sender report (SR) packet from source SSRC_n. If no SR has been received yet, the field is set to zero.

delay since last SR (DLSR): 32 bits
: The delay, expressed in units of 1/65536 seconds, between receiving the last SR packet from source SSRC_n and sending this reception report block. If no SR packet has been received yet from SSRC_n, the DLSR field is set to zero.

Let SSRC_r denote the receiver issuing this receiver report. Source SSRC_n can compute the round propagation delay to SSRC_r by recording the time A when this reception report block is received. It calculates the total round-trip time A-LSR using the last SR timestamp (LSR) field, and then subtracting this field to leave the round-trip propagation delay as (A- LSR - DLSR). This is illustrated in Fig. 2.

This may be used as an approximate measure of distance to cluster receivers, although some links have very asymmetric delays.

Microsoft Exhibit 1019

6.3.2 RR: Receiver report RTCP packet

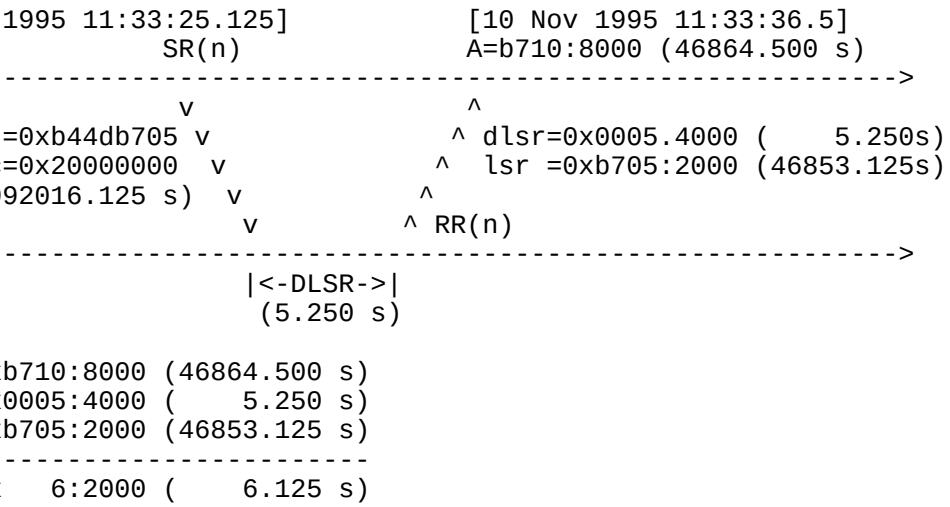
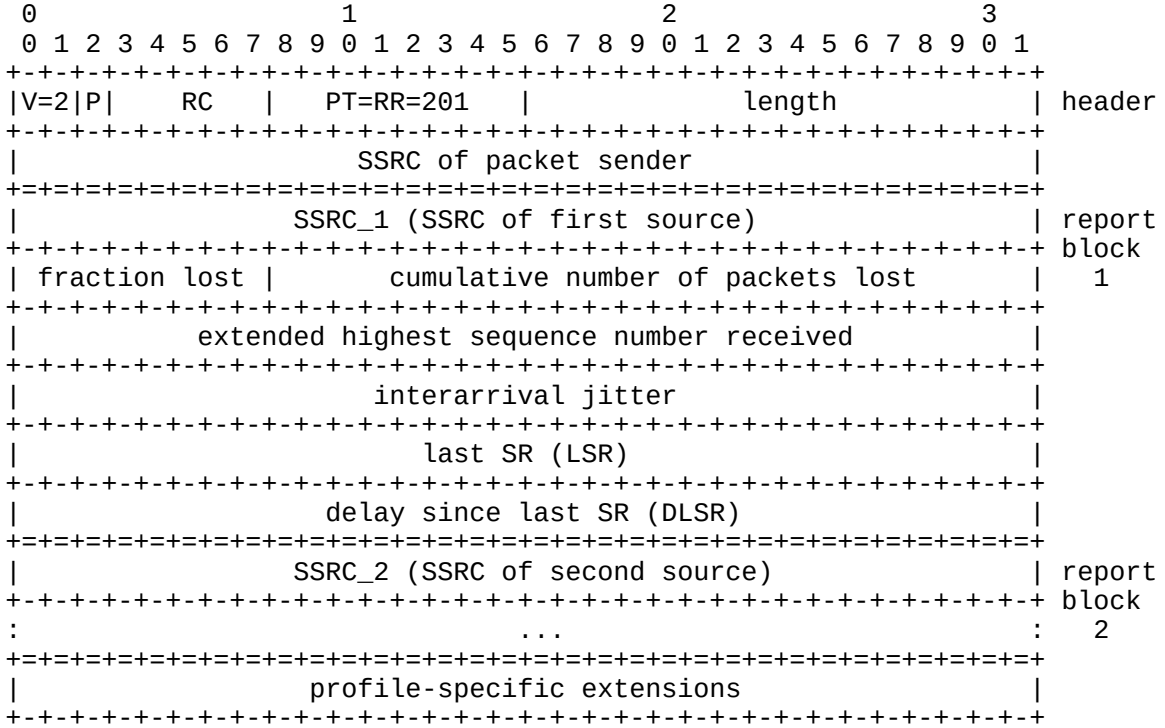


Figure 2: Example for round-trip time computation



Microsoft Exhibit 1019

The format of the receiver report (RR) packet is the same as that of the SR packet except that the packet type field contains the constant 201 and the five words of sender information are omitted (these are the NTP and RTP timestamps and sender's packet and octet counts). The remaining fields have the same meaning as for the SR packet.

An empty RR packet (RC = 0) is put at the head of a compound RTCP packet when there is no data transmission or reception to report.

6.3.3 Extending the sender and receiver reports

A profile should define profile- or application-specific extensions to the sender report and receiver if there is additional information that should be reported regularly about the sender or receivers. This method should be used in preference to defining another RTCP packet type because it requires less overhead:

- fewer octets in the packet (no RTCP header or SSRC field);
- simpler and faster parsing because applications running under that profile would be programmed to always expect the extension fields in the directly accessible location after the reception reports.

If additional sender information is required, it should be included first in the extension for sender reports, but would not be present in receiver reports. If information about receivers is to be included, that data may be structured as an array of blocks parallel to the existing array of reception report blocks; that is, the number of blocks would be indicated by the RC field.

6.3.4 Analyzing sender and receiver reports

It is expected that reception quality feedback will be useful not only for the sender but also for other receivers and third-party monitors. The sender may modify its transmissions based on the feedback; receivers can determine whether problems are local, regional or global; network managers may use profile-independent monitors that receive only the RTCP packets and not the corresponding RTP data packets to evaluate the performance of their networks for multicast distribution.

Cumulative counts are used in both the sender information and receiver report blocks so that differences may be calculated between any two reports to make measurements over both short and long time periods, and to provide resilience against the loss of a report. The difference between the last two reports received can be used to estimate the recent quality of the distribution. The NTP timestamp is

Microsoft Exhibit 1019

included so that rates may be calculated from these differences over the interval between two reports. Since that timestamp is independent of the clock rate for the data encoding, it is possible to implement encoding- and profile-independent quality monitors.

An example calculation is the packet loss rate over the interval between two reception reports. The difference in the cumulative number of packets lost gives the number lost during that interval. The difference in the extended last sequence numbers received gives the number of packets expected during the interval. The ratio of these two is the packet loss fraction over the interval. This ratio should equal the fraction lost field if the two reports are consecutive, but otherwise not. The loss rate per second can be obtained by dividing the loss fraction by the difference in NTP timestamps, expressed in seconds. The number of packets received is the number of packets expected minus the number lost. The number of packets expected may also be used to judge the statistical validity of any loss estimates. For example, 1 out of 5 packets lost has a lower significance than 200 out of 1000.

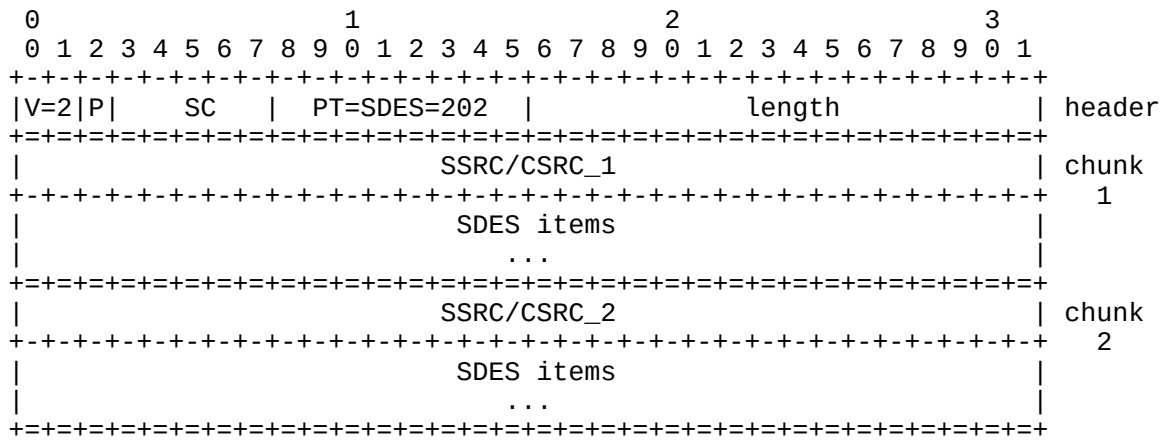
From the sender information, a third-party monitor can calculate the average payload data rate and the average packet rate over an interval without receiving the data. Taking the ratio of the two gives the average payload size. If it can be assumed that packet loss is independent of packet size, then the number of packets received by a particular receiver times the average payload size (or the corresponding packet size) gives the apparent throughput available to that receiver.

In addition to the cumulative counts which allow long-term packet loss measurements using differences between reports, the fraction lost field provides a short-term measurement from a single report. This becomes more important as the size of a session scales up enough that reception state information might not be kept for all receivers or the interval between reports becomes long enough that only one report might have been received from a particular receiver.

The interarrival jitter field provides a second short-term measure of network congestion. Packet loss tracks persistent congestion while the jitter measure tracks transient congestion. The jitter measure may indicate congestion before it leads to packet loss. Since the interarrival jitter field is only a snapshot of the jitter at the time of a report, it may be necessary to analyze a number of reports from one receiver over time or from multiple receivers, e.g., within a single network.

Microsoft Exhibit 1019

6.4 SDES: Source description RTCP packet



The SDES packet is a three-level structure composed of a header and zero or more chunks, each of of which is composed of items describing the source identified in that chunk. The items are described individually in subsequent sections.

version (V), padding (P), length:
As described for the SR packet (see Section 6.3.1).

packet type (PT): 8 bits
Contains the constant 202 to identify this as an RTCP SDES packet.

source count (SC): 5 bits
The number of SSRC/CSRC chunks contained in this SDES packet. A value of zero is valid but useless.

Each chunk consists of an SSRC/CSRC identifier followed by a list of zero or more items, which carry information about the SSRC/CSRC. Each chunk starts on a 32-bit boundary. Each item consists of an 8-bit type field, an 8-bit octet count describing the length of the text (thus, not including this two-octet header), and the text itself. Note that the text can be no longer than 255 octets, but this is consistent with the need to limit RTCP bandwidth consumption.

The text is encoded according to the UTF-2 encoding specified in Annex F of ISO standard 10646 [12,13]. This encoding is also known as UTF-8 or UTF-FSS. It is described in "File System Safe UCS Transformation Format (FSS_UTF)", X/Open Preliminary Specification, Document Number P316 and Unicode Technical Report #4. US-ASCII is a subset of this encoding and requires no additional encoding. The

Microsoft Exhibit 1019

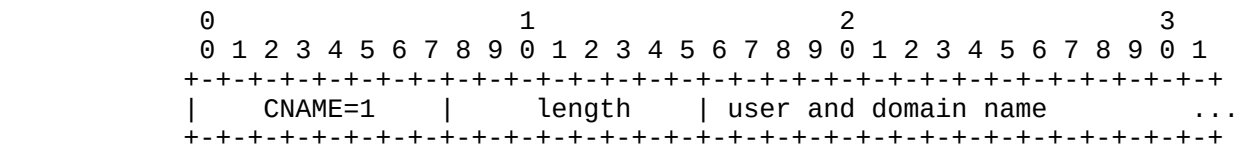
presence of multi-octet encodings is indicated by setting the most significant bit of a character to a value of one.

Items are contiguous, i.e., items are not individually padded to a 32-bit boundary. Text is not null terminated because some multi-octet encodings include null octets. The list of items in each chunk is terminated by one or more null octets, the first of which is interpreted as an item type of zero to denote the end of the list, and the remainder as needed to pad until the next 32-bit boundary. A chunk with zero items (four null octets) is valid but useless.

End systems send one SDES packet containing their own source identifier (the same as the SSRC in the fixed RTP header). A mixer sends one SDES packet containing a chunk for each contributing source from which it is receiving SDES information, or multiple complete SDES packets in the format above if there are more than 31 such sources (see Section 7).

The SDES items currently defined are described in the next sections. Only the CNAME item is mandatory. Some items shown here may be useful only for particular profiles, but the item types are all assigned from one common space to promote shared use and to simplify profile-independent applications. Additional items may be defined in a profile by registering the type numbers with IANA.

6.4.1 CNAME: Canonical end-point identifier SDES item



The CNAME identifier has the following properties:

- Because the randomly allocated SSRC identifier may change if a conflict is discovered or if a program is restarted, the CNAME item is required to provide the binding from the SSRC identifier to an identifier for the source that remains constant.
- Like the SSRC identifier, the CNAME identifier should also be unique among all participants within one RTP session.
- To provide a binding across multiple media tools used by one participant in a set of related RTP sessions, the CNAME should be fixed for that participant.

Microsoft Exhibit 1019

o To facilitate third-party monitoring, the CNAME should be suitable for either a program or a person to locate the source.

Therefore, the CNAME should be derived algorithmically and not entered manually, when possible. To meet these requirements, the following format should be used unless a profile specifies an alternate syntax or semantics. The CNAME item should have the format "user@host", or "host" if a user name is not available as on single-user systems. For both formats, "host" is either the fully qualified domain name of the host from which the real-time data originates, formatted according to the rules specified in RFC 1034 [14], RFC 1035 [15] and Section 2.1 of RFC 1123 [16]; or the standard ASCII representation of the host's numeric address on the interface used for the RTP communication. For example, the standard ASCII representation of an IP Version 4 address is "dotted decimal", also known as dotted quad. Other address types are expected to have ASCII representations that are mutually unique. The fully qualified domain name is more convenient for a human observer and may avoid the need to send a NAME item in addition, but it may be difficult or impossible to obtain reliably in some operating environments. Applications that may be run in such environments should use the ASCII representation of the address instead.

Examples are "doe@sleepy.megacorp.com" or "doe@192.0.2.89" for a multi-user system. On a system with no user name, examples would be "sleepy.megacorp.com" or "192.0.2.89".

The user name should be in a form that a program such as "finger" or "talk" could use, i.e., it typically is the login name rather than the personal name. The host name is not necessarily identical to the one in the participant's electronic mail address.

This syntax will not provide unique identifiers for each source if an application permits a user to generate multiple sources from one host. Such an application would have to rely on the SSRC to further identify the source, or the profile for that application would have to specify additional syntax for the CNAME identifier.

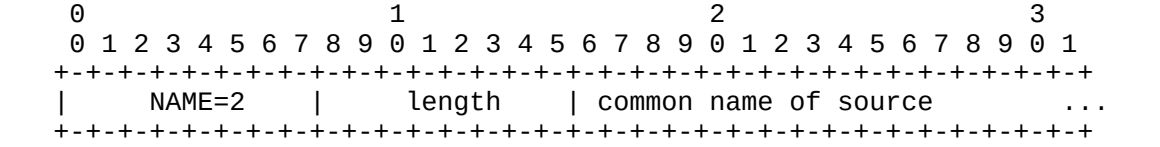
If each application creates its CNAME independently, the resulting CNAMEs may not be identical as would be required to provide a binding across multiple media tools belonging to one participant in a set of related RTP sessions. If cross-media binding is required, it may be necessary for the CNAME of each tool to be externally configured with the same value by a coordination tool.

Application writers should be aware that private network address assignments such as the Net-10 assignment proposed in RFC 1597 [17] may create network addresses that are not globally unique. This would

Microsoft Exhibit 1019

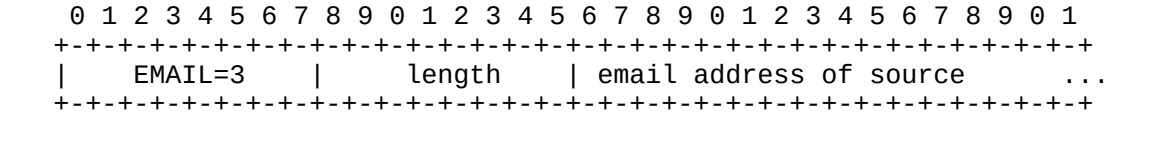
lead to non-unique CNAMEs if hosts with private addresses and no direct IP connectivity to the public Internet have their RTP packets forwarded to the public Internet through an RTP-level translator. (See also RFC 1627 [18].) To handle this case, applications may provide a means to configure a unique CNAME, but the burden is on the translator to translate CNAMEs from private addresses to public addresses if necessary to keep private addresses from being exposed.

6.4.2 NAME: User name SDES item



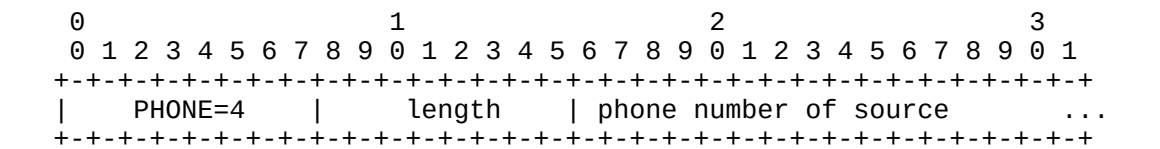
This is the real name used to describe the source, e.g., "John Doe, Bit Recycler, Megacorp". It may be in any form desired by the user. For applications such as conferencing, this form of name may be the most desirable for display in participant lists, and therefore might be sent most frequently of those items other than CNAME. Profiles may establish such priorities. The NAME value is expected to remain constant at least for the duration of a session. It should not be relied upon to be unique among all participants in the session.

6.4.3 EMAIL: Electronic mail address SDES item



The email address is formatted according to RFC 822 [19], for example, "John.Doe@megacorp.com". The EMAIL value is expected to remain constant for the duration of a session.

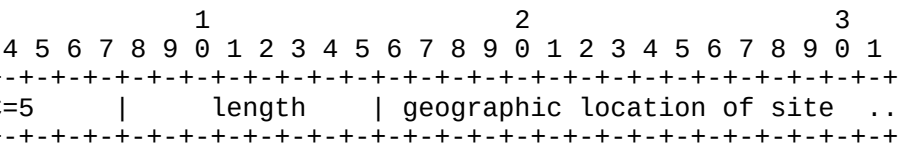
6.4.4 PHONE: Phone number SDES item



The phone number should be formatted with the plus sign replacing the international access code. For example, "+1 908 555 1212" for a number in the United States.

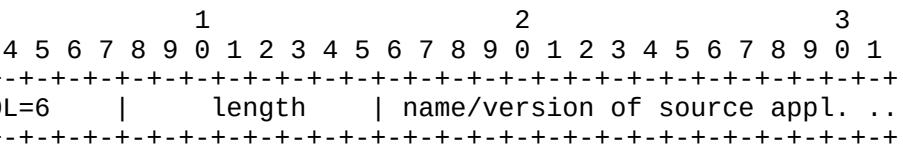
Microsoft Exhibit 1019

6.4.5 LOC: Geographic user location SDES item



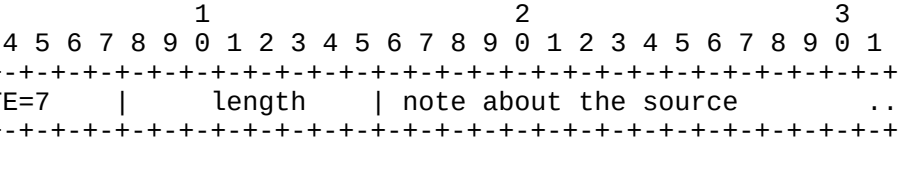
Depending on the application, different degrees of detail are appropriate for this item. For conference applications, a string like "Murray Hill, New Jersey" may be sufficient, while, for an active badge system, strings like "Room 2A244, AT&T BL MH" might be appropriate. The degree of detail is left to the implementation and/or user, but format and content may be prescribed by a profile. The LOC value is expected to remain constant for the duration of a session, except for mobile hosts.

6.4.6 TOOL: Application or tool name SDES item



A string giving the name and possibly version of the application generating the stream, e.g., "videotool 1.2". This information may be useful for debugging purposes and is similar to the Mailer or Mail-System-Version SMTP headers. The TOOL value is expected to remain constant for the duration of the session.

6.4.7 NOTE: Notice/status SDES item



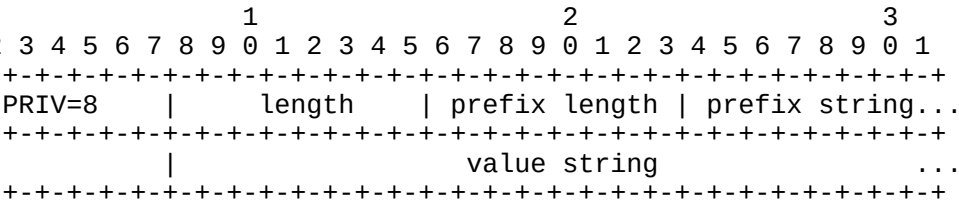
The following semantics are suggested for this item, but these or other semantics may be explicitly defined by a profile. The NOTE item is intended for transient messages describing the current state of the source, e.g., "on the phone, can't talk". Or, during a seminar, this item might be used to convey the title of the talk. It should be used only to carry exceptional information and should not be included routinely by all participants because this would slow down the rate at which reception reports and CNAME are sent, thus impairing the performance of the protocol. In particular, it should not be included

Microsoft Exhibit 1019

as an item in a user's configuration file nor automatically generated as in a quote-of-the-day.

Since the NOTE item may be important to display while it is active, the rate at which other non-CNAME items such as NAME are transmitted might be reduced so that the NOTE item can take that part of the RTCP bandwidth. When the transient message becomes inactive, the NOTE item should continue to be transmitted a few times at the same repetition rate but with a string of length zero to signal the receivers. However, receivers should also consider the NOTE item inactive if it is not received for a small multiple of the repetition rate, or perhaps 20-30 RTCP intervals.

6.4.8 PRIV: Private extensions SDES item



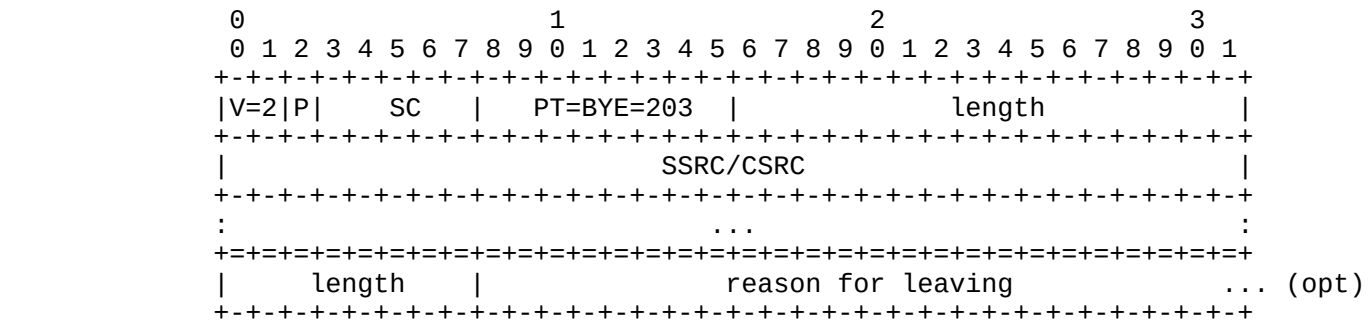
This item is used to define experimental or application-specific SDES extensions. The item contains a prefix consisting of a length-string pair, followed by the value string filling the remainder of the item and carrying the desired information. The prefix length field is 8 bits long. The prefix string is a name chosen by the person defining the PRIV item to be unique with respect to other PRIV items this application might receive. The application creator might choose to use the application name plus an additional subtype identification if needed. Alternatively, it is recommended that others choose a name based on the entity they represent, then coordinate the use of the name within that entity.

Note that the prefix consumes some space within the item's total length of 255 octets, so the prefix should be kept as short as possible. This facility and the constrained RTCP bandwidth should not be overloaded; it is not intended to satisfy all the control communication requirements of all applications.

SDES PRIV prefixes will not be registered by IANA. If some form of the PRIV item proves to be of general utility, it should instead be assigned a regular SDES item type registered with IANA so that no prefix is required. This simplifies use and increases transmission efficiency.

Microsoft Exhibit 1019

6.5 BYE: Goodbye RTCP packet



The BYE packet indicates that one or more sources are no longer active.

version (V), padding (P), length:
As described for the SR packet (see Section 6.3.1).

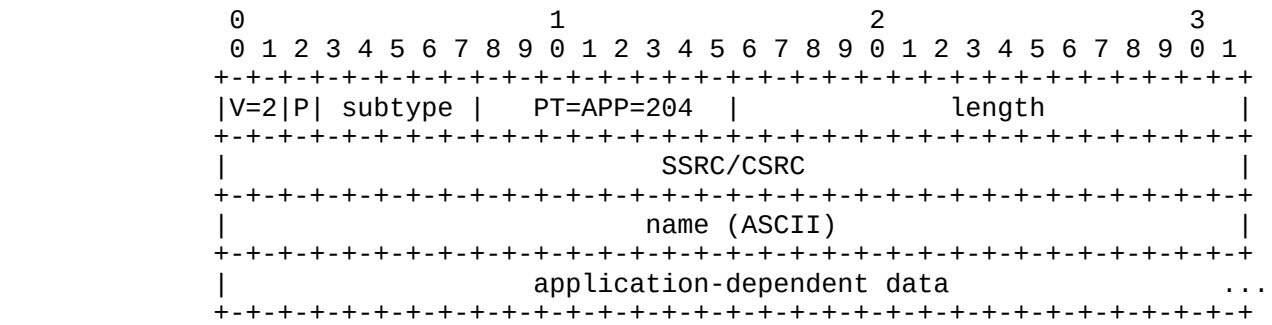
packet type (PT): 8 bits
Contains the constant 203 to identify this as an RTCP BYE packet.

source count (SC): 5 bits
The number of SSRC/CSRC identifiers included in this BYE packet. A count value of zero is valid, but useless.

If a BYE packet is received by a mixer, the mixer forwards the BYE packet with the SSRC/CSRC identifier(s) unchanged. If a mixer shuts down, it should send a BYE packet listing all contributing sources it handles, as well as its own SSRC identifier. Optionally, the BYE packet may include an 8-bit octet count followed by that many octets of text indicating the reason for leaving, e.g., "camera malfunction" or "RTP loop detected". The string has the same encoding as that described for SDES. If the string fills the packet to the next 32-bit boundary, the string is not null terminated. If not, the BYE packet is padded with null octets.

Microsoft Exhibit 1019

6.6 APP: Application-defined RTCP packet



The APP packet is intended for experimental use as new applications and new features are developed, without requiring packet type value registration. APP packets with unrecognized names should be ignored. After testing and if wider use is justified, it is recommended that each APP packet be redefined without the subtype and name fields and registered with the Internet Assigned Numbers Authority using an RTCP packet type.

version (V), padding (P), length:
As described for the SR packet (see Section 6.3.1).

subtype: 5 bits
May be used as a subtype to allow a set of APP packets to be defined under one unique name, or for any application-dependent data.

packet type (PT): 8 bits
Contains the constant 204 to identify this as an RTCP APP packet.

name: 4 octets
A name chosen by the person defining the set of APP packets to be unique with respect to other APP packets this application might receive. The application creator might choose to use the application name, and then coordinate the allocation of subtype values to others who want to define new packet types for the application. Alternatively, it is recommended that others choose a name based on the entity they represent, then coordinate the use of the name within that entity. The name is interpreted as a sequence of four ASCII characters, with uppercase and lowercase characters treated as distinct.

Microsoft Exhibit 1019

application-dependent data: variable length
Application-dependent data may or may not appear in an APP packet. It is interpreted by the application and not RTP itself. It must be a multiple of 32 bits long.

7. RTP Translators and Mixers

In addition to end systems, RTP supports the notion of "translators" and "mixers", which could be considered as "intermediate systems" at the RTP level. Although this support adds some complexity to the protocol, the need for these functions has been clearly established by experiments with multicast audio and video applications in the Internet. Example uses of translators and mixers given in Section 2.3 stem from the presence of firewalls and low bandwidth connections, both of which are likely to remain.

7.1 General Description

An RTP translator/mixer connects two or more transport-level "clouds". Typically, each cloud is defined by a common network and transport protocol (e.g., IP/UDP), multicast address or pair of unicast addresses, and transport level destination port. (Network-level protocol translators, such as IP version 4 to IP version 6, may be present within a cloud invisibly to RTP.) One system may serve as a translator or mixer for a number of RTP sessions, but each is considered a logically separate entity.

In order to avoid creating a loop when a translator or mixer is installed, the following rules must be observed:

- Each of the clouds connected by translators and mixers participating in one RTP session either must be distinct from all the others in at least one of these parameters (protocol, address, port), or must be isolated at the network level from the others.

- A derivative of the first rule is that there must not be multiple translators or mixers connected in parallel unless by some arrangement they partition the set of sources to be forwarded.

Similarly, all RTP end systems that can communicate through one or more RTP translators or mixers share the same SSRC space, that is, the SSRC identifiers must be unique among all these end systems. Section 8.2 describes the collision resolution algorithm by which SSRC identifiers are kept unique and loops are detected.

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There may be many varieties of translators and mixers designed for different purposes and applications. Some examples are to add or remove encryption, change the encoding of the data or the underlying protocols, or replicate between a multicast address and one or more unicast addresses. The distinction between translators and mixers is that a translator passes through the data streams from different sources separately, whereas a mixer combines them to form one new stream:

Translator: Forwards RTP packets with their SSRC identifier intact; this makes it possible for receivers to identify individual sources even though packets from all the sources pass through the same translator and carry the translator's network source address. Some kinds of translators will pass through the data untouched, but others may change the encoding of the data and thus the RTP data payload type and timestamp. If multiple data packets are re-encoded into one, or vice versa, a translator must assign new sequence numbers to the outgoing packets. Losses in the incoming packet stream may induce corresponding gaps in the outgoing sequence numbers. Receivers cannot detect the presence of a translator unless they know by some other means what payload type or transport address was used by the original source.

Mixer: Receives streams of RTP data packets from one or more sources, possibly changes the data format, combines the streams in some manner and then forwards the combined stream. Since the timing among multiple input sources will not generally be synchronized, the mixer will make timing adjustments among the streams and generate its own timing for the combined stream, so it is the synchronization source. Thus, all data packets forwarded by a mixer will be marked with the mixer's own SSRC identifier. In order to preserve the identity of the original sources contributing to the mixed packet, the mixer should insert their SSRC identifiers into the CSRC identifier list following the fixed RTP header of the packet. A mixer that is also itself a contributing source for some packet should explicitly include its own SSRC identifier in the CSRC list for that packet.

For some applications, it may be acceptable for a mixer not to identify sources in the CSRC list. However, this introduces the danger that loops involving those sources could not be detected.

The advantage of a mixer over a translator for applications like audio is that the output bandwidth is limited to that of one source even when multiple sources are active on the input side. This may be important for low-bandwidth links. The disadvantage is that receivers on the output side don't have any control over which sources are

Microsoft Exhibit 1019

passed through or muted, unless some mechanism is implemented for remote control of the mixer. The regeneration of synchronization information by mixers also means that receivers can't do inter-media synchronization of the original streams. A multi-media mixer could do it.

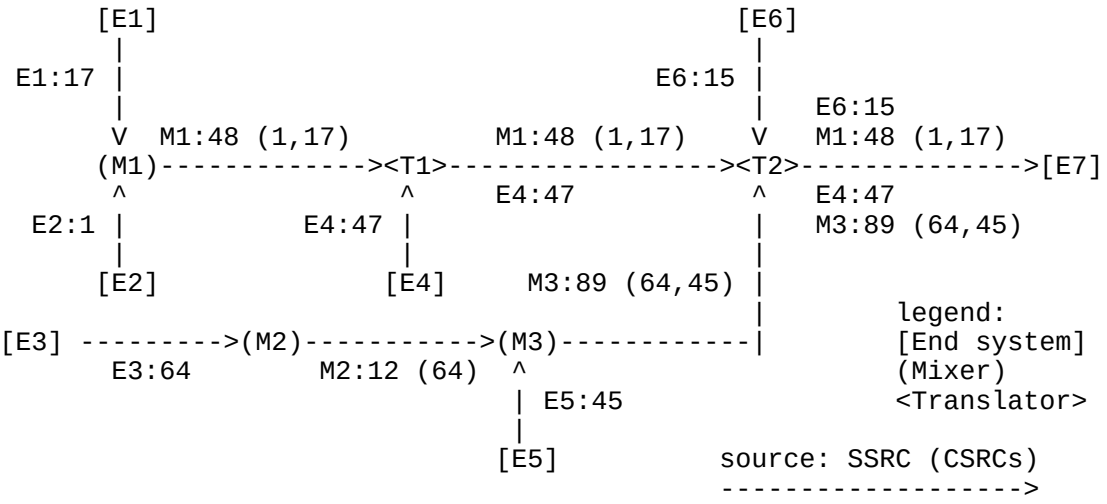

Figure 3: Sample RTP network with end systems, mixers and translators

A collection of mixers and translators is shown in Figure 3 to illustrate their effect on SSRC and CSRC identifiers. In the figure, end systems are shown as rectangles (named E), translators as triangles (named T) and mixers as ovals (named M). The notation "M1: 48(1,17)" designates a packet originating a mixer M1, identified with M1's (random) SSRC value of 48 and two CSRC identifiers, 1 and 17, copied from the SSRC identifiers of packets from E1 and E2.

7.2 RTCP Processing in Translators

In addition to forwarding data packets, perhaps modified, translators and mixers must also process RTCP packets. In many cases, they will take apart the compound RTCP packets received from end systems to aggregate SDES information and to modify the SR or RR packets. Retransmission of this information may be triggered by the packet arrival or by the RTCP interval timer of the translator or mixer itself.

A translator that does not modify the data packets, for example one that just replicates between a multicast address and a unicast address, may simply forward RTCP packets unmodified as well. A

Microsoft Exhibit 1019

translator that transforms the payload in some way must make corresponding transformations in the SR and RR information so that it still reflects the characteristics of the data and the reception quality. These translators must not simply forward RTCP packets. In general, a translator should not aggregate SR and RR packets from different sources into one packet since that would reduce the accuracy of the propagation delay measurements based on the LSR and DLSR fields.

SR sender information: A translator does not generate its own sender information, but forwards the SR packets received from one cloud to the others. The SSRC is left intact but the sender information must be modified if required by the translation. If a translator changes the data encoding, it must change the "sender's byte count" field. If it also combines several data packets into one output packet, it must change the "sender's packet count" field. If it changes the timestamp frequency, it must change the "RTP timestamp" field in the SR packet.

SR/RR reception report blocks: A translator forwards reception reports received from one cloud to the others. Note that these flow in the direction opposite to the data. The SSRC is left intact. If a translator combines several data packets into one output packet, and therefore changes the sequence numbers, it must make the inverse manipulation for the packet loss fields and the "extended last sequence number" field. This may be complex. In the extreme case, there may be no meaningful way to translate the reception reports, so the translator may pass on no reception report at all or a synthetic report based on its own reception. The general rule is to do what makes sense for a particular translation.

A translator does not require an SSRC identifier of its own, but may choose to allocate one for the purpose of sending reports about what it has received. These would be sent to all the connected clouds, each corresponding to the translation of the data stream as sent to that cloud, since reception reports are normally multicast to all participants.

SDES: Translators typically forward without change the SDES information they receive from one cloud to the others, but may, for example, decide to filter non-CNAME SDES information if bandwidth is limited. The CNAMEs must be forwarded to allow SSRC identifier collision detection to work. A translator that generates its own RR packets must send SDES CNAME information about itself to the same clouds that it sends those RR packets.

Microsoft Exhibit 1019

BYE: Translators forward BYE packets unchanged. Translators with their own SSRC should generate BYE packets with that SSRC identifier if they are about to cease forwarding packets.

APP: Translators forward APP packets unchanged.

7.3 RTCP Processing in Mixers

Since a mixer generates a new data stream of its own, it does not pass through SR or RR packets at all and instead generates new information for both sides.

SR sender information: A mixer does not pass through sender information from the sources it mixes because the characteristics of the source streams are lost in the mix. As a synchronization source, the mixer generates its own SR packets with sender information about the mixed data stream and sends them in the same direction as the mixed stream.

SR/RR reception report blocks: A mixer generates its own reception reports for sources in each cloud and sends them out only to the same cloud. It does not send these reception reports to the other clouds and does not forward reception reports from one cloud to the others because the sources would not be SSRCs there (only CSRCs).

SDES: Mixers typically forward without change the SDES information they receive from one cloud to the others, but may, for example, decide to filter non-CNAME SDES information if bandwidth is limited. The CNAMEs must be forwarded to allow SSRC identifier collision detection to work. (An identifier in a CSRC list generated by a mixer might collide with an SSRC identifier generated by an end system.) A mixer must send SDES CNAME information about itself to the same clouds that it sends SR or RR packets.

Since mixers do not forward SR or RR packets, they will typically be extracting SDES packets from a compound RTCP packet. To minimize overhead, chunks from the SDES packets may be aggregated into a single SDES packet which is then stacked on an SR or RR packet originating from the mixer. The RTCP packet rate may be different on each side of the mixer.

A mixer that does not insert CSRC identifiers may also refrain from forwarding SDES CNAMEs. In this case, the SSRC identifier spaces in the two clouds are independent. As mentioned earlier, this mode of operation creates a danger that loops can’t be detected.

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BYE: Mixers need to forward BYE packets. They should generate BYE packets with their own SSRC identifiers if they are about to cease forwarding packets.

APP: The treatment of APP packets by mixers is application-specific.

7.4 Cascaded Mixers

An RTP session may involve a collection of mixers and translators as shown in Figure 3. If two mixers are cascaded, such as M2 and M3 in the figure, packets received by a mixer may already have been mixed and may include a CSRC list with multiple identifiers. The second mixer should build the CSRC list for the outgoing packet using the CSRC identifiers from already-mixed input packets and the SSRC identifiers from unmixed input packets. This is shown in the output arc from mixer M3 labeled M3:89(64,45) in the figure. As in the case of mixers that are not cascaded, if the resulting CSRC list has more than 15 identifiers, the remainder cannot be included.

8. SSRC Identifier Allocation and Use

The SSRC identifier carried in the RTP header and in various fields of RTCP packets is a random 32-bit number that is required to be globally unique within an RTP session. It is crucial that the number be chosen with care in order that participants on the same network or starting at the same time are not likely to choose the same number.

It is not sufficient to use the local network address (such as an IPv4 address) for the identifier because the address may not be unique. Since RTP translators and mixers enable interoperation among multiple networks with different address spaces, the allocation patterns for addresses within two spaces might result in a much higher rate of collision than would occur with random allocation.

Multiple sources running on one host would also conflict.

It is also not sufficient to obtain an SSRC identifier simply by calling random() without carefully initializing the state. An example of how to generate a random identifier is presented in Appendix A.6.

8.1 Probability of Collision

Since the identifiers are chosen randomly, it is possible that two or more sources will choose the same number. Collision occurs with the highest probability when all sources are started simultaneously, for example when triggered automatically by some session management event. If N is the number of sources and L the length of the identifier (here, 32 bits), the probability that two sources

Microsoft Exhibit 1019

independently pick the same value can be approximated for large N [20] as 1 - exp(-N**2 / 2**(L+1)). For N=1000, the probability is roughly 10**-4.

The typical collision probability is much lower than the worst-case above. When one new source joins an RTP session in which all the other sources already have unique identifiers, the probability of collision is just the fraction of numbers used out of the space. Again, if N is the number of sources and L the length of the identifier, the probability of collision is N / 2**L. For N=1000, the probability is roughly 2*10**-7.

The probability of collision is further reduced by the opportunity for a new source to receive packets from other participants before sending its first packet (either data or control). If the new source keeps track of the other participants (by SSRC identifier), then before transmitting its first packet the new source can verify that its identifier does not conflict with any that have been received, or else choose again.

8.2 Collision Resolution and Loop Detection

Although the probability of SSRC identifier collision is low, all RTP implementations must be prepared to detect collisions and take the appropriate actions to resolve them. If a source discovers at any time that another source is using the same SSRC identifier as its own, it must send an RTCP BYE packet for the old identifier and choose another random one. If a receiver discovers that two other sources are colliding, it may keep the packets from one and discard the packets from the other when this can be detected by different source transport addresses or CNAMEs. The two sources are expected to resolve the collision so that the situation doesn’t last.

Because the random identifiers are kept globally unique for each RTP session, they can also be used to detect loops that may be introduced by mixers or translators. A loop causes duplication of data and control information, either unmodified or possibly mixed, as in the following examples:

- A translator may incorrectly forward a packet to the same multicast group from which it has received the packet, either directly or through a chain of translators. In that case, the same packet appears several times, originating from different network sources.

- Two translators incorrectly set up in parallel, i.e., with the same multicast groups on both sides, would both forward packets from one multicast group to the other. Unidirectional

Microsoft Exhibit 1019

translators would produce two copies; bidirectional translators would form a loop.

- o A mixer can close a loop by sending to the same transport destination upon which it receives packets, either directly or through another mixer or translator. In this case a source might show up both as an SSRC on a data packet and a CSRC in a mixed data packet.

A source may discover that its own packets are being looped, or that packets from another source are being looped (a third-party loop).

Both loops and collisions in the random selection of a source identifier result in packets arriving with the same SSRC identifier but a different source transport address, which may be that of the end system originating the packet or an intermediate system. Consequently, if a source changes its source transport address, it must also choose a new SSRC identifier to avoid being interpreted as a looped source. Loops or collisions occurring on the far side of a translator or mixer cannot be detected using the source transport address if all copies of the packets go through the translator or mixer, however collisions may still be detected when chunks from two RTCP SDES packets contain the same SSRC identifier but different CNAMEs.

To detect and resolve these conflicts, an RTP implementation must include an algorithm similar to the one described below. It ignores packets from a new source or loop that collide with an established source. It resolves collisions with the participant's own SSRC identifier by sending an RTCP BYE for the old identifier and choosing a new one. However, when the collision was induced by a loop of the participant's own packets, the algorithm will choose a new identifier only once and thereafter ignore packets from the looping source transport address. This is required to avoid a flood of BYE packets.

This algorithm depends upon the source transport address being the same for both RTP and RTCP packets from a source. The algorithm would require modifications to support applications that don't meet this constraint.

This algorithm requires keeping a table indexed by source identifiers and containing the source transport address from which the identifier was (first) received, along with other state for that source. Each SSRC or CSRC identifier received in a data or control packet is looked up in this table in order to process that data or control information. For control packets, each element with its own SSRC, for example an SDES chunk, requires a separate lookup. (The SSRC in a reception report block is an exception.) If the SSRC or CSRC is not

Microsoft Exhibit 1019

found, a new entry is created. These table entries are removed when an RTCP BYE packet is received with the corresponding SSRC, or after no packets have arrived for a relatively long time (see Section 6.2.1).

In order to track loops of the participant's own data packets, it is also necessary to keep a separate list of source transport addresses (not identifiers) that have been found to be conflicting. Note that this should be a short list, usually empty. Each element in this list stores the source address plus the time when the most recent conflicting packet was received. An element may be removed from the list when no conflicting packet has arrived from that source for a time on the order of 10 RTCP report intervals (see Section 6.2).

For the algorithm as shown, it is assumed that the participant's own source identifier and state are included in the source identifier table. The algorithm could be restructured to first make a separate comparison against the participant's own source identifier.

```
    IF the SSRC or CSRC identifier is not found in the source
       identifier table:
    THEN create a new entry storing the source transport address
         and the SSRC or CSRC along with other state.
         CONTINUE with normal processing.

    (identifier is found in the table)

    IF the source transport address from the packet matches
       the one saved in the table entry for this identifier:
    THEN CONTINUE with normal processing.

    (an identifier collision or a loop is indicated)

    IF the source identifier is not the participant's own:
    THEN IF the source identifier is from an RTCP SDES chunk
            containing a CNAME item that differs from the CNAME
            in the table entry:
         THEN (optionally) count a third-party collision.
         ELSE (optionally) count a third-party loop.
         ABORT processing of data packet or control element.

    (a collision or loop of the participant's own data)

    IF the source transport address is found in the list of
      conflicting addresses:
    THEN IF the source identifier is not from an RTCP SDES chunk
            containing a CNAME item OR if that CNAME is the
            participant's own:
```


Microsoft Exhibit 1019

```
            THEN (optionally) count occurrence of own traffic looped.
                 mark current time in conflicting address list entry.
                 ABORT processing of data packet or control element.
        log occurrence of a collision.
        create a new entry in the conflicting address list and
        mark current time.
        send an RTCP BYE packet with the old SSRC identifier.
        choose a new identifier.
        create a new entry in the source identifier table with the
          old SSRC plus the source transport address from the packet
          being processed.
        CONTINUE with normal processing.
```

In this algorithm, packets from a newly conflicting source address will be ignored and packets from the original source will be kept. (If the original source was through a mixer and later the same source is received directly, the receiver may be well advised to switch unless other sources in the mix would be lost.) If no packets arrive from the original source for an extended period, the table entry will be timed out and the new source will be able to take over. This might occur if the original source detects the collision and moves to a new source identifier, but in the usual case an RTCP BYE packet will be received from the original source to delete the state without having to wait for a timeout.

When a new SSRC identifier is chosen due to a collision, the candidate identifier should first be looked up in the source identifier table to see if it was already in use by some other source. If so, another candidate should be generated and the process repeated.

A loop of data packets to a multicast destination can cause severe network flooding. All mixers and translators are required to implement a loop detection algorithm like the one here so that they can break loops. This should limit the excess traffic to no more than one duplicate copy of the original traffic, which may allow the session to continue so that the cause of the loop can be found and fixed. However, in extreme cases where a mixer or translator does not properly break the loop and high traffic levels result, it may be necessary for end systems to cease transmitting data or control packets entirely. This decision may depend upon the application. An error condition should be indicated as appropriate. Transmission might be attempted again periodically after a long, random time (on the order of minutes).

Microsoft Exhibit 1019

9. Security

Lower layer protocols may eventually provide all the security services that may be desired for applications of RTP, including authentication, integrity, and confidentiality. These services have recently been specified for IP. Since the need for a confidentiality service is well established in the initial audio and video applications that are expected to use RTP, a confidentiality service is defined in the next section for use with RTP and RTCP until lower layer services are available. The overhead on the protocol for this service is low, so the penalty will be minimal if this service is obsoleted by lower layer services in the future.

Alternatively, other services, other implementations of services and other algorithms may be defined for RTP in the future if warranted. The selection presented here is meant to simplify implementation of interoperable, secure applications and provide guidance to implementors. No claim is made that the methods presented here are appropriate for a particular security need. A profile may specify which services and algorithms should be offered by applications, and may provide guidance as to their appropriate use.

Key distribution and certificates are outside the scope of this document.

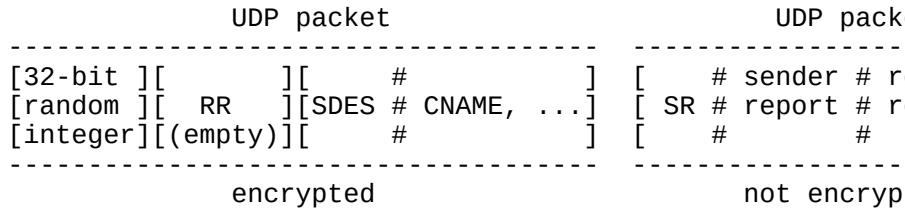
9.1 Confidentiality

Confidentiality means that only the intended receiver(s) can decode the received packets; for others, the packet contains no useful information. Confidentiality of the content is achieved by encryption.

When encryption of RTP or RTCP is desired, all the octets that will be encapsulated for transmission in a single lower-layer packet are encrypted as a unit. For RTCP, a 32-bit random number is prepended to the unit before encryption to deter known plaintext attacks. For RTP, no prefix is required because the sequence number and timestamp fields are initialized with random offsets.

For RTCP, it is allowed to split a compound RTCP packet into two lower-layer packets, one to be encrypted and one to be sent in the clear. For example, SDES information might be encrypted while reception reports were sent in the clear to accommodate third-party monitors that are not privy to the encryption key. In this example, depicted in Fig. 4, the SDES information must be appended to an RR packet with no reports (and the encrypted) to satisfy the requirement that all compound RTCP packets begin with an SR or RR packet.

Microsoft Exhibit 1019



#: SSRC

Figure 4: Encrypted and non-encrypted RTCP packets

The presence of encryption and the use of the correct key are confirmed by the receiver through header or payload validity checks. Examples of such validity checks for RTP and RTCP headers are given in Appendices A.1 and A.2.

The default encryption algorithm is the Data Encryption Standard (DES) algorithm in cipher block chaining (CBC) mode, as described in Section 1.1 of RFC 1423 [21], except that padding to a multiple of 8 octets is indicated as described for the P bit in Section 5.1. The initialization vector is zero because random values are supplied in the RTP header or by the random prefix for compound RTCP packets. For details on the use of CBC initialization vectors, see [22]. Implementations that support encryption should always support the DES algorithm in CBC mode as the default to maximize interoperability. This method is chosen because it has been demonstrated to be easy and practical to use in experimental audio and video tools in operation on the Internet. Other encryption algorithms may be specified dynamically for a session by non-RTP means.

As an alternative to encryption at the RTP level as described above, profiles may define additional payload types for encrypted encodings. Those encodings must specify how padding and other aspects of the encryption should be handled. This method allows encrypting only the data while leaving the headers in the clear for applications where that is desired. It may be particularly useful for hardware devices that will handle both decryption and decoding.

9.2 Authentication and Message Integrity

Authentication and message integrity are not defined in the current specification of RTP since these services would not be directly feasible without a key management infrastructure. It is expected that authentication and integrity services will be provided by lower layer protocols in the future.

Microsoft Exhibit 1019

10. RTP over Network and Transport Protocols

This section describes issues specific to carrying RTP packets within particular network and transport protocols. The following rules apply unless superseded by protocol-specific definitions outside this specification.

RTP relies on the underlying protocol(s) to provide demultiplexing of RTP data and RTCP control streams. For UDP and similar protocols, RTP uses an even port number and the corresponding RTCP stream uses the next higher (odd) port number. If an application is supplied with an odd number for use as the RTP port, it should replace this number with the next lower (even) number.

RTP data packets contain no length field or other delineation, therefore RTP relies on the underlying protocol(s) to provide a length indication. The maximum length of RTP packets is limited only by the underlying protocols.

If RTP packets are to be carried in an underlying protocol that provides the abstraction of a continuous octet stream rather than messages (packets), an encapsulation of the RTP packets must be defined to provide a framing mechanism. Framing is also needed if the underlying protocol may contain padding so that the extent of the RTP payload cannot be determined. The framing mechanism is not defined here.

A profile may specify a framing method to be used even when RTP is carried in protocols that do provide framing in order to allow carrying several RTP packets in one lower-layer protocol data unit, such as a UDP packet. Carrying several RTP packets in one network or transport packet reduces header overhead and may simplify synchronization between different streams.

11. Summary of Protocol Constants

This section contains a summary listing of the constants defined in this specification.

The RTP payload type (PT) constants are defined in profiles rather than this document. However, the octet of the RTP header which contains the marker bit(s) and payload type must avoid the reserved values 200 and 201 (decimal) to distinguish RTP packets from the RTCP SR and RR packet types for the header validation procedure described in Appendix A.1. For the standard definition of one marker bit and a 7-bit payload type field as shown in this specification, this restriction means that payload types 72 and 73 are reserved.

Microsoft Exhibit 1019

11.1 RTCP packet types

abbrev.	name	value
SR	sender report	200
RR	receiver report	201
SDES	source description	202
BYE	goodbye	203
APP	application-defined	204

These type values were chosen in the range 200-204 for improved header validity checking of RTCP packets compared to RTP packets or other unrelated packets. When the RTCP packet type field is compared to the corresponding octet of the RTP header, this range corresponds to the marker bit being 1 (which it usually is not in data packets) and to the high bit of the standard payload type field being 1 (since the static payload types are typically defined in the low half). This range was also chosen to be some distance numerically from 0 and 255 since all-zeros and all-ones are common data patterns.

Since all compound RTCP packets must begin with SR or RR, these codes were chosen as an even/odd pair to allow the RTCP validity check to test the maximum number of bits with mask and value.

Other constants are assigned by IANA. Experimenters are encouraged to register the numbers they need for experiments, and then unregister those which prove to be unneeded.

11.2 SDES types

abbrev.	name	value
END	end of SDES list	0
CNAME	canonical name	1
NAME	user name	2
EMAIL	user's electronic mail address	3
PHONE	user's phone number	4
LOC	geographic user location	5
TOOL	name of application or tool	6
NOTE	notice about the source	7
PRIV	private extensions	8

Other constants are assigned by IANA. Experimenters are encouraged to register the numbers they need for experiments, and then unregister those which prove to be unneeded.

Microsoft Exhibit 1019

12. RTP Profiles and Payload Format Specifications

A complete specification of RTP for a particular application will require one or more companion documents of two types described here: profiles, and payload format specifications.

RTP may be used for a variety of applications with somewhat differing requirements. The flexibility to adapt to those requirements is provided by allowing multiple choices in the main protocol specification, then selecting the appropriate choices or defining extensions for a particular environment and class of applications in a separate profile document. Typically an application will operate under only one profile so there is no explicit indication of which profile is in use. A profile for audio and video applications may be found in the companion Internet-Draft draft-ietf-avt-profile for

The second type of companion document is a payload format specification, which defines how a particular kind of payload data, such as H.261 encoded video, should be carried in RTP. These documents are typically titled "RTP Payload Format for XYZ Audio/Video Encoding". Payload formats may be useful under multiple profiles and may therefore be defined independently of any particular profile. The profile documents are then responsible for assigning a default mapping of that format to a payload type value if needed.

Within this specification, the following items have been identified for possible definition within a profile, but this list is not meant to be exhaustive:

RTP data header: The octet in the RTP data header that contains the marker bit and payload type field may be redefined by a profile to suit different requirements, for example with more or fewer marker bits (Section 5.3).

Payload types: Assuming that a payload type field is included, the profile will usually define a set of payload formats (e.g., media encodings) and a default static mapping of those formats to payload type values. Some of the payload formats may be defined by reference to separate payload format specifications. For each payload type defined, the profile must specify the RTP timestamp clock rate to be used (Section 5.1).

RTP data header additions: Additional fields may be appended to the fixed RTP data header if some additional functionality is required across the profile’s class of applications independent of payload type (Section 5.3).

Microsoft Exhibit 1019

RTP data header extensions: The contents of the first 16 bits of the RTP data header extension structure must be defined if use of that mechanism is to be allowed under the profile for implementation-specific extensions (Section 5.3.1).

RTCP packet types: New application-class-specific RTCP packet types may be defined and registered with IANA.

RTCP report interval: A profile should specify that the values suggested in Section 6.2 for the constants employed in the calculation of the RTCP report interval will be used. Those are the RTCP fraction of session bandwidth, the minimum report interval, and the bandwidth split between senders and receivers. A profile may specify alternate values if they have been demonstrated to work in a scalable manner.

SR/RR extension: An extension section may be defined for the RTCP SR and RR packets if there is additional information that should be reported regularly about the sender or receivers (Section 6.3.3).

SDES use: The profile may specify the relative priorities for RTCP SDES items to be transmitted or excluded entirely (Section 6.2.2); an alternate syntax or semantics for the CNAME item (Section 6.4.1); the format of the LOC item (Section 6.4.5); the semantics and use of the NOTE item (Section 6.4.7); or new SDES item types to be registered with IANA.

Security: A profile may specify which security services and algorithms should be offered by applications, and may provide guidance as to their appropriate use (Section 9).

String-to-key mapping: A profile may specify how a user-provided password or pass phrase is mapped into an encryption key.

Underlying protocol: Use of a particular underlying network or transport layer protocol to carry RTP packets may be required.

Transport mapping: A mapping of RTP and RTCP to transport-level addresses, e.g., UDP ports, other than the standard mapping defined in Section 10 may be specified.

Encapsulation: An encapsulation of RTP packets may be defined to allow multiple RTP data packets to be carried in one lower-layer packet or to provide framing over underlying protocols that do not already do so (Section 10).

Microsoft Exhibit 1019

It is not expected that a new profile will be required for every application. Within one application class, it would be better to extend an existing profile rather than make a new one in order to facilitate interoperation among the applications since each will typically run under only one profile. Simple extensions such as the definition of additional payload type values or RTCP packet types may be accomplished by registering them through the Internet Assigned Numbers Authority and publishing their descriptions in an addendum to the profile or in a payload format specification.

Microsoft Exhibit 1019

A. Algorithms

We provide examples of C code for aspects of RTP sender and receiver algorithms. There may be other implementation methods that are faster in particular operating environments or have other advantages. These implementation notes are for informational purposes only and are meant to clarify the RTP specification.

The following definitions are used for all examples; for clarity and brevity, the structure definitions are only valid for 32-bit big-endian (most significant octet first) architectures. Bit fields are assumed to be packed tightly in big-endian bit order, with no additional padding. Modifications would be required to construct a portable implementation.

```
/*
 * rtp.h  --  RTP header file (RFC XXXX)
 */
#include <sys/types.h>

/*
 * The type definitions below are valid for 32-bit architectures and
 * may have to be adjusted for 16- or 64-bit architectures.
 */
typedef unsigned char  u_int8;
typedef unsigned short u_int16;
typedef unsigned int   u_int32;
typedef          short int16;

/*
 * Current protocol version.
 */
#define RTP_VERSION    2

#define RTP_SEQ_MOD (1<<16)
#define RTP_MAX_SDES 255      /* maximum text length for SDES */

typedef enum {
    RTCP_SR   = 200,
    RTCP_RR   = 201,
    RTCP_SDES = 202,
    RTCP_BYE  = 203,
    RTCP_APP  = 204
} rtcp_type_t;

typedef enum {
    RTCP_SDES_END   = 0,
    RTCP_SDES_CNAME = 1,
```


Microsoft Exhibit 1019

```
        RTCP_SDES_NAME  = 2,
        RTCP_SDES_EMAIL = 3,
        RTCP_SDES_PHONE = 4,
        RTCP_SDES_LOC   = 5,
        RTCP_SDES_TOOL  = 6,
        RTCP_SDES_NOTE  = 7,
        RTCP_SDES_PRIV  = 8
    } rtcp_sdes_type_t;

    /*
     * RTP data header
     */
    typedef struct {
        unsigned int version:2;   /* protocol version */
        unsigned int p:1;         /* padding flag */
        unsigned int x:1;         /* header extension flag */
        unsigned int cc:4;        /* CSRC count */
        unsigned int m:1;         /* marker bit */
        unsigned int pt:7;        /* payload type */
        u_int16 seq;              /* sequence number */
        u_int32 ts;               /* timestamp */
        u_int32 ssrc;             /* synchronization source */
        u_int32 csrc[1];          /* optional CSRC list */
    } rtp_hdr_t;

    /*
     * RTCP common header word
     */
    typedef struct {
        unsigned int version:2;   /* protocol version */
        unsigned int p:1;         /* padding flag */
        unsigned int count:5;     /* varies by packet type */
        unsigned int pt:8;        /* RTCP packet type */
        u_int16 length;           /* pkt len in words, w/o this word */
    } rtcp_common_t;

    /*
     * Big-endian mask for version, padding bit and packet type pair
     */
    #define RTCP_VALID_MASK (0xc000 | 0x2000 | 0xfe)
    #define RTCP_VALID_VALUE ((RTP_VERSION << 14) | RTCP_SR)

    /*
     * Reception report block
     */
    typedef struct {
        u_int32 ssrc;             /* data source being reported */
        unsigned int fraction:8;  /* fraction lost since last SR/RR */
```


Microsoft Exhibit 1019

```
        int lost:24;             /* cumul. no. pkts lost (signed!) */
        u_int32 last_seq;        /* extended last seq. no. received */
        u_int32 jitter;          /* interarrival jitter */
        u_int32 lsr;             /* last SR packet from this source */
        u_int32 dlsr;            /* delay since last SR packet */
    } rtcp_rr_t;

    /*
     * SDES item
     */
    typedef struct {
        u_int8 type;             /* type of item (rtcp_sdes_type_t) */
        u_int8 length;           /* length of item (in octets) */
        char data[1];            /* text, not null-terminated */
    } rtcp_sdes_item_t;

    /*
     * One RTCP packet
     */
    typedef struct {
        rtcp_common_t common;    /* common header */
        union {
            /* sender report (SR) */
            struct {
                u_int32 ssrc;    /* sender generating this report */
                u_int32 ntp_sec;  /* NTP timestamp */
                u_int32 ntp_frac;
                u_int32 rtp_ts;  /* RTP timestamp */
                u_int32 psent;   /* packets sent */
                u_int32 osent;   /* octets sent */
                rtcp_rr_t rr[1];  /* variable-length list */
            } sr;

            /* reception report (RR) */
            struct {
                u_int32 ssrc;    /* receiver generating this report */
                rtcp_rr_t rr[1];  /* variable-length list */
            } rr;

            /* source description (SDES) */
            struct rtcp_sdes {
                u_int32 src;     /* first SSRC/CSRC */
                rtcp_sdes_item_t item[1]; /* list of SDES items */
            } sdes;

            /* BYE */
            struct {
                u_int32 src[1];  /* list of sources */
```


Microsoft Exhibit 1019

```
            /* can't express trailing text for reason */
        } bye;
    } r;
} rtcp_t;

typedef struct rtcp_sdes rtcp_sdes_t;

/*
 * Per-source state information
 */
typedef struct {
    u_int16 max_seq;        /* highest seq. number seen */
    u_int32 cycles;         /* shifted count of seq. number cycles */
    u_int32 base_seq;       /* base seq number */
    u_int32 bad_seq;        /* last 'bad' seq number + 1 */
    u_int32 probation;      /* sequ. packets till source is valid */
    u_int32 received;       /* packets received */
    u_int32 expected_prior; /* packet expected at last interval */
    u_int32 received_prior; /* packet received at last interval */
    u_int32 transit;        /* relative trans time for prev pkt */
    u_int32 jitter;         /* estimated jitter */
    /* ... */
} source;
```

A.1 RTP Data Header Validity Checks

An RTP receiver should check the validity of the RTP header on incoming packets since they might be encrypted or might be from a different application that happens to be misaddressed. Similarly, if encryption is enabled, the header validity check is needed to verify that incoming packets have been correctly decrypted, although a failure of the header validity check (e.g., unknown payload type) may not necessarily indicate decryption failure.

Only weak validity checks are possible on an RTP data packet from a source that has not been heard before:

- RTP version field must equal 2.
- The payload type must be known, in particular it must not be equal to SR or RR.
- If the P bit is set, then the last octet of the packet must contain a valid octet count, in particular, less than the total packet length minus the header size.
- The X bit must be zero if the profile does not specify that the header extension mechanism may be used. Otherwise, the

Microsoft Exhibit 1019

extension length field must be less than the total packet size minus the fixed header length and padding.

o The length of the packet must be consistent with CC and payload type (if payloads have a known length).

The last three checks are somewhat complex and not always possible, leaving only the first two which total just a few bits. If the SSRC identifier in the packet is one that has been received before, then the packet is probably valid and checking if the sequence number is in the expected range provides further validation. If the SSRC identifier has not been seen before, then data packets carrying that identifier may be considered invalid until a small number of them arrive with consecutive sequence numbers.

The routine update_seq shown below ensures that a source is declared valid only after MIN_SEQUENTIAL packets have been received in sequence. It also validates the sequence number seq of a newly received packet and updates the sequence state for the packet's source in the structure to which s points.

When a new source is heard for the first time, that is, its SSRC identifier is not in the table (see Section 8.2), and the per-source state is allocated for it, s->probation should be set to the number of sequential packets required before declaring a source valid (parameter MIN_SEQUENTIAL) and s->max_seq initialized to seq-1 s->probation marks the source as not yet valid so the state may be discarded after a short timeout rather than a long one, as discussed in Section 6.2.1.

After a source is considered valid, the sequence number is considered valid if it is no more than MAX_DROPOUT ahead of s->max_seq nor more than MAX_MISORDER behind. If the new sequence number is ahead of max_seq modulo the RTP sequence number range (16 bits), but is smaller than max_seq , it has wrapped around and the (shifted) count of sequence number cycles is incremented. A value of one is returned to indicate a valid sequence number.

Otherwise, the value zero is returned to indicate that the validation failed, and the bad sequence number is stored. If the next packet received carries the next higher sequence number, it is considered the valid start of a new packet sequence presumably caused by an extended dropout or a source restart. Since multiple complete sequence number cycles may have been missed, the packet loss statistics are reset.

Typical values for the parameters are shown, based on a maximum misordering time of 2 seconds at 50 packets/second and a maximum

Microsoft Exhibit 1019

dropout of 1 minute. The dropout parameter MAX_DROPOUT should be a small fraction of the 16-bit sequence number space to give a reasonable probability that new sequence numbers after a restart will not fall in the acceptable range for sequence numbers from before the restart.

```
void init_seq(source *s, u_int16 seq)
{
    s->base_seq = seq - 1;
    s->max_seq = seq;
    s->bad_seq = RTP_SEQ_MOD + 1;
    s->cycles = 0;
    s->received = 0;
    s->received_prior = 0;
    s->expected_prior = 0;
    /* other initialization */
}

int update_seq(source *s, u_int16 seq)
{
    u_int16 udelta = seq - s->max_seq;
    const int MAX_DROPOUT = 3000;
    const int MAX_MISORDER = 100;
    const int MIN_SEQUENTIAL = 2;

    /*
     * Source is not valid until MIN_SEQUENTIAL packets with
     * sequential sequence numbers have been received.
     */
    if (s->probation) {
        /* packet is in sequence */
        if (seq == s->max_seq + 1) {
            s->probation--;
            s->max_seq = seq;
            if (s->probation == 0) {
                init_seq(s, seq);
                s->received++;
                return 1;
            }
        } else {
            s->probation = MIN_SEQUENTIAL - 1;
            s->max_seq = seq;
        }
        return 0;
    } else if (udelta < MAX_DROPOUT) {
        /* in order, with permissible gap */
        if (seq < s->max_seq) {
            /*
```


Microsoft Exhibit 1019

```
                 * Sequence number wrapped - count another 64K cycle.
                 */
                s->cycles += RTP_SEQ_MOD;
            }
            s->max_seq = seq;
        } else if (udelta <= RTP_SEQ_MOD - MAX_MISORDER) {
            /* the sequence number made a very large jump */
            if (seq == s->bad_seq) {
                /*
                 * Two sequential packets -- assume that the other side
                 * restarted without telling us so just re-sync
                 * (i.e., pretend this was the first packet).
                 */
                init_seq(s, seq);
            }
            else {
                s->bad_seq = (seq + 1) & (RTP_SEQ_MOD-1);
                return 0;
            }
        } else {
            /* duplicate or reordered packet */
        }
        s->received++;
        return 1;
    }
```

The validity check can be made stronger requiring more than two packets in sequence. The disadvantages are that a larger number of initial packets will be discarded and that high packet loss rates could prevent validation. However, because the RTCP header validation is relatively strong, if an RTCP packet is received from a source before the data packets, the count could be adjusted so that only two packets are required in sequence. If initial data loss for a few seconds can be tolerated, an application could choose to discard all data packets from a source until a valid RTCP packet has been received from that source.

Depending on the application and encoding, algorithms may exploit additional knowledge about the payload format for further validation. For payload types where the timestamp increment is the same for all packets, the timestamp values can be predicted from the previous packet received from the same source using the sequence number difference (assuming no change in payload type).

A strong "fast-path" check is possible since with high probability the first four octets in the header of a newly received RTP data packet will be just the same as that of the previous packet from the same SSRC except that the sequence number will have increased by one.

Microsoft Exhibit 1019

Similarly, a single-entry cache may be used for faster SSRC lookups in applications where data is typically received from one source at a time.

A.2 RTCP Header Validity Checks

The following checks can be applied to RTCP packets.

- o RTP version field must equal 2.
- o The payload type field of the first RTCP packet in a compound packet must be equal to SR or RR.
- o The padding bit (P) should be zero for the first packet of a compound RTCP packet because only the last should possibly need padding.
- o The length fields of the individual RTCP packets must total to the overall length of the compound RTCP packet as received. This is a fairly strong check.

The code fragment below performs all of these checks. The packet type is not checked for subsequent packets since unknown packet types may be present and should be ignored.

```
u_int32 len;        /* length of compound RTCP packet in words */
rtcp_t *r;          /* RTCP header */
rtcp_t *end;        /* end of compound RTCP packet */

if ((*(u_int16 *)r & RTCP_VALID_MASK) != RTCP_VALID_VALUE) {
    /* something wrong with packet format */
}
end = (rtcp_t *)((u_int32 *)r + len);

do r = (rtcp_t *)((u_int32 *)r + r->common.length + 1);
while (r < end && r->common.version == 2);

if (r != end) {
    /* something wrong with packet format */
}
```

A.3 Determining the Number of RTP Packets Expected and Lost

In order to compute packet loss rates, the number of packets expected and actually received from each source needs to be known, using per-source state information defined in struct source referenced via pointer s in the code below. The number of packets received is simply the count of packets as they arrive, including any late or duplicate

Microsoft Exhibit 1019

packets. The number of packets expected can be computed by the receiver as the difference between the highest sequence number received (s->max_seq) and the first sequence number received (s->base_seq). Since the sequence number is only 16 bits and will wrap around, it is necessary to extend the highest sequence number with the (shifted) count of sequence number wraparounds (s->cycles). Both the received packet count and the count of cycles are maintained the RTP header validity check routine in Appendix A.1.

```
    extended_max = s->cycles + s->max_seq;
    expected = extended_max - s->base_seq + 1;
```

The number of packets lost is defined to be the number of packets expected less the number of packets actually received:

```
    lost = expected - s->received;
```

Since this number is carried in 24 bits, it should be clamped at 0xffffff rather than wrap around to zero.

The fraction of packets lost during the last reporting interval (since the previous SR or RR packet was sent) is calculated from differences in the expected and received packet counts across the interval, where expected_prior and received_prior are the values saved when the previous reception report was generated:

```
    expected_interval = expected - s->expected_prior;
    s->expected_prior = expected;
    received_interval = s->received - s->received_prior;
    s->received_prior = s->received;
    lost_interval = expected_interval - received_interval;
    if (expected_interval == 0 || lost_interval <= 0) fraction = 0;
    else fraction = (lost_interval << 8) / expected_interval;
```

The resulting fraction is an 8-bit fixed point number with the binary point at the left edge.

A.4 Generating SDES RTCP Packets

This function builds one SDES chunk into buffer b composed of argc items supplied in arrays type , value and length b

```
char *rtp_write_sdes(char *b, u_int32 src, int argc,
                     rtcp_sdes_type_t type[], char *value[],
                     int length[])
{
    rtcp_sdes_t *s = (rtcp_sdes_t *)b;
    rtcp_sdes_item_t *rsp;
```


Microsoft Exhibit 1019

```
        int i;
        int len;
        int pad;

        /* SSRC header */
        s->src = src;
        rsp = &s->item[0];

        /* SDES items */
        for (i = 0; i < argc; i++) {
            rsp->type = type[i];
            len = length[i];
            if (len > RTP_MAX_SDES) {
                /* invalid length, may want to take other action */
                len = RTP_MAX_SDES;
            }
            rsp->length = len;
            memcpy(rsp->data, value[i], len);
            rsp = (rtcp_sdes_item_t *)&rsp->data[len];
        }

        /* terminate with end marker and pad to next 4-octet boundary */
        len = ((char *) rsp) - b;
        pad = 4 - (len & 0x3);
        b = (char *) rsp;
        while (pad--) *b++ = RTCP_SDES_END;

        return b;
    }
```

A.5 Parsing RTCP SDES Packets

This function parses an SDES packet, calling functions find_member() to find a pointer to the information for a session member given the SSRC identifier and member_sdes() to store the new SDES information for that member. This function expects a pointer to the header of the RTCP packet.

```
    void rtp_read_sdes(rtcp_t *r)
    {
        int count = r->common.count;
        rtcp_sdes_t *sd = &r->r.sdes;
        rtcp_sdes_item_t *rsp, *rspn;
        rtcp_sdes_item_t *end = (rtcp_sdes_item_t *)
                                ((u_int32 *)r + r->common.length + 1);
        source *s;

        while (--count >= 0) {
```


Microsoft Exhibit 1019

```
            rsp = &sd->item[0];
            if (rsp >= end) break;
            s = find_member(sd->src);

            for (; rsp->type; rsp = rspn ) {
                rspn = (rtcp_sdes_item_t *)((char*)rsp+rsp->length+2);
                if (rspn >= end) {
                    rsp = rspn;
                    break;
                }
                member_sdes(s, rsp->type, rsp->data, rsp->length);
            }
            sd = (rtcp_sdes_t *)
                 ((u_int32 *)sd + (((char *)rsp - (char *)sd) >> 2)+1);
        }
        if (count >= 0) {
            /* invalid packet format */
        }
    }
```

A.6 Generating a Random 32-bit Identifier

The following subroutine generates a random 32-bit identifier using the MD5 routines published in RFC 1321 [23]. The system routines may not be present on all operating systems, but they should serve as hints as to what kinds of information may be used. Other system calls that may be appropriate include

- o getdomainname() ,
- o getwd() , or
- o getrusage()

"Live" video or audio samples are also a good source of random numbers, but care must be taken to avoid using a turned-off microphone or blinded camera as a source [7].

Use of this or similar routine is suggested to generate the initial seed for the random number generator producing the RTCP period (as shown in Appendix A.7), to generate the initial values for the sequence number and timestamp, and to generate SSRC values. Since this routine is likely to be CPU-intensive, its direct use to generate RTCP periods is inappropriate because predictability is not an issue. Note that this routine produces the same result on repeated calls until the value of the system clock changes unless different values are supplied for the type argument.

Microsoft Exhibit 1019

```
/*
 * Generate a random 32-bit quantity.
 */
#include <sys/types.h>   /* u_long */
#include <sys/time.h>    /* gettimeofday() */
#include <unistd.h>      /* get..() */
#include <stdio.h>       /* printf() */
#include <time.h>        /* clock() */
#include <sys/utsname.h> /* uname() */
#include "global.h"      /* from RFC 1321 */
#include "md5.h"         /* from RFC 1321 */

#define MD_CTX MD5_CTX
#define MDInit MD5Init
#define MDUpdate MD5Update
#define MDFinal MD5Final

static u_long md_32(char *string, int length)
{
    MD_CTX context;
    union {
        char   c[16];
        u_long x[4];
    } digest;
    u_long r;
    int i;

    MDInit (&context);
    MDUpdate (&context, string, length);
    MDFinal ((unsigned char *)&digest, &context);
    r = 0;
    for (i = 0; i < 3; i++) {
        r ^= digest.x[i];
    }
    return r;
}                               /* md_32 */


/*
 * Return random unsigned 32-bit quantity. Use 'type' argument if you
 * need to generate several different values in close succession.
 */
u_int32 random32(int type)
{
    struct {
        int     type;
        struct  timeval tv;
        clock_t cpu;
```


Microsoft Exhibit 1019

```
            pid_t   pid;
            u_long  hid;
            uid_t   uid;
            gid_t   gid;
            struct  utsname name;
        } s;

        gettimeofday(&s.tv, 0);
        uname(&s.name);
        s.type = type;
        s.cpu  = clock();
        s.pid  = getpid();
        s.hid  = gethostid();
        s.uid  = getuid();
        s.gid  = getgid();

        return md_32((char *)&s, sizeof(s));
    }                               /* random32 */
```

A.7 Computing the RTCP Transmission Interval

The following function returns the time between transmissions of RTCP packets, measured in seconds. It should be called after sending one compound RTCP packet to calculate the delay until the next should be sent. This function should also be called to calculate the delay before sending the first RTCP packet upon startup rather than send the packet immediately. This avoids any burst of RTCP packets if an application is started at many sites simultaneously, for example as a result of a session announcement.

The parameters have the following meaning:

rtcp_bw: The target RTCP bandwidth, i.e., the total bandwidth that will be used for RTCP packets by all members of this session, in octets per second. This should be 5% of the "session bandwidth" parameter supplied to the application at startup.

senders: Number of active senders since sending last report, known from construction of receiver reports for this RTCP packet. Includes ourselves, if we also sent during this interval.

members: The estimated number of session members, including ourselves. Incremented as we discover new session members from the receipt of RTP or RTCP packets, and decremented as session members leave (via RTCP BYE) or their state is timed out (30 minutes is recommended). On the first call, this parameter should have the value 1.

Microsoft Exhibit 1019

we_sent: Flag that is true if we have sent data during the last two RTCP intervals. If the flag is true, the compound RTCP packet just sent contained an SR packet.

packet_size: The size of the compound RTCP packet just sent, in octets, including the network encapsulation (e.g., 28 octets for UDP over IP).

avg_rtcp_size: Pointer to estimator for compound RTCP packet size; initialized and updated by this function for the packet just sent, and also updated by an identical line of code in the RTCP receive routine for every RTCP packet received from other participants in the session.

initial: Flag that is true for the first call upon startup to calculate the time until the first report should be sent.

```
#include <math.h>

double rtcp_interval(int members,
                     int senders,
                     double rtcp_bw,
                     int we_sent,
                     int packet_size,
                     int *avg_rtcp_size,
                     int initial)
{
    /*
     * Minimum time between RTCP packets from this site (in seconds).
     * This time prevents the reports from 'clumping' when sessions
     * are small and the law of large numbers isn't helping to smooth
     * out the traffic.  It also keeps the report interval from
     * becoming ridiculously small during transient outages like a
     * network partition.
     */
    double const RTCP_MIN_TIME = 5.;
    /*
     * Fraction of the RTCP bandwidth to be shared among active
     * senders.  (This fraction was chosen so that in a typical
     * session with one or two active senders, the computed report
     * time would be roughly equal to the minimum report time so that
     * we don't unnecessarily slow down receiver reports.) The
     * receiver fraction must be 1 - the sender fraction.
     */
    double const RTCP_SENDER_BW_FRACTION = 0.25;
    double const RTCP_RCVR_BW_FRACTION = (1-RTCP_SENDER_BW_FRACTION);
    /*
     * Gain (smoothing constant) for the low-pass filter that
```


Microsoft Exhibit 1019

```
         * estimates the average RTCP packet size (see Cadzow reference).
         */
        double const RTCP_SIZE_GAIN = (1./16.);

        double t;                   /* interval */
        double rtcp_min_time = RTCP_MIN_TIME;
        int n;                      /* no. of members for computation */

        /*
         * Very first call at application start-up uses half the min
         * delay for quicker notification while still allowing some time
         * before reporting for randomization and to learn about other
         * sources so the report interval will converge to the correct
         * interval more quickly.  The average RTCP size is initialized
         * to 128 octets which is conservative (it assumes everyone else
         * is generating SRs instead of RRs: 20 IP + 8 UDP + 52 SR + 48
         * SDES CNAME).
         */
        if (initial) {
            rtcp_min_time /= 2;
            *avg_rtcp_size = 128;
        }

        /*
         * If there were active senders, give them at least a minimum
         * share of the RTCP bandwidth.  Otherwise all participants share
         * the RTCP bandwidth equally.
         */
        n = members;
        if (senders > 0 && senders < members * RTCP_SENDER_BW_FRACTION) {
            if (we_sent) {
                rtcp_bw *= RTCP_SENDER_BW_FRACTION;
                n = senders;
            } else {
                rtcp_bw *= RTCP_RCVR_BW_FRACTION;
                n -= senders;
            }
        }

        /*
         * Update the average size estimate by the size of the report
         * packet we just sent.
         */
        *avg_rtcp_size += (packet_size - *avg_rtcp_size)*RTCP_SIZE_GAIN;

        /*
         * The effective number of sites times the average packet size is
         * the total number of octets sent when each site sends a report.
```


Microsoft Exhibit 1019

```
         * Dividing this by the effective bandwidth gives the time
         * interval over which those packets must be sent in order to
         * meet the bandwidth target, with a minimum enforced.  In that
         * time interval we send one report so this time is also our
         * average time between reports.
         */
        t = (*avg_rtcp_size) * n / rtcp_bw;
        if (t < rtcp_min_time) t = rtcp_min_time;

        /*
         * To avoid traffic bursts from unintended synchronization with
         * other sites, we then pick our actual next report interval as a
         * random number uniformly distributed between 0.5*t and 1.5*t.
         */
        return t * (drand48() + 0.5);
    }
```

A.8 Estimating the Interarrival Jitter

The code fragments below implement the algorithm given in Section 6.3.1 for calculating an estimate of the statistical variance of the RTP data interarrival time to be inserted in the interarrival jitter field of reception reports. The inputs are r->ts , the timestamp from the incoming packet, and arrival , the current time in the same units. Here s points to state for the source; s->transit holds the relative transit time for the previous packet, and s->jitter holds the estimated jitter. The jitter field of the reception report is measured in timestamp units and expressed as an unsigned integer, but the jitter estimate is kept in a floating point. As each data packet arrives, the jitter estimate is updated:

```
    int transit = arrival - r->ts;
    int d = transit - s->transit;
    s->transit = transit;
    if (d < 0) d = -d;
    s->jitter += (1./16.) * ((double)d - s->jitter);
```

When a reception report block (to which rr points) is generated for this member, the current jitter estimate is returned:

```
    rr->jitter = (u_int32) s->jitter;
```

Alternatively, the jitter estimate can be kept as an integer, but scaled to reduce round-off error. The calculation is the same except for the last line:

```
    s->jitter += d - ((s->jitter + 8) >> 4);
```


Microsoft Exhibit 1019

In this case, the estimate is sampled for the reception report as:

```
rr->jitter = s->jitter >> 4;
```

B. Security Considerations

RTP suffers from the same security liabilities as the underlying protocols. For example, an impostor can fake source or destination network addresses, or change the header or payload. Within RTCP, the CNAME and NAME information may be used to impersonate another participant. In addition, RTP may be sent via IP multicast, which provides no direct means for a sender to know all the receivers of the data sent and therefore no measure of privacy. Rightly or not, users may be more sensitive to privacy concerns with audio and video communication than they have been with more traditional forms of network communication [24]. Therefore, the use of security mechanisms with RTP is important. These mechanisms are discussed in Section 9.

RTP-level translators or mixers may be used to allow RTP traffic to reach hosts behind firewalls. Appropriate firewall security principles and practices, which are beyond the scope of this document, should be followed in the design and installation of these devices and in the admission of RTP applications for use behind the firewall.

C. Authors' Addresses

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OpenPGP Message Format

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

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IESG Note

This document defines many tag values, yet it doesn't describe a mechanism for adding new tags (for new features). Traditionally the Internet Assigned Numbers Authority (IANA) handles the allocation of new values for future expansion and RFCs usually define the procedure to be used by the IANA. However, there are subtle (and not so subtle) interactions that may occur in this protocol between new features and existing features which result in a significant reduction in over all security. Therefore, this document does not define an extension procedure. Instead requests to define new tag values (say for new encryption algorithms for example) should be forwarded to the IESG Security Area Directors for consideration or forwarding to the appropriate IETF Working Group for consideration.

Abstract

This document is maintained in order to publish all necessary information needed to develop interoperable applications based on the OpenPGP format. It is not a step-by-step cookbook for writing an application. It describes only the format and methods needed to read, check, generate, and write conforming packets crossing any network. It does not deal with storage and implementation questions. It does,

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however, discuss implementation issues necessary to avoid security flaws.

Open-PGP software uses a combination of strong public-key and symmetric cryptography to provide security services for electronic communications and data storage. These services include confidentiality, key management, authentication, and digital signatures. This document specifies the message formats used in OpenPGP.

Table of Contents

Status of this Memo 1
IESG Note 1
Abstract 1
Table of Contents 2
1. Introduction 4
1.1. Terms 5
2. General functions 5
2.1. Confidentiality via Encryption 5
2.2. Authentication via Digital signature 6
2.3. Compression 7
2.4. Conversion to Radix-64 7
2.5. Signature-Only Applications 7
3. Data Element Formats 7
3.1. Scalar numbers 8
3.2. Multi-Precision Integers 8
3.3. Key IDs 8
3.4. Text 8
3.5. Time fields 9
3.6. String-to-key (S2K) specifiers 9
3.6.1. String-to-key (S2k) specifier types 9
3.6.1.1. Simple S2K 9
3.6.1.2. Salted S2K 10
3.6.1.3. Iterated and Salted S2K 10
3.6.2. String-to-key usage 11
3.6.2.1. Secret key encryption 11
3.6.2.2. Symmetric-key message encryption 11
4. Packet Syntax 12
4.1. Overview 12
4.2. Packet Headers 12
4.2.1. Old-Format Packet Lengths 13
4.2.2. New-Format Packet Lengths 13
4.2.2.1. One-Octet Lengths 14
4.2.2.2. Two-Octet Lengths 14
4.2.2.3. Five-Octet Lengths 14
4.2.2.4. Partial Body Lengths 14
4.2.3. Packet Length Examples 14

Microsoft Exhibit 1019

4.3. Packet Tags 15
5. Packet Types 16
5.1. Public-Key Encrypted Session Key Packets (Tag 1) 16
5.2. Signature Packet (Tag 2) 17
5.2.1. Signature Types 17
5.2.2. Version 3 Signature Packet Format 19
5.2.3. Version 4 Signature Packet Format 21
5.2.3.1. Signature Subpacket Specification 22
5.2.3.2. Signature Subpacket Types 24
5.2.3.3. Signature creation time 25
5.2.3.4. Issuer 25
5.2.3.5. Key expiration time 25
5.2.3.6. Preferred symmetric algorithms 25
5.2.3.7. Preferred hash algorithms 25
5.2.3.8. Preferred compression algorithms 26
5.2.3.9. Signature expiration time 26
5.2.3.10. Exportable Certification 26
5.2.3.11. Revocable 27
5.2.3.12. Trust signature 27
5.2.3.13. Regular expression 27
5.2.3.14. Revocation key 27
5.2.3.15. Notation Data 28
5.2.3.16. Key server preferences 28
5.2.3.17. Preferred key server 29
5.2.3.18. Primary user id 29
5.2.3.19. Policy URL 29
5.2.3.20. Key Flags 29
5.2.3.21. Signer's User ID 30
5.2.3.22. Reason for Revocation 30
5.2.4. Computing Signatures 31
5.2.4.1. Subpacket Hints 32
5.3. Symmetric-Key Encrypted Session-Key Packets (Tag 3) 32
5.4. One-Pass Signature Packets (Tag 4) 33
5.5. Key Material Packet 34
5.5.1. Key Packet Variants 34
5.5.1.1. Public Key Packet (Tag 6) 34
5.5.1.2. Public Subkey Packet (Tag 14) 34
5.5.1.3. Secret Key Packet (Tag 5) 35
5.5.1.4. Secret Subkey Packet (Tag 7) 35
5.5.2. Public Key Packet Formats 35
5.5.3. Secret Key Packet Formats 37
5.6. Compressed Data Packet (Tag 8) 38
5.7. Symmetrically Encrypted Data Packet (Tag 9) 39
5.8. Marker Packet (Obsolete Literal Packet) (Tag 10) 39
5.9. Literal Data Packet (Tag 11) 40
5.10. Trust Packet (Tag 12) 40
5.11. User ID Packet (Tag 13) 41
6. Radix-64 Conversions 41

Microsoft Exhibit 1019

6.1. An Implementation of the CRC-24 in "C" 42
6.2. Forming ASCII Armor 42
6.3. Encoding Binary in Radix-64 44
6.4. Decoding Radix-64 46
6.5. Examples of Radix-64 46
6.6. Example of an ASCII Armored Message 47
7. Cleartext signature framework 47
7.1. Dash-Escaped Text 47
8. Regular Expressions 48
9. Constants 49
9.1. Public Key Algorithms 49
9.2. Symmetric Key Algorithms 49
9.3. Compression Algorithms 50
9.4. Hash Algorithms 50
10. Packet Composition 50
10.1. Transferable Public Keys 50
10.2. OpenPGP Messages 52
10.3. Detached Signatures 52
11. Enhanced Key Formats 52
11.1. Key Structures 52
11.2. Key IDs and Fingerprints 53
12. Notes on Algorithms 54
12.1. Symmetric Algorithm Preferences 54
12.2. Other Algorithm Preferences 55
12.2.1. Compression Preferences 56
12.2.2. Hash Algorithm Preferences 56
12.3. Plaintext 56
12.4. RSA 56
12.5. Elgamal 57
12.6. DSA 58
12.7. Reserved Algorithm Numbers 58
12.8. OpenPGP CFB mode 58
13. Security Considerations 59
14. Implementation Nits 60
15. Authors and Working Group Chair 62
16. References 63
17. Full Copyright Statement 65

1. Introduction

This document provides information on the message-exchange packet formats used by OpenPGP to provide encryption, decryption, signing, and key management functions. It builds on the foundation provided in RFC 1991 "PGP Message Exchange Formats."

Microsoft Exhibit 1019

1.1. Terms

* OpenPGP - This is a definition for security software that uses PGP 5.x as a basis.

* PGP - Pretty Good Privacy. PGP is a family of software systems developed by Philip R. Zimmermann from which OpenPGP is based.

* PGP 2.6.x - This version of PGP has many variants, hence the term PGP 2.6.x. It used only RSA, MD5, and IDEA for its cryptographic transforms. An informational RFC, RFC 1991, was written describing this version of PGP.

* PGP 5.x - This version of PGP is formerly known as "PGP 3" in the community and also in the predecessor of this document, RFC 1991. It has new formats and corrects a number of problems in the PGP 2.6.x design. It is referred to here as PGP 5.x because that software was the first release of the "PGP 3" code base.

"PGP", "Pretty Good", and "Pretty Good Privacy" are trademarks of Network Associates, Inc. and are used with permission.

This document uses the terms "MUST", "SHOULD", and "MAY" as defined in RFC 2119, along with the negated forms of those terms.

2. General functions

OpenPGP provides data integrity services for messages and data files by using these core technologies:

- digital signatures

- encryption

- compression

- radix-64 conversion

In addition, OpenPGP provides key management and certificate services, but many of these are beyond the scope of this document.

2.1. Confidentiality via Encryption

OpenPGP uses two encryption methods to provide confidentiality: symmetric-key encryption and public key encryption. With public-key encryption, the object is encrypted using a symmetric encryption algorithm. Each symmetric key is used only once. A new "session key" is generated as a random number for each message. Since it is used

Microsoft Exhibit 1019

only once, the session key is bound to the message and transmitted with it. To protect the key, it is encrypted with the receiver's public key. The sequence is as follows:

1. The sender creates a message.

2. The sending OpenPGP generates a random number to be used as a session key for this message only.

3. The session key is encrypted using each recipient's public key. These "encrypted session keys" start the message.

4. The sending OpenPGP encrypts the message using the session key, which forms the remainder of the message. Note that the message is also usually compressed.

5. The receiving OpenPGP decrypts the session key using the recipient's private key.

6. The receiving OpenPGP decrypts the message using the session key. If the message was compressed, it will be decompressed.

With symmetric-key encryption, an object may be encrypted with a symmetric key derived from a passphrase (or other shared secret), or a two-stage mechanism similar to the public-key method described above in which a session key is itself encrypted with a symmetric algorithm keyed from a shared secret.

Both digital signature and confidentiality services may be applied to the same message. First, a signature is generated for the message and attached to the message. Then, the message plus signature is encrypted using a symmetric session key. Finally, the session key is encrypted using public-key encryption and prefixed to the encrypted block.

2.2. Authentication via Digital signature

The digital signature uses a hash code or message digest algorithm, and a public-key signature algorithm. The sequence is as follows:

1. The sender creates a message.

2. The sending software generates a hash code of the message.

3. The sending software generates a signature from the hash code using the sender's private key.

4. The binary signature is attached to the message.

Microsoft Exhibit 1019

5. The receiving software keeps a copy of the message signature.

6. The receiving software generates a new hash code for the received message and verifies it using the message's signature. If the verification is successful, the message is accepted as authentic.

2.3. Compression

OpenPGP implementations MAY compress the message after applying the signature but before encryption.

2.4. Conversion to Radix-64

OpenPGP's underlying native representation for encrypted messages, signature certificates, and keys is a stream of arbitrary octets. Some systems only permit the use of blocks consisting of seven-bit, printable text. For transporting OpenPGP's native raw binary octets through channels that are not safe to raw binary data, a printable encoding of these binary octets is needed. OpenPGP provides the service of converting the raw 8-bit binary octet stream to a stream of printable ASCII characters, called Radix-64 encoding or ASCII Armor.

Implementations SHOULD provide Radix-64 conversions.

Note that many applications, particularly messaging applications, will want more advanced features as described in the OpenPGP-MIME document, RFC 2015. An application that implements OpenPGP for messaging SHOULD implement OpenPGP-MIME.

2.5. Signature-Only Applications

OpenPGP is designed for applications that use both encryption and signatures, but there are a number of problems that are solved by a signature-only implementation. Although this specification requires both encryption and signatures, it is reasonable for there to be subset implementations that are non-comformant only in that they omit encryption.

3. Data Element Formats

This section describes the data elements used by OpenPGP.

Microsoft Exhibit 1019

3.1. Scalar numbers

Scalar numbers are unsigned, and are always stored in big-endian format. Using n[k] to refer to the kth octet being interpreted, the value of a two-octet scalar is ((n[0] << 8) + n[1]). The value of a four-octet scalar is ((n[0] << 24) + (n[1] << 16) + (n[2] << 8) + n[3]).

3.2. Multi-Precision Integers

Multi-Precision Integers (also called MPIs) are unsigned integers used to hold large integers such as the ones used in cryptographic calculations.

An MPI consists of two pieces: a two-octet scalar that is the length of the MPI in bits followed by a string of octets that contain the actual integer.

These octets form a big-endian number; a big-endian number can be made into an MPI by prefixing it with the appropriate length.

Examples:

(all numbers are in hexadecimal)

The string of octets [00 01 01] forms an MPI with the value 1. The string [00 09 01 FF] forms an MPI with the value of 511.

Additional rules:

The size of an MPI is ((MPI.length + 7) / 8) + 2 octets.

The length field of an MPI describes the length starting from its most significant non-zero bit. Thus, the MPI [00 02 01] is not formed correctly. It should be [00 01 01].

3.3. Key IDs

A Key ID is an eight-octet scalar that identifies a key. Implementations SHOULD NOT assume that Key IDs are unique. The section, "Enhanced Key Formats" below describes how Key IDs are formed.

3.4. Text

The default character set for text is the UTF-8 [RFC2279] encoding of Unicode [ISO10646].

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3.5. Time fields

A time field is an unsigned four-octet number containing the number of seconds elapsed since midnight, 1 January 1970 UTC.

3.6. String-to-key (S2K) specifiers

String-to-key (S2K) specifiers are used to convert passphrase strings into symmetric-key encryption/decryption keys. They are used in two places, currently: to encrypt the secret part of private keys in the private keyring, and to convert passphrases to encryption keys for symmetrically encrypted messages.

3.6.1. String-to-key (S2k) specifier types

There are three types of S2K specifiers currently supported, as follows:

3.6.1.1. Simple S2K

This directly hashes the string to produce the key data. See below for how this hashing is done.

```
Octet 0:        0x00
Octet 1:        hash algorithm
```

Simple S2K hashes the passphrase to produce the session key. The manner in which this is done depends on the size of the session key (which will depend on the cipher used) and the size of the hash algorithm's output. If the hash size is greater than or equal to the session key size, the high-order (leftmost) octets of the hash are used as the key.

If the hash size is less than the key size, multiple instances of the hash context are created -- enough to produce the required key data. These instances are preloaded with 0, 1, 2, ... octets of zeros (that is to say, the first instance has no preloading, the second gets preloaded with 1 octet of zero, the third is preloaded with two octets of zeros, and so forth).

As the data is hashed, it is given independently to each hash context. Since the contexts have been initialized differently, they will each produce different hash output. Once the passphrase is hashed, the output data from the multiple hashes is concatenated, first hash leftmost, to produce the key data, with any excess octets on the right discarded.

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3.6.1.2. Salted S2K

This includes a "salt" value in the S2K specifier -- some arbitrary data -- that gets hashed along with the passphrase string, to help prevent dictionary attacks.

```
Octet 0:        0x01
Octet 1:        hash algorithm
Octets 2-9:     8-octet salt value
```

Salted S2K is exactly like Simple S2K, except that the input to the hash function(s) consists of the 8 octets of salt from the S2K specifier, followed by the passphrase.

3.6.1.3. Iterated and Salted S2K

This includes both a salt and an octet count. The salt is combined with the passphrase and the resulting value is hashed repeatedly. This further increases the amount of work an attacker must do to try dictionary attacks.

```
Octet  0:        0x03
Octet  1:        hash algorithm
Octets 2-9:      8-octet salt value
Octet  10:       count, a one-octet, coded value
```

The count is coded into a one-octet number using the following formula:

```
#define EXPBIAS 6
    count = ((Int32)16 + (c & 15)) << ((c >> 4) + EXPBIAS);
```

The above formula is in C, where "Int32" is a type for a 32-bit integer, and the variable "c" is the coded count, Octet 10.

Iterated-Salted S2K hashes the passphrase and salt data multiple times. The total number of octets to be hashed is specified in the encoded count in the S2K specifier. Note that the resulting count value is an octet count of how many octets will be hashed, not an iteration count.

Initially, one or more hash contexts are set up as with the other S2K algorithms, depending on how many octets of key data are needed. Then the salt, followed by the passphrase data is repeatedly hashed until the number of octets specified by the octet count has been hashed. The one exception is that if the octet count is less than the size of the salt plus passphrase, the full salt plus passphrase will be hashed even though that is greater than the octet count.

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After the hashing is done the data is unloaded from the hash context(s) as with the other S2K algorithms.

3.6.2. String-to-key usage

Implementations SHOULD use salted or iterated-and-salted S2K specifiers, as simple S2K specifiers are more vulnerable to dictionary attacks.

3.6.2.1. Secret key encryption

An S2K specifier can be stored in the secret keyring to specify how to convert the passphrase to a key that unlocks the secret data. Older versions of PGP just stored a cipher algorithm octet preceding the secret data or a zero to indicate that the secret data was unencrypted. The MD5 hash function was always used to convert the passphrase to a key for the specified cipher algorithm.

For compatibility, when an S2K specifier is used, the special value 255 is stored in the position where the hash algorithm octet would have been in the old data structure. This is then followed immediately by a one-octet algorithm identifier, and then by the S2K specifier as encoded above.

Therefore, preceding the secret data there will be one of these possibilities:

0:	secret data is unencrypted (no pass phrase)
255:	followed by algorithm octet and S2K specifier
Cipher alg:	use Simple S2K algorithm using MD5 hash

This last possibility, the cipher algorithm number with an implicit use of MD5 and IDEA, is provided for backward compatibility; it MAY be understood, but SHOULD NOT be generated, and is deprecated.

These are followed by an 8-octet Initial Vector for the decryption of the secret values, if they are encrypted, and then the secret key values themselves.

3.6.2.2. Symmetric-key message encryption

OpenPGP can create a Symmetric-key Encrypted Session Key (ESK) packet at the front of a message. This is used to allow S2K specifiers to be used for the passphrase conversion or to create messages with a mix of symmetric-key ESKs and public-key ESKs. This allows a message to be decrypted either with a passphrase or a public key.

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PGP 2.X always used IDEA with Simple string-to-key conversion when encrypting a message with a symmetric algorithm. This is deprecated, but MAY be used for backward-compatibility.

4. Packet Syntax

This section describes the packets used by OpenPGP.

4.1. Overview

An OpenPGP message is constructed from a number of records that are traditionally called packets. A packet is a chunk of data that has a tag specifying its meaning. An OpenPGP message, keyring, certificate, and so forth consists of a number of packets. Some of those packets may contain other OpenPGP packets (for example, a compressed data packet, when uncompressed, contains OpenPGP packets).

Each packet consists of a packet header, followed by the packet body. The packet header is of variable length.

4.2. Packet Headers

The first octet of the packet header is called the "Packet Tag." It determines the format of the header and denotes the packet contents. The remainder of the packet header is the length of the packet.

Note that the most significant bit is the left-most bit, called bit 7. A mask for this bit is 0x80 in hexadecimal.

```
     +---------------+
PTag |7 6 5 4 3 2 1 0|
     +---------------+
Bit 7 -- Always one
Bit 6 -- New packet format if set
```

PGP 2.6.x only uses old format packets. Thus, software that interoperates with those versions of PGP must only use old format packets. If interoperability is not an issue, either format may be used. Note that old format packets have four bits of content tags, and new format packets have six; some features cannot be used and still be backward-compatible.

Old format packets contain:

```
Bits 5-2 -- content tag
Bits 1-0 - length-type
```


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New format packets contain:

Bits 5-0 -- content tag

4.2.1. Old-Format Packet Lengths

The meaning of the length-type in old-format packets is:

0 - The packet has a one-octet length. The header is 2 octets long.

1 - The packet has a two-octet length. The header is 3 octets long.

2 - The packet has a four-octet length. The header is 5 octets long.

3 - The packet is of indeterminate length. The header is 1 octet long, and the implementation must determine how long the packet is. If the packet is in a file, this means that the packet extends until the end of the file. In general, an implementation SHOULD NOT use indeterminate length packets except where the end of the data will be clear from the context, and even then it is better to use a definite length, or a new-format header. The new-format headers described below have a mechanism for precisely encoding data of indeterminate length.

4.2.2. New-Format Packet Lengths

New format packets have four possible ways of encoding length:

1. A one-octet Body Length header encodes packet lengths of up to 191 octets.

2. A two-octet Body Length header encodes packet lengths of 192 to 8383 octets.

3. A five-octet Body Length header encodes packet lengths of up to 4,294,967,295 (0xFFFFFFFF) octets in length. (This actually encodes a four-octet scalar number.)

4. When the length of the packet body is not known in advance by the issuer, Partial Body Length headers encode a packet of indeterminate length, effectively making it a stream.

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4.2.2.1. One-Octet Lengths

A one-octet Body Length header encodes a length of from 0 to 191 octets. This type of length header is recognized because the one octet value is less than 192. The body length is equal to:

```
bodyLen = 1st_octet;
```

4.2.2.2. Two-Octet Lengths

A two-octet Body Length header encodes a length of from 192 to 8383 octets. It is recognized because its first octet is in the range 192 to 223. The body length is equal to:

```
bodyLen = ((1st_octet - 192) << 8) + (2nd_octet) + 192
```

4.2.2.3. Five-Octet Lengths

A five-octet Body Length header consists of a single octet holding the value 255, followed by a four-octet scalar. The body length is equal to:

```
bodyLen = (2nd_octet << 24) | (3rd_octet << 16) |
          (4th_octet << 8)  | 5th_octet
```

4.2.2.4. Partial Body Lengths

A Partial Body Length header is one octet long and encodes the length of only part of the data packet. This length is a power of 2, from 1 to 1,073,741,824 (2 to the 30th power). It is recognized by its one octet value that is greater than or equal to 224, and less than 255. The partial body length is equal to:

```
partialBodyLen = 1 << (1st_octet & 0x1f);
```

Each Partial Body Length header is followed by a portion of the packet body data. The Partial Body Length header specifies this portion’s length. Another length header (of one of the three types -- one octet, two-octet, or partial) follows that portion. The last length header in the packet MUST NOT be a partial Body Length header. Partial Body Length headers may only be used for the non-final parts of the packet.

4.2.3. Packet Length Examples

These examples show ways that new-format packets might encode the packet lengths.

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A packet with length 100 may have its length encoded in one octet: 0x64. This is followed by 100 octets of data.

A packet with length 1723 may have its length coded in two octets: 0xC5, 0xFB. This header is followed by the 1723 octets of data.

A packet with length 100000 may have its length encoded in five octets: 0xFF, 0x00, 0x01, 0x86, 0xA0.

It might also be encoded in the following octet stream: 0xEF, first 32768 octets of data; 0xE1, next two octets of data; 0xE0, next one octet of data; 0xF0, next 65536 octets of data; 0xC5, 0xDD, last 1693 octets of data. This is just one possible encoding, and many variations are possible on the size of the Partial Body Length headers, as long as a regular Body Length header encodes the last portion of the data. Note also that the last Body Length header can be a zero-length header.

An implementation MAY use Partial Body Lengths for data packets, be they literal, compressed, or encrypted. The first partial length MUST be at least 512 octets long. Partial Body Lengths MUST NOT be used for any other packet types.

Please note that in all of these explanations, the total length of the packet is the length of the header(s) plus the length of the body.

4.3. Packet Tags

The packet tag denotes what type of packet the body holds. Note that old format headers can only have tags less than 16, whereas new format headers can have tags as great as 63. The defined tags (in decimal) are:

```
0        -- Reserved - a packet tag must not have this value
1        -- Public-Key Encrypted Session Key Packet
2        -- Signature Packet
3        -- Symmetric-Key Encrypted Session Key Packet
4        -- One-Pass Signature Packet
5        -- Secret Key Packet
6        -- Public Key Packet
7        -- Secret Subkey Packet
8        -- Compressed Data Packet
9        -- Symmetrically Encrypted Data Packet
10       -- Marker Packet
11       -- Literal Data Packet
12       -- Trust Packet
```


Microsoft Exhibit 1019

```
13       -- User ID Packet
14       -- Public Subkey Packet
60 to 63 -- Private or Experimental Values
```

5. Packet Types

5.1. Public-Key Encrypted Session Key Packets (Tag 1)

A Public-Key Encrypted Session Key packet holds the session key used to encrypt a message. Zero or more Encrypted Session Key packets (either Public-Key or Symmetric-Key) may precede a Symmetrically Encrypted Data Packet, which holds an encrypted message. The message is encrypted with the session key, and the session key is itself encrypted and stored in the Encrypted Session Key packet(s). The Symmetrically Encrypted Data Packet is preceded by one Public-Key Encrypted Session Key packet for each OpenPGP key to which the message is encrypted. The recipient of the message finds a session key that is encrypted to their public key, decrypts the session key, and then uses the session key to decrypt the message.

The body of this packet consists of:

- A one-octet number giving the version number of the packet type. The currently defined value for packet version is 3. An implementation should accept, but not generate a version of 2, which is equivalent to V3 in all other respects.

- An eight-octet number that gives the key ID of the public key that the session key is encrypted to.

- A one-octet number giving the public key algorithm used.

- A string of octets that is the encrypted session key. This string takes up the remainder of the packet, and its contents are dependent on the public key algorithm used.

Algorithm Specific Fields for RSA encryption

- multiprecision integer (MPI) of RSA encrypted value m**e mod n.

Algorithm Specific Fields for Elgamal encryption:

- MPI of Elgamal (Diffie-Hellman) value g**k mod p.

- MPI of Elgamal (Diffie-Hellman) value m * y**k mod p.

Microsoft Exhibit 1019

The value "m" in the above formulas is derived from the session key as follows. First the session key is prefixed with a one-octet algorithm identifier that specifies the symmetric encryption algorithm used to encrypt the following Symmetrically Encrypted Data Packet. Then a two-octet checksum is appended which is equal to the sum of the preceding session key octets, not including the algorithm identifier, modulo 65536. This value is then padded as described in PKCS-1 block type 02 [RFC2313] to form the "m" value used in the formulas above.

Note that when an implementation forms several PKESKs with one session key, forming a message that can be decrypted by several keys, the implementation MUST make new PKCS-1 padding for each key.

An implementation MAY accept or use a Key ID of zero as a "wild card" or "speculative" Key ID. In this case, the receiving implementation would try all available private keys, checking for a valid decrypted session key. This format helps reduce traffic analysis of messages.

5.2. Signature Packet (Tag 2)

A signature packet describes a binding between some public key and some data. The most common signatures are a signature of a file or a block of text, and a signature that is a certification of a user ID.

Two versions of signature packets are defined. Version 3 provides basic signature information, while version 4 provides an expandable format with subpackets that can specify more information about the signature. PGP 2.6.x only accepts version 3 signatures.

Implementations MUST accept V3 signatures. Implementations SHOULD generate V4 signatures. Implementations MAY generate a V3 signature that can be verified by PGP 2.6.x.

Note that if an implementation is creating an encrypted and signed message that is encrypted to a V3 key, it is reasonable to create a V3 signature.

5.2.1. Signature Types

There are a number of possible meanings for a signature, which are specified in a signature type octet in any given signature. These meanings are:

0x00: Signature of a binary document.
The signature is calculated over the data bytes, without any conversion. Typically, this means the signer owns it, created it, or certifies that it has not been modified.

Microsoft Exhibit 1019

0x01: Signature of a canonical text document.
Typically, this means the signer owns it, created it, or certifies that it has not been modified. The signature is calculated over the text data with its line endings converted to <CR><LF> and trailing blanks removed.

0x02: Standalone signature.
This signature is a signature of only its own subpacket contents. It is calculated identically to a signature over a zero-length binary document. Note that it doesn't make sense to have a V3 standalone signature.

0x10: Generic certification of a User ID and Public Key packet.
The issuer of this certification does not make any particular assertion as to how well the certifier has checked that the owner of the key is in fact the person described by the user ID. Note that all PGP "key signatures" are this type of certification.

0x11: Persona certification of a User ID and Public Key packet.
The issuer of this certification has not done any verification of the claim that the owner of this key is the user ID specified.

0x12: Casual certification of a User ID and Public Key packet.
The issuer of this certification has done some casual verification of the claim of identity.

0x13: Positive certification of a User ID and Public Key packet.
The issuer of this certification has done substantial verification of the claim of identity.

Please note that the vagueness of these certification claims is not a flaw, but a feature of the system. Because PGP places final authority for validity upon the receiver of a certification, it may be that one authority's casual certification might be more rigorous than some other authority's positive certification. These classifications allow a certification authority to issue fine-grained claims.

0x18: Subkey Binding Signature
This signature is a statement by the top-level signing key indicates that it owns the subkey. This signature is calculated directly on the subkey itself, not on any User ID or other packets.

Microsoft Exhibit 1019

0x1F: Signature directly on a key
This signature is calculated directly on a key. It binds the information in the signature subpackets to the key, and is appropriate to be used for subpackets that provide information about the key, such as the revocation key subpacket. It is also appropriate for statements that non-self certifiers want to make about the key itself, rather than the binding between a key and a name.

0x20: Key revocation signature
The signature is calculated directly on the key being revoked. A revoked key is not to be used. Only revocation signatures by the key being revoked, or by an authorized revocation key, should be considered valid revocation signatures.

0x28: Subkey revocation signature
The signature is calculated directly on the subkey being revoked. A revoked subkey is not to be used. Only revocation signatures by the top-level signature key that is bound to this subkey, or by an authorized revocation key, should be considered valid revocation signatures.

0x30: Certification revocation signature
This signature revokes an earlier user ID certification signature (signature class 0x10 through 0x13). It should be issued by the same key that issued the revoked signature or an authorized revocation key The signature should have a later creation date than the signature it revokes.

0x40: Timestamp signature.
This signature is only meaningful for the timestamp contained in it.

5.2.2. Version 3 Signature Packet Format

The body of a version 3 Signature Packet contains:

- One-octet version number (3).
- One-octet length of following hashed material. MUST be 5.
    - One-octet signature type.
    - Four-octet creation time.
- Eight-octet key ID of signer.
- One-octet public key algorithm.

Microsoft Exhibit 1019

- One-octet hash algorithm.

- Two-octet field holding left 16 bits of signed hash value.

- One or more multi-precision integers comprising the signature. This portion is algorithm specific, as described below.

The data being signed is hashed, and then the signature type and creation time from the signature packet are hashed (5 additional octets). The resulting hash value is used in the signature algorithm. The high 16 bits (first two octets) of the hash are included in the signature packet to provide a quick test to reject some invalid signatures.

Algorithm Specific Fields for RSA signatures:

- multiprecision integer (MPI) of RSA signature value m**d.

Algorithm Specific Fields for DSA signatures:

- MPI of DSA value r.

- MPI of DSA value s.

The signature calculation is based on a hash of the signed data, as described above. The details of the calculation are different for DSA signature than for RSA signatures.

With RSA signatures, the hash value is encoded as described in PKCS-1 section 10.1.2, "Data encoding", producing an ASN.1 value of type DigestInfo, and then padded using PKCS-1 block type 01 [RFC2313]. This requires inserting the hash value as an octet string into an ASN.1 structure. The object identifier for the type of hash being used is included in the structure. The hexadecimal representations for the currently defined hash algorithms are:

- MD2: 0x2A, 0x86, 0x48, 0x86, 0xF7, 0x0D, 0x02, 0x02

- MD5: 0x2A, 0x86, 0x48, 0x86, 0xF7, 0x0D, 0x02, 0x05

- RIPEMD-160: 0x2B, 0x24, 0x03, 0x02, 0x01

- SHA-1: 0x2B, 0x0E, 0x03, 0x02, 0x1A

Microsoft Exhibit 1019

The ASN.1 OIDs are:

- MD2: 1.2.840.113549.2.2

- MD5: 1.2.840.113549.2.5

- RIPEMD-160: 1.3.36.3.2.1

- SHA-1: 1.3.14.3.2.26

The full hash prefixes for these are:

MD2: 0x30, 0x20, 0x30, 0x0C, 0x06, 0x08, 0x2A, 0x86,
0x48, 0x86, 0xF7, 0x0D, 0x02, 0x02, 0x05, 0x00,
0x04, 0x10

MD5: 0x30, 0x20, 0x30, 0x0C, 0x06, 0x08, 0x2A, 0x86,
0x48, 0x86, 0xF7, 0x0D, 0x02, 0x05, 0x05, 0x00,
0x04, 0x10

RIPEMD-160: 0x30, 0x21, 0x30, 0x09, 0x06, 0x05, 0x2B, 0x24,
0x03, 0x02, 0x01, 0x05, 0x00, 0x04, 0x14

SHA-1: 0x30, 0x21, 0x30, 0x09, 0x06, 0x05, 0x2b, 0x0E,
0x03, 0x02, 0x1A, 0x05, 0x00, 0x04, 0x14

DSA signatures MUST use hashes with a size of 160 bits, to match q, the size of the group generated by the DSA key's generator value. The hash function result is treated as a 160 bit number and used directly in the DSA signature algorithm.

5.2.3. Version 4 Signature Packet Format

The body of a version 4 Signature Packet contains:

- One-octet version number (4).

- One-octet signature type.

- One-octet public key algorithm.

- One-octet hash algorithm.

- Two-octet scalar octet count for following hashed subpacket data. Note that this is the length in octets of all of the hashed subpackets; a pointer incremented by this number will skip over the hashed subpackets.

Microsoft Exhibit 1019

- Hashed subpacket data. (zero or more subpackets)

- Two-octet scalar octet count for following unhashed subpacket data. Note that this is the length in octets of all of the unhashed subpackets; a pointer incremented by this number will skip over the unhashed subpackets.

- Unhashed subpacket data. (zero or more subpackets)

- Two-octet field holding left 16 bits of signed hash value.

- One or more multi-precision integers comprising the signature. This portion is algorithm specific, as described above.

The data being signed is hashed, and then the signature data from the version number through the hashed subpacket data (inclusive) is hashed. The resulting hash value is what is signed. The left 16 bits of the hash are included in the signature packet to provide a quick test to reject some invalid signatures.

There are two fields consisting of signature subpackets. The first field is hashed with the rest of the signature data, while the second is unhashed. The second set of subpackets is not cryptographically protected by the signature and should include only advisory information.

The algorithms for converting the hash function result to a signature are described in a section below.

5.2.3.1. Signature Subpacket Specification

The subpacket fields consist of zero or more signature subpackets. Each set of subpackets is preceded by a two-octet scalar count of the length of the set of subpackets.

Each subpacket consists of a subpacket header and a body. The header consists of:

- the subpacket length (1, 2, or 5 octets)

- the subpacket type (1 octet)

and is followed by the subpacket specific data.

The length includes the type octet but not this length. Its format is similar to the "new" format packet header lengths, but cannot have partial body lengths. That is:

Microsoft Exhibit 1019

```
    if the 1st octet <  192, then
        lengthOfLength = 1
        subpacketLen = 1st_octet

    if the 1st octet >= 192 and < 255, then
        lengthOfLength = 2
        subpacketLen = ((1st_octet - 192) << 8) + (2nd_octet) + 192

    if the 1st octet = 255, then
        lengthOfLength = 5
        subpacket length = [four-octet scalar starting at 2nd_octet]
```

The value of the subpacket type octet may be:

```
    2 = signature creation time
    3 = signature expiration time
    4 = exportable certification
    5 = trust signature
    6 = regular expression
    7 = revocable
    9 = key expiration time
    10 = placeholder for backward compatibility
    11 = preferred symmetric algorithms
    12 = revocation key
    16 = issuer key ID
    20 = notation data
    21 = preferred hash algorithms
    22 = preferred compression algorithms
    23 = key server preferences
    24 = preferred key server
    25 = primary user id
    26 = policy URL
    27 = key flags
    28 = signer's user id
    29 = reason for revocation
    100 to 110 = internal or user-defined
```

An implementation SHOULD ignore any subpacket of a type that it does not recognize.

Bit 7 of the subpacket type is the "critical" bit. If set, it denotes that the subpacket is one that is critical for the evaluator of the signature to recognize. If a subpacket is encountered that is marked critical but is unknown to the evaluating software, the evaluator SHOULD consider the signature to be in error.

Microsoft Exhibit 1019

An evaluator may "recognize" a subpacket, but not implement it. The purpose of the critical bit is to allow the signer to tell an evaluator that it would prefer a new, unknown feature to generate an error than be ignored.

Implementations SHOULD implement "preferences".

5.2.3.2. Signature Subpacket Types

A number of subpackets are currently defined. Some subpackets apply to the signature itself and some are attributes of the key. Subpackets that are found on a self-signature are placed on a user id certification made by the key itself. Note that a key may have more than one user id, and thus may have more than one self-signature, and differing subpackets.

A self-signature is a binding signature made by the key the signature refers to. There are three types of self-signatures, the certification signatures (types 0x10-0x13), the direct-key signature (type 0x1f), and the subkey binding signature (type 0x18). For certification self-signatures, each user ID may have a self-signature, and thus different subpackets in those self-signatures. For subkey binding signatures, each subkey in fact has a self-signature. Subpackets that appear in a certification self-signature apply to the username, and subpackets that appear in the subkey self-signature apply to the subkey. Lastly, subpackets on the direct key signature apply to the entire key.

Implementing software should interpret a self-signature’s preference subpackets as narrowly as possible. For example, suppose a key has two usernames, Alice and Bob. Suppose that Alice prefers the symmetric algorithm CAST5, and Bob prefers IDEA or Triple-DES. If the software locates this key via Alice’s name, then the preferred algorithm is CAST5, if software locates the key via Bob’s name, then the preferred algorithm is IDEA. If the key is located by key id, then algorithm of the default user id of the key provides the default symmetric algorithm.

A subpacket may be found either in the hashed or unhashed subpacket sections of a signature. If a subpacket is not hashed, then the information in it cannot be considered definitive because it is not part of the signature proper.

Microsoft Exhibit 1019

5.2.3.3. Signature creation time

(4 octet time field)

The time the signature was made.

MUST be present in the hashed area.

5.2.3.4. Issuer

(8 octet key ID)

The OpenPGP key ID of the key issuing the signature.

5.2.3.5. Key expiration time

(4 octet time field)

The validity period of the key. This is the number of seconds after the key creation time that the key expires. If this is not present or has a value of zero, the key never expires. This is found only on a self-signature.

5.2.3.6. Preferred symmetric algorithms

(sequence of one-octet values)

Symmetric algorithm numbers that indicate which algorithms the key holder prefers to use. The subpacket body is an ordered list of octets with the most preferred listed first. It is assumed that only algorithms listed are supported by the recipient's software. Algorithm numbers in section 9. This is only found on a self-signature.

5.2.3.7. Preferred hash algorithms

(array of one-octet values)

Message digest algorithm numbers that indicate which algorithms the key holder prefers to receive. Like the preferred symmetric algorithms, the list is ordered. Algorithm numbers are in section 6. This is only found on a self-signature.

Microsoft Exhibit 1019

5.2.3.8. Preferred compression algorithms

(array of one-octet values)

Compression algorithm numbers that indicate which algorithms the key holder prefers to use. Like the preferred symmetric algorithms, the list is ordered. Algorithm numbers are in section 6. If this subpacket is not included, ZIP is preferred. A zero denotes that uncompressed data is preferred; the key holder's software might have no compression software in that implementation. This is only found on a self-signature.

5.2.3.9. Signature expiration time

(4 octet time field)

The validity period of the signature. This is the number of seconds after the signature creation time that the signature expires. If this is not present or has a value of zero, it never expires.

5.2.3.10. Exportable Certification

(1 octet of exportability, 0 for not, 1 for exportable)

This subpacket denotes whether a certification signature is "exportable", to be used by other users than the signature's issuer. The packet body contains a boolean flag indicating whether the signature is exportable. If this packet is not present, the certification is exportable; it is equivalent to a flag containing a 1.

Non-exportable, or "local", certifications are signatures made by a user to mark a key as valid within that user's implementation only. Thus, when an implementation prepares a user's copy of a key for transport to another user (this is the process of "exporting" the key), any local certification signatures are deleted from the key.

The receiver of a transported key "imports" it, and likewise trims any local certifications. In normal operation, there won't be any, assuming the import is performed on an exported key. However, there are instances where this can reasonably happen. For example, if an implementation allows keys to be imported from a key database in addition to an exported key, then this situation can arise.

Some implementations do not represent the interest of a single user (for example, a key server). Such implementations always trim local certifications from any key they handle.

Microsoft Exhibit 1019

5.2.3.11. Revocable

(1 octet of revocability, 0 for not, 1 for revocable)

Signature's revocability status. Packet body contains a boolean flag indicating whether the signature is revocable. Signatures that are not revocable have any later revocation signatures ignored. They represent a commitment by the signer that he cannot revoke his signature for the life of his key. If this packet is not present, the signature is revocable.

5.2.3.12. Trust signature

(1 octet "level" (depth), 1 octet of trust amount)

Signer asserts that the key is not only valid, but also trustworthy, at the specified level. Level 0 has the same meaning as an ordinary validity signature. Level 1 means that the signed key is asserted to be a valid trusted introducer, with the 2nd octet of the body specifying the degree of trust. Level 2 means that the signed key is asserted to be trusted to issue level 1 trust signatures, i.e. that it is a "meta introducer". Generally, a level n trust signature asserts that a key is trusted to issue level n-1 trust signatures. The trust amount is in a range from 0-255, interpreted such that values less than 120 indicate partial trust and values of 120 or greater indicate complete trust. Implementations SHOULD emit values of 60 for partial trust and 120 for complete trust.

5.2.3.13. Regular expression

(null-terminated regular expression)

Used in conjunction with trust signature packets (of level > 0) to limit the scope of trust that is extended. Only signatures by the target key on user IDs that match the regular expression in the body of this packet have trust extended by the trust signature subpacket. The regular expression uses the same syntax as the Henry Spencer's "almost public domain" regular expression package. A description of the syntax is found in a section below.

5.2.3.14. Revocation key

(1 octet of class, 1 octet of algid, 20 octets of fingerprint)

Authorizes the specified key to issue revocation signatures for this key. Class octet must have bit 0x80 set. If the bit 0x40 is set, then this means that the revocation information is sensitive. Other bits are for future expansion to other kinds of authorizations. This

Microsoft Exhibit 1019

is found on a self-signature.

If the "sensitive" flag is set, the keyholder feels this subpacket contains private trust information that describes a real-world sensitive relationship. If this flag is set, implementations SHOULD NOT export this signature to other users except in cases where the data needs to be available: when the signature is being sent to the designated revoker, or when it is accompanied by a revocation signature from that revoker. Note that it may be appropriate to isolate this subpacket within a separate signature so that it is not combined with other subpackets that need to be exported.

5.2.3.15. Notation Data

(4 octets of flags, 2 octets of name length (M),
2 octets of value length (N),
M octets of name data,
N octets of value data)

This subpacket describes a "notation" on the signature that the issuer wishes to make. The notation has a name and a value, each of which are strings of octets. There may be more than one notation in a signature. Notations can be used for any extension the issuer of the signature cares to make. The "flags" field holds four octets of flags.

All undefined flags MUST be zero. Defined flags are:

First octet: 0x80 = human-readable. This note is text, a note from one person to another, and has no meaning to software.
Other octets: none.

5.2.3.16. Key server preferences

(N octets of flags)

This is a list of flags that indicate preferences that the key holder has about how the key is handled on a key server. All undefined flags MUST be zero.

First octet: 0x80 = No-modify
the key holder requests that this key only be modified or updated by the key holder or an administrator of the key server.

This is found only on a self-signature.

Microsoft Exhibit 1019

5.2.3.17. Preferred key server

(String)

This is a URL of a key server that the key holder prefers be used for updates. Note that keys with multiple user ids can have a preferred key server for each user id. Note also that since this is a URL, the key server can actually be a copy of the key retrieved by ftp, http, finger, etc.

5.2.3.18. Primary user id

(1 octet, boolean)

This is a flag in a user id's self signature that states whether this user id is the main user id for this key. It is reasonable for an implementation to resolve ambiguities in preferences, etc. by referring to the primary user id. If this flag is absent, its value is zero. If more than one user id in a key is marked as primary, the implementation may resolve the ambiguity in any way it sees fit.

5.2.3.19. Policy URL

(String)

This subpacket contains a URL of a document that describes the policy that the signature was issued under.

5.2.3.20. Key Flags

(Octet string)

This subpacket contains a list of binary flags that hold information about a key. It is a string of octets, and an implementation MUST NOT assume a fixed size. This is so it can grow over time. If a list is shorter than an implementation expects, the unstated flags are considered to be zero. The defined flags are:

First octet:

0x01 - This key may be used to certify other keys.

0x02 - This key may be used to sign data.

0x04 - This key may be used to encrypt communications.

0x08 - This key may be used to encrypt storage.

Microsoft Exhibit 1019

0x10 - The private component of this key may have been split by a secret-sharing mechanism.

0x80 - The private component of this key may be in the possession of more than one person.

Usage notes:

The flags in this packet may appear in self-signatures or in certification signatures. They mean different things depending on who is making the statement -- for example, a certification signature that has the "sign data" flag is stating that the certification is for that use. On the other hand, the "communications encryption" flag in a self-signature is stating a preference that a given key be used for communications. Note however, that it is a thorny issue to determine what is "communications" and what is "storage." This decision is left wholly up to the implementation; the authors of this document do not claim any special wisdom on the issue, and realize that accepted opinion may change.

The "split key" (0x10) and "group key" (0x80) flags are placed on a self-signature only; they are meaningless on a certification signature. They SHOULD be placed only on a direct-key signature (type 0x1f) or a subkey signature (type 0x18), one that refers to the key the flag applies to.

5.2.3.21. Signer's User ID

This subpacket allows a keyholder to state which user id is responsible for the signing. Many keyholders use a single key for different purposes, such as business communications as well as personal communications. This subpacket allows such a keyholder to state which of their roles is making a signature.

5.2.3.22. Reason for Revocation

(1 octet of revocation code, N octets of reason string)

This subpacket is used only in key revocation and certification revocation signatures. It describes the reason why the key or certificate was revoked.

The first octet contains a machine-readable code that denotes the reason for the revocation:

Microsoft Exhibit 1019

0x00 - No reason specified (key revocations or cert revocations)
0x01 - Key is superceded (key revocations)
0x02 - Key material has been compromised (key revocations)
0x03 - Key is no longer used (key revocations)
0x20 - User id information is no longer valid (cert revocations)

Following the revocation code is a string of octets which gives information about the reason for revocation in human-readable form (UTF-8). The string may be null, that is, of zero length. The length of the subpacket is the length of the reason string plus one.

5.2.4. Computing Signatures

All signatures are formed by producing a hash over the signature data, and then using the resulting hash in the signature algorithm.

The signature data is simple to compute for document signatures (types 0x00 and 0x01), for which the document itself is the data. For standalone signatures, this is a null string.

When a signature is made over a key, the hash data starts with the octet 0x99, followed by a two-octet length of the key, and then body of the key packet. (Note that this is an old-style packet header for a key packet with two-octet length.) A subkey signature (type 0x18) then hashes the subkey, using the same format as the main key. Key revocation signatures (types 0x20 and 0x28) hash only the key being revoked.

A certification signature (type 0x10 through 0x13) hashes the user id being bound to the key into the hash context after the above data. A V3 certification hashes the contents of the name packet, without any header. A V4 certification hashes the constant 0xb4 (which is an old-style packet header with the length-of-length set to zero), a four-octet number giving the length of the username, and then the username data.

Once the data body is hashed, then a trailer is hashed. A V3 signature hashes five octets of the packet body, starting from the signature type field. This data is the signature type, followed by the four-octet signature time. A V4 signature hashes the packet body starting from its first field, the version number, through the end of the hashed subpacket data. Thus, the fields hashed are the signature version, the signature type, the public key algorithm, the hash algorithm, the hashed subpacket length, and the hashed subpacket body.

Microsoft Exhibit 1019

V4 signatures also hash in a final trailer of six octets: the version of the signature packet, i.e. 0x04; 0xFF; a four-octet, big-endian number that is the length of the hashed data from the signature packet (note that this number does not include these final six octets.

After all this has been hashed, the resulting hash field is used in the signature algorithm, and placed at the end of the signature packet.

5.2.4.1. Subpacket Hints

An implementation SHOULD put the two mandatory subpackets, creation time and issuer, as the first subpackets in the subpacket list, simply to make it easier for the implementer to find them.

It is certainly possible for a signature to contain conflicting information in subpackets. For example, a signature may contain multiple copies of a preference or multiple expiration times. In most cases, an implementation SHOULD use the last subpacket in the signature, but MAY use any conflict resolution scheme that makes more sense. Please note that we are intentionally leaving conflict resolution to the implementer; most conflicts are simply syntax errors, and the wishy-washy language here allows a receiver to be generous in what they accept, while putting pressure on a creator to be stingy in what they generate.

Some apparent conflicts may actually make sense -- for example, suppose a keyholder has an V3 key and a V4 key that share the same RSA key material. Either of these keys can verify a signature created by the other, and it may be reasonable for a signature to contain an issuer subpacket for each key, as a way of explicitly tying those keys to the signature.

5.3. Symmetric-Key Encrypted Session-Key Packets (Tag 3)

The Symmetric-Key Encrypted Session Key packet holds the symmetric-key encryption of a session key used to encrypt a message. Zero or more Encrypted Session Key packets and/or Symmetric-Key Encrypted Session Key packets may precede a Symmetrically Encrypted Data Packet that holds an encrypted message. The message is encrypted with a session key, and the session key is itself encrypted and stored in the Encrypted Session Key packet or the Symmetric-Key Encrypted Session Key packet.

If the Symmetrically Encrypted Data Packet is preceded by one or more Symmetric-Key Encrypted Session Key packets, each specifies a passphrase that may be used to decrypt the message. This allows a

Microsoft Exhibit 1019

message to be encrypted to a number of public keys, and also to one or more pass phrases. This packet type is new, and is not generated by PGP 2.x or PGP 5.0.

The body of this packet consists of:

- A one-octet version number. The only currently defined version is 4.
- A one-octet number describing the symmetric algorithm used.
- A string-to-key (S2K) specifier, length as defined above.
- Optionally, the encrypted session key itself, which is decrypted with the string-to-key object.

If the encrypted session key is not present (which can be detected on the basis of packet length and S2K specifier size), then the S2K algorithm applied to the passphrase produces the session key for decrypting the file, using the symmetric cipher algorithm from the Symmetric-Key Encrypted Session Key packet.

If the encrypted session key is present, the result of applying the S2K algorithm to the passphrase is used to decrypt just that encrypted session key field, using CFB mode with an IV of all zeros. The decryption result consists of a one-octet algorithm identifier that specifies the symmetric-key encryption algorithm used to encrypt the following Symmetrically Encrypted Data Packet, followed by the session key octets themselves.

Note: because an all-zero IV is used for this decryption, the S2K specifier MUST use a salt value, either a Salted S2K or an Iterated-Salted S2K. The salt value will insure that the decryption key is not repeated even if the passphrase is reused.

5.4. One-Pass Signature Packets (Tag 4)

The One-Pass Signature packet precedes the signed data and contains enough information to allow the receiver to begin calculating any hashes needed to verify the signature. It allows the Signature Packet to be placed at the end of the message, so that the signer can compute the entire signed message in one pass.

A One-Pass Signature does not interoperate with PGP 2.6.x or earlier.

The body of this packet consists of:

Microsoft Exhibit 1019

- A one-octet version number. The current version is 3.

- A one-octet signature type. Signature types are described in section 5.2.1.

- A one-octet number describing the hash algorithm used.

- A one-octet number describing the public key algorithm used.

- An eight-octet number holding the key ID of the signing key.

- A one-octet number holding a flag showing whether the signature is nested. A zero value indicates that the next packet is another One-Pass Signature packet that describes another signature to be applied to the same message data.

Note that if a message contains more than one one-pass signature, then the signature packets bracket the message; that is, the first signature packet after the message corresponds to the last one-pass packet and the final signature packet corresponds to the first one-pass packet.

5.5. Key Material Packet

A key material packet contains all the information about a public or private key. There are four variants of this packet type, and two major versions. Consequently, this section is complex.

5.5.1. Key Packet Variants

5.5.1.1. Public Key Packet (Tag 6)

A Public Key packet starts a series of packets that forms an OpenPGP key (sometimes called an OpenPGP certificate).

5.5.1.2. Public Subkey Packet (Tag 14)

A Public Subkey packet (tag 14) has exactly the same format as a Public Key packet, but denotes a subkey. One or more subkeys may be associated with a top-level key. By convention, the top-level key provides signature services, and the subkeys provide encryption services.

Note: in PGP 2.6.x, tag 14 was intended to indicate a comment packet. This tag was selected for reuse because no previous version of PGP ever emitted comment packets but they did properly ignore them. Public Subkey packets are ignored by PGP 2.6.x and do not cause it to fail, providing a limited degree of backward compatibility.

Microsoft Exhibit 1019

5.5.1.3. Secret Key Packet (Tag 5)

A Secret Key packet contains all the information that is found in a Public Key packet, including the public key material, but also includes the secret key material after all the public key fields.

5.5.1.4. Secret Subkey Packet (Tag 7)

A Secret Subkey packet (tag 7) is the subkey analog of the Secret Key packet, and has exactly the same format.

5.5.2. Public Key Packet Formats

There are two versions of key-material packets. Version 3 packets were first generated by PGP 2.6. Version 2 packets are identical in format to Version 3 packets, but are generated by PGP 2.5 or before. V2 packets are deprecated and they MUST NOT be generated. PGP 5.0 introduced version 4 packets, with new fields and semantics. PGP 2.6.x will not accept key-material packets with versions greater than 3.

OpenPGP implementations SHOULD create keys with version 4 format. An implementation MAY generate a V3 key to ensure interoperability with old software; note, however, that V4 keys correct some security deficiencies in V3 keys. These deficiencies are described below. An implementation MUST NOT create a V3 key with a public key algorithm other than RSA.

A version 3 public key or public subkey packet contains:

- A one-octet version number (3).
- A four-octet number denoting the time that the key was created.
- A two-octet number denoting the time in days that this key is valid. If this number is zero, then it does not expire.
- A one-octet number denoting the public key algorithm of this key
- A series of multi-precision integers comprising the key material:
    - a multiprecision integer (MPI) of RSA public modulus n;
    - an MPI of RSA public encryption exponent e.

Microsoft Exhibit 1019

V3 keys SHOULD only be used for backward compatibility because of three weaknesses in them. First, it is relatively easy to construct a V3 key that has the same key ID as any other key because the key ID is simply the low 64 bits of the public modulus. Secondly, because the fingerprint of a V3 key hashes the key material, but not its length, which increases the opportunity for fingerprint collisions. Third, there are minor weaknesses in the MD5 hash algorithm that make developers prefer other algorithms. See below for a fuller discussion of key IDs and fingerprints.

The version 4 format is similar to the version 3 format except for the absence of a validity period. This has been moved to the signature packet. In addition, fingerprints of version 4 keys are calculated differently from version 3 keys, as described in section "Enhanced Key Formats."

A version 4 packet contains:

- A one-octet version number (4).

- A four-octet number denoting the time that the key was created.

- A one-octet number denoting the public key algorithm of this key

- A series of multi-precision integers comprising the key material. This algorithm-specific portion is:

  Algorithm Specific Fields for RSA public keys:

  - multiprecision integer (MPI) of RSA public modulus n;

  - MPI of RSA public encryption exponent e.

  Algorithm Specific Fields for DSA public keys:

  - MPI of DSA prime p;

  - MPI of DSA group order q (q is a prime divisor of p-1);

  - MPI of DSA group generator g;

  - MPI of DSA public key value y (= g**x where x is secret).

  Algorithm Specific Fields for Elgamal public keys:

  - MPI of Elgamal prime p;

  - MPI of Elgamal group generator g;

Microsoft Exhibit 1019

- MPI of Elgamal public key value y (= g**x where x is secret).

5.5.3. Secret Key Packet Formats

The Secret Key and Secret Subkey packets contain all the data of the Public Key and Public Subkey packets, with additional algorithm-specific secret key data appended, in encrypted form.

The packet contains:

- A Public Key or Public Subkey packet, as described above
- One octet indicating string-to-key usage conventions. 0 indicates that the secret key data is not encrypted. 255 indicates that a string-to-key specifier is being given. Any other value is a symmetric-key encryption algorithm specifier.
- [Optional] If string-to-key usage octet was 255, a one-octet symmetric encryption algorithm.
- [Optional] If string-to-key usage octet was 255, a string-to-key specifier. The length of the string-to-key specifier is implied by its type, as described above.
- [Optional] If secret data is encrypted, eight-octet Initial Vector (IV).
- Encrypted multi-precision integers comprising the secret key data. These algorithm-specific fields are as described below.
- Two-octet checksum of the plaintext of the algorithm-specific portion (sum of all octets, mod 65536).

  Algorithm Specific Fields for RSA secret keys:

  - multiprecision integer (MPI) of RSA secret exponent d.
  - MPI of RSA secret prime value p.
  - MPI of RSA secret prime value q (p < q).
  - MPI of u, the multiplicative inverse of p, mod q.

  Algorithm Specific Fields for DSA secret keys:

  - MPI of DSA secret exponent x.

Microsoft Exhibit 1019

Algorithm Specific Fields for Elgamal secret keys:

- MPI of Elgamal secret exponent x.

Secret MPI values can be encrypted using a passphrase. If a string-to-key specifier is given, that describes the algorithm for converting the passphrase to a key, else a simple MD5 hash of the passphrase is used. Implementations SHOULD use a string-to-key specifier; the simple hash is for backward compatibility. The cipher for encrypting the MPIs is specified in the secret key packet.

Encryption/decryption of the secret data is done in CFB mode using the key created from the passphrase and the Initial Vector from the packet. A different mode is used with V3 keys (which are only RSA) than with other key formats. With V3 keys, the MPI bit count prefix (i.e., the first two octets) is not encrypted. Only the MPI non-prefix data is encrypted. Furthermore, the CFB state is resynchronized at the beginning of each new MPI value, so that the CFB block boundary is aligned with the start of the MPI data.

With V4 keys, a simpler method is used. All secret MPI values are encrypted in CFB mode, including the MPI bitcount prefix.

The 16-bit checksum that follows the algorithm-specific portion is the algebraic sum, mod 65536, of the plaintext of all the algorithm-specific octets (including MPI prefix and data). With V3 keys, the checksum is stored in the clear. With V4 keys, the checksum is encrypted like the algorithm-specific data. This value is used to check that the passphrase was correct.

5.6. Compressed Data Packet (Tag 8)

The Compressed Data packet contains compressed data. Typically, this packet is found as the contents of an encrypted packet, or following a Signature or One-Pass Signature packet, and contains literal data packets.

The body of this packet consists of:

- One octet that gives the algorithm used to compress the packet.

- The remainder of the packet is compressed data.

A Compressed Data Packet’s body contains an block that compresses some set of packets. See section "Packet Composition" for details on how messages are formed.

Microsoft Exhibit 1019

ZIP-compressed packets are compressed with raw RFC 1951 DEFLATE blocks. Note that PGP V2.6 uses 13 bits of compression. If an implementation uses more bits of compression, PGP V2.6 cannot decompress it.

ZLIB-compressed packets are compressed with RFC 1950 ZLIB-style blocks.

5.7. Symmetrically Encrypted Data Packet (Tag 9)

The Symmetrically Encrypted Data packet contains data encrypted with a symmetric-key algorithm. When it has been decrypted, it will typically contain other packets (often literal data packets or compressed data packets).

The body of this packet consists of:

- Encrypted data, the output of the selected symmetric-key cipher operating in PGP's variant of Cipher Feedback (CFB) mode.

The symmetric cipher used may be specified in an Public-Key or Symmetric-Key Encrypted Session Key packet that precedes the Symmetrically Encrypted Data Packet. In that case, the cipher algorithm octet is prefixed to the session key before it is encrypted. If no packets of these types precede the encrypted data, the IDEA algorithm is used with the session key calculated as the MD5 hash of the passphrase.

The data is encrypted in CFB mode, with a CFB shift size equal to the cipher's block size. The Initial Vector (IV) is specified as all zeros. Instead of using an IV, OpenPGP prefixes a 10-octet string to the data before it is encrypted. The first eight octets are random, and the 9th and 10th octets are copies of the 7th and 8th octets, respectively. After encrypting the first 10 octets, the CFB state is resynchronized if the cipher block size is 8 octets or less. The last 8 octets of ciphertext are passed through the cipher and the block boundary is reset.

The repetition of 16 bits in the 80 bits of random data prefixed to the message allows the receiver to immediately check whether the session key is incorrect.

5.8. Marker Packet (Obsolete Literal Packet) (Tag 10)

An experimental version of PGP used this packet as the Literal packet, but no released version of PGP generated Literal packets with this tag. With PGP 5.x, this packet has been re-assigned and is reserved for use as the Marker packet.

Microsoft Exhibit 1019

The body of this packet consists of:

- The three octets 0x50, 0x47, 0x50 (which spell "PGP" in UTF-8).

Such a packet MUST be ignored when received. It may be placed at the beginning of a message that uses features not available in PGP 2.6.x in order to cause that version to report that newer software is necessary to process the message.

5.9. Literal Data Packet (Tag 11)

A Literal Data packet contains the body of a message; data that is not to be further interpreted.

The body of this packet consists of:

- A one-octet field that describes how the data is formatted.

If it is a 'b' (0x62), then the literal packet contains binary data. If it is a 't' (0x74), then it contains text data, and thus may need line ends converted to local form, or other text-mode changes. RFC 1991 also defined a value of 'l' as a 'local' mode for machine-local conversions. This use is now deprecated.

- File name as a string (one-octet length, followed by file name), if the encrypted data should be saved as a file.

If the special name "_CONSOLE" is used, the message is considered to be "for your eyes only". This advises that the message data is unusually sensitive, and the receiving program should process it more carefully, perhaps avoiding storing the received data to disk, for example.

- A four-octet number that indicates the modification date of the file, or the creation time of the packet, or a zero that indicates the present time.

- The remainder of the packet is literal data.

Text data is stored with <CR><LF> text endings (i.e. network-normal line endings). These should be converted to native line endings by the receiving software.

5.10. Trust Packet (Tag 12)

The Trust packet is used only within keyrings and is not normally exported. Trust packets contain data that record the user's specifications of which key holders are trustworthy introducers,

Microsoft Exhibit 1019

along with other information that implementing software uses for trust information.

Trust packets SHOULD NOT be emitted to output streams that are transferred to other users, and they SHOULD be ignored on any input other than local keyring files.

5.11. User ID Packet (Tag 13)

A User ID packet consists of data that is intended to represent the name and email address of the key holder. By convention, it includes an RFC 822 mail name, but there are no restrictions on its content. The packet length in the header specifies the length of the user id. If it is text, it is encoded in UTF-8.

6. Radix-64 Conversions

As stated in the introduction, OpenPGP's underlying native representation for objects is a stream of arbitrary octets, and some systems desire these objects to be immune to damage caused by character set translation, data conversions, etc.

In principle, any printable encoding scheme that met the requirements of the unsafe channel would suffice, since it would not change the underlying binary bit streams of the native OpenPGP data structures. The OpenPGP standard specifies one such printable encoding scheme to ensure interoperability.

OpenPGP's Radix-64 encoding is composed of two parts: a base64 encoding of the binary data, and a checksum. The base64 encoding is identical to the MIME base64 content-transfer-encoding [RFC2231, Section 6.8]. An OpenPGP implementation MAY use ASCII Armor to protect the raw binary data.

The checksum is a 24-bit CRC converted to four characters of radix-64 encoding by the same MIME base64 transformation, preceded by an equals sign (=). The CRC is computed by using the generator 0x864CFB and an initialization of 0xB704CE. The accumulation is done on the data before it is converted to radix-64, rather than on the converted data. A sample implementation of this algorithm is in the next section.

The checksum with its leading equal sign MAY appear on the first line after the Base64 encoded data.

Rationale for CRC-24: The size of 24 bits fits evenly into printable base64. The nonzero initialization can detect more errors than a zero initialization.

Microsoft Exhibit 1019

6.1. An Implementation of the CRC-24 in "C"

```
#define CRC24_INIT 0xb704ceL
#define CRC24_POLY 0x1864cfbL

typedef long crc24;
crc24 crc_octets(unsigned char *octets, size_t len)
{
    crc24 crc = CRC24_INIT;
    int i;

    while (len--) {
        crc ^= (*octets++) << 16;
        for (i = 0; i < 8; i++) {
            crc <<= 1;
            if (crc & 0x1000000)
                crc ^= CRC24_POLY;
        }
    }
    return crc & 0xffffffL;
}
```

6.2. Forming ASCII Armor

When OpenPGP encodes data into ASCII Armor, it puts specific headers around the data, so OpenPGP can reconstruct the data later. OpenPGP informs the user what kind of data is encoded in the ASCII armor through the use of the headers.

Concatenating the following data creates ASCII Armor:

- An Armor Header Line, appropriate for the type of data
- Armor Headers
- A blank (zero-length, or containing only whitespace) line
- The ASCII-Armored data
- An Armor Checksum
- The Armor Tail, which depends on the Armor Header Line.

An Armor Header Line consists of the appropriate header line text surrounded by five (5) dashes ('-', 0x2D) on either side of the header line text. The header line text is chosen based upon the type of data that is being encoded in Armor, and how it is being encoded. Header line texts include the following strings:

Microsoft Exhibit 1019

BEGIN PGP MESSAGE
Used for signed, encrypted, or compressed files.

BEGIN PGP PUBLIC KEY BLOCK
Used for armoring public keys

BEGIN PGP PRIVATE KEY BLOCK
Used for armoring private keys

BEGIN PGP MESSAGE, PART X/Y
Used for multi-part messages, where the armor is split amongst Y parts, and this is the Xth part out of Y.

BEGIN PGP MESSAGE, PART X
Used for multi-part messages, where this is the Xth part of an unspecified number of parts. Requires the MESSAGE-ID Armor Header to be used.

BEGIN PGP SIGNATURE
Used for detached signatures, OpenPGP/MIME signatures, and natures following clearsigned messages. Note that PGP 2.x s BEGIN PGP MESSAGE for detached signatures.

The Armor Headers are pairs of strings that can give the user or the receiving OpenPGP implementation some information about how to decode or use the message. The Armor Headers are a part of the armor, not a part of the message, and hence are not protected by any signatures applied to the message.

The format of an Armor Header is that of a key-value pair. A colon (':' 0x38) and a single space (0x20) separate the key and value. OpenPGP should consider improperly formatted Armor Headers to be corruption of the ASCII Armor. Unknown keys should be reported to the user, but OpenPGP should continue to process the message.

Currently defined Armor Header Keys are:

- "Version", that states the OpenPGP Version used to encode the message.

- "Comment", a user-defined comment.

- "MessageID", a 32-character string of printable characters. The string must be the same for all parts of a multi-part message that uses the "PART X" Armor Header. MessageID strings should be

Microsoft Exhibit 1019

unique enough that the recipient of the mail can associate all the parts of a message with each other. A good checksum or cryptographic hash function is sufficient.

- "Hash", a comma-separated list of hash algorithms used in this message. This is used only in clear-signed messages.

- "Charset", a description of the character set that the plaintext is in. Please note that OpenPGP defines text to be in UTF-8 by default. An implementation will get best results by translating into and out of UTF-8. However, there are many instances where this is easier said than done. Also, there are communities of users who have no need for UTF-8 because they are all happy with a character set like ISO Latin-5 or a Japanese character set. In such instances, an implementation MAY override the UTF-8 default by using this header key. An implementation MAY implement this key and any translations it cares to; an implementation MAY ignore it and assume all text is UTF-8.

  The MessageID SHOULD NOT appear unless it is in a multi-part message. If it appears at all, it MUST be computed from the finished (encrypted, signed, etc.) message in a deterministic fashion, rather than contain a purely random value. This is to allow the legitimate recipient to determine that the MessageID cannot serve as a covert means of leaking cryptographic key information.

The Armor Tail Line is composed in the same manner as the Armor Header Line, except the string "BEGIN" is replaced by the string "END."

6.3. Encoding Binary in Radix-64

The encoding process represents 24-bit groups of input bits as output strings of 4 encoded characters. Proceeding from left to right, a 24-bit input group is formed by concatenating three 8-bit input groups. These 24 bits are then treated as four concatenated 6-bit groups, each of which is translated into a single digit in the Radix-64 alphabet. When encoding a bit stream with the Radix-64 encoding, the bit stream must be presumed to be ordered with the most-significant-bit first. That is, the first bit in the stream will be the high-order bit in the first 8-bit octet, and the eighth bit will be the low-order bit in the first 8-bit octet, and so on.

Microsoft Exhibit 1019

```
+--first octet--+-second octet--+--third octet--+
|7 6 5 4 3 2 1 0|7 6 5 4 3 2 1 0|7 6 5 4 3 2 1 0|
+-----------+---+-------+-------+---+-----------+
|5 4 3 2 1 0|5 4 3 2 1 0|5 4 3 2 1 0|5 4 3 2 1 0|
+--1.index--+--2.index--+--3.index--+--4.index--+
```

Each 6-bit group is used as an index into an array of 64 printable characters from the table below. The character referenced by the index is placed in the output string.

Value	Encoding	Value	Encoding	Value	Encoding	Value	Encoding
0	A	17	R	34	i	51	z
1	B	18	S	35	j	52	0
2	C	19	T	36	k	53	1
3	D	20	U	37	l	54	2
4	E	21	V	38	m	55	3
5	F	22	W	39	n	56	4
6	G	23	X	40	o	57	5
7	H	24	Y	41	p	58	6
8	I	25	Z	42	q	59	7
9	J	26	a	43	r	60	8
10	K	27	b	44	s	61	9
11	L	28	c	45	t	62	+
12	M	29	d	46	u	63	/
13	N	30	e	47	v		
14	O	31	f	48	w	(pad)	=
15	P	32	g	49	x		
16	Q	33	h	50	y		

The encoded output stream must be represented in lines of no more than 76 characters each.

Special processing is performed if fewer than 24 bits are available at the end of the data being encoded. There are three possibilities:

1. The last data group has 24 bits (3 octets). No special processing is needed.

2. The last data group has 16 bits (2 octets). The first two 6-bit groups are processed as above. The third (incomplete) data group has two zero-value bits added to it, and is processed as above. A pad character (=) is added to the output.

3. The last data group has 8 bits (1 octet). The first 6-bit group is processed as above. The second (incomplete) data group has four zero-value bits added to it, and is processed as above. Two pad characters (=) are added to the output.

Microsoft Exhibit 1019

6.4. Decoding Radix-64

Any characters outside of the base64 alphabet are ignored in Radix-64 data. Decoding software must ignore all line breaks or other characters not found in the table above.

In Radix-64 data, characters other than those in the table, line breaks, and other white space probably indicate a transmission error, about which a warning message or even a message rejection might be appropriate under some circumstances.

Because it is used only for padding at the end of the data, the occurrence of any "=" characters may be taken as evidence that the end of the data has been reached (without truncation in transit). No such assurance is possible, however, when the number of octets transmitted was a multiple of three and no "=" characters are present.

6.5. Examples of Radix-64

```
Input data:  0x14fb9c03d97e
Hex:     1   4    f   b    9   c     | 0   3    d   9    7   e
8-bit:   00010100 11111011 10011100  | 00000011 11011001
11111110
6-bit:   000101 001111 101110 011100 | 000000 111101 100111
111110
Decimal: 5      15     46     28       0      61     37     62
Output:  F      P      u      c        A      9      l      +

Input data:  0x14fb9c03d9
Hex:     1   4    f   b    9   c     | 0   3    d   9
8-bit:   00010100 11111011 10011100  | 00000011 11011001
                                       pad with 00
6-bit:   000101 001111 101110 011100 | 000000 111101 100100
Decimal: 5      15     46     28       0      61     36
                                           pad with =
Output:  F      P      u      c        A      9      k      =

Input data:  0x14fb9c03
Hex:     1   4    f   b    9   c     | 0   3
8-bit:   00010100 11111011 10011100  | 00000011
                                       pad with 0000
6-bit:   000101 001111 101110 011100 | 000000 110000
Decimal: 5      15     46     28       0      48
                                        pad with =      =
Output:  F      P      u      c        A      w      =      =
```


Microsoft Exhibit 1019

6.6. Example of an ASCII Armored Message

```
-----BEGIN PGP MESSAGE-----
Version: OpenPrivacy 0.99

yDgBO22WxBHv7O8X7O/jygAEzol56iUKiXmV+XmpCtmpqQUKiQrFqclFqUDBovzS
vBSFjNSiVHsuAA==
=njUN
-----END PGP MESSAGE-----
```

Note that this example is indented by two spaces.

7. Cleartext signature framework

It is desirable to sign a textual octet stream without ASCII armoring the stream itself, so the signed text is still readable without special software. In order to bind a signature to such a cleartext, this framework is used. (Note that RFC 2015 defines another way to clear sign messages for environments that support MIME.)

The cleartext signed message consists of:

- The cleartext header '-----BEGIN PGP SIGNED MESSAGE-----' on a single line,
- One or more "Hash" Armor Headers,
- Exactly one empty line not included into the message digest,
- The dash-escaped cleartext that is included into the message digest,
- The ASCII armored signature(s) including the '-----BEGIN PGP SIGNATURE-----' Armor Header and Armor Tail Lines.

If the "Hash" armor header is given, the specified message digest algorithm is used for the signature. If there are no such headers, MD5 is used, an implementation MAY omit them for V2.x compatibility. If more than one message digest is used in the signature, the "Hash" armor header contains a comma-delimited list of used message digests.

Current message digest names are described below with the algorithm IDs.

7.1. Dash-Escaped Text

The cleartext content of the message must also be dash-escaped.

Microsoft Exhibit 1019

Dash escaped cleartext is the ordinary cleartext where every line starting with a dash '-' (0x2D) is prefixed by the sequence dash '-' (0x2D) and space ' ' (0x20). This prevents the parser from recognizing armor headers of the cleartext itself. The message digest is computed using the cleartext itself, not the dash escaped form.

As with binary signatures on text documents, a cleartext signature is calculated on the text using canonical <CR><LF> line endings. The line ending (i.e. the <CR><LF>) before the '-----BEGIN PGP SIGNATURE-----' line that terminates the signed text is not considered part of the signed text.

Also, any trailing whitespace (spaces, and tabs, 0x09) at the end of any line is ignored when the cleartext signature is calculated.

8. Regular Expressions

A regular expression is zero or more branches, separated by '|'. It matches anything that matches one of the branches.

A branch is zero or more pieces, concatenated. It matches a match for the first, followed by a match for the second, etc.

A piece is an atom possibly followed by '*', '+', or '?'. An atom followed by '*' matches a sequence of 0 or more matches of the atom. An atom followed by '+' matches a sequence of 1 or more matches of the atom. An atom followed by '?' matches a match of the atom, or the null string.

An atom is a regular expression in parentheses (matching a match for the regular expression), a range (see below), '.' (matching any single character), '^' (matching the null string at the beginning of the input string), '$' (matching the null string at the end of the input string), a '\' followed by a single character (matching that character), or a single character with no other significance (matching that character).

A range is a sequence of characters enclosed in '[]'. It normally matches any single character from the sequence. If the sequence begins with '^', it matches any single character not from the rest of the sequence. If two characters in the sequence are separated by '-', this is shorthand for the full list of ASCII characters between them (e.g. '[0-9]' matches any decimal digit). To include a literal ']' in the sequence, make it the first character (following a possible '^'). To include a literal '-', make it the first or last character.

Microsoft Exhibit 1019

9. Constants

This section describes the constants used in OpenPGP.

Note that these tables are not exhaustive lists; an implementation MAY implement an algorithm not on these lists.

See the section "Notes on Algorithms" below for more discussion of the algorithms.

9.1. Public Key Algorithms

ID	Algorithm
1	- RSA (Encrypt or Sign)
2	- RSA Encrypt-Only
3	- RSA Sign-Only
16	- Elgamal (Encrypt-Only), see [ELGAMAL]
17	- DSA (Digital Signature Standard)
18	- Reserved for Elliptic Curve
19	- Reserved for ECDSA
20	- Elgamal (Encrypt or Sign)
21	- Reserved for Diffie-Hellman (X9.42, as defined for IETF-S/MIME)
100 to 110	- Private/Experimental algorithm.

Implementations MUST implement DSA for signatures, and Elgamal for encryption. Implementations SHOULD implement RSA keys. Implementations MAY implement any other algorithm.

9.2. Symmetric Key Algorithms

ID	Algorithm
0	- Plaintext or unencrypted data
1	- IDEA [IDEA]
2	- Triple-DES (DES-EDE, as per spec - 168 bit key derived from 192)
3	- CAST5 (128 bit key, as per RFC 2144)
4	- Blowfish (128 bit key, 16 rounds) [BLOWFISH]
5	- SAFER-SK128 (13 rounds) [SAFER]
6	- Reserved for DES/SK
7	- Reserved for AES with 128-bit key

Microsoft Exhibit 1019

```
      8          - Reserved for AES with 192-bit key
      9          - Reserved for AES with 256-bit key
      100 to 110 - Private/Experimental algorithm.
```

Implementations MUST implement Triple-DES. Implementations SHOULD implement IDEA and CAST5.Implementations MAY implement any other algorithm.

9.3. Compression Algorithms

```
      ID           Algorithm
      --           ---------
      0          - Uncompressed
      1          - ZIP (RFC 1951)
      2          - ZLIB (RFC 1950)
      100 to 110 - Private/Experimental algorithm.
```

Implementations MUST implement uncompressed data. Implementations SHOULD implement ZIP. Implementations MAY implement ZLIB.

9.4. Hash Algorithms

```
      ID           Algorithm                                Text Name
      --           ---------                                ---- ----
      1          - MD5                                      "MD5"
      2          - SHA-1                                    "SHA1"
      3          - RIPE-MD/160                              "RIPEMD160"
      4          - Reserved for double-width SHA (experimental)
      5          - MD2                                      "MD2"
      6          - Reserved for TIGER/192                   "TIGER192"
      7          - Reserved for HAVAL (5 pass, 160-bit)
      "HAVAL-5-160"
      100 to 110 - Private/Experimental algorithm.
```

Implementations MUST implement SHA-1. Implementations SHOULD implement MD5.

10. Packet Composition

OpenPGP packets are assembled into sequences in order to create messages and to transfer keys. Not all possible packet sequences are meaningful and correct. This describes the rules for how packets should be placed into sequences.

10.1. Transferable Public Keys

OpenPGP users may transfer public keys. The essential elements of a transferable public key are:

Microsoft Exhibit 1019

- One Public Key packet
- Zero or more revocation signatures
- One or more User ID packets
- After each User ID packet, zero or more signature packets (certifications)
- Zero or more Subkey packets
- After each Subkey packet, one signature packet, optionally a revocation.

The Public Key packet occurs first. Each of the following User ID packets provides the identity of the owner of this public key. If there are multiple User ID packets, this corresponds to multiple means of identifying the same unique individual user; for example, a user may have more than one email address, and construct a User ID for each one.

Immediately following each User ID packet, there are zero or more signature packets. Each signature packet is calculated on the immediately preceding User ID packet and the initial Public Key packet. The signature serves to certify the corresponding public key and user ID. In effect, the signer is testifying to his or her belief that this public key belongs to the user identified by this user ID.

After the User ID packets there may be one or more Subkey packets. In general, subkeys are provided in cases where the top-level public key is a signature-only key. However, any V4 key may have subkeys, and the subkeys may be encryption-only keys, signature-only keys, or general-purpose keys.

Each Subkey packet must be followed by one Signature packet, which should be a subkey binding signature issued by the top level key.

Subkey and Key packets may each be followed by a revocation Signature packet to indicate that the key is revoked. Revocation signatures are only accepted if they are issued by the key itself, or by a key that is authorized to issue revocations via a revocation key subpacket in a self-signature by the top level key.

Transferable public key packet sequences may be concatenated to allow transferring multiple public keys in one operation.

Microsoft Exhibit 1019

10.2. OpenPGP Messages

An OpenPGP message is a packet or sequence of packets that corresponds to the following grammatical rules (comma represents sequential composition, and vertical bar separates alternatives):

OpenPGP Message :- Encrypted Message | Signed Message | Compressed Message | Literal Message.

Compressed Message :- Compressed Data Packet.

Literal Message :- Literal Data Packet.

ESK :- Public Key Encrypted Session Key Packet | Symmetric-Key Encrypted Session Key Packet.

ESK Sequence :- ESK | ESK Sequence, ESK.

Encrypted Message :- Symmetrically Encrypted Data Packet | ESK Sequence, Symmetrically Encrypted Data Packet.

One-Pass Signed Message :- One-Pass Signature Packet, OpenPGP Message, Corresponding Signature Packet.

Signed Message :- Signature Packet, OpenPGP Message | One-Pass Signed Message.

In addition, decrypting a Symmetrically Encrypted Data packet and

decompressing a Compressed Data packet must yield a valid OpenPGP Message.

10.3. Detached Signatures

Some OpenPGP applications use so-called "detached signatures." For example, a program bundle may contain a file, and with it a second file that is a detached signature of the first file. These detached signatures are simply a signature packet stored separately from the data that they are a signature of.

11. Enhanced Key Formats

11.1. Key Structures

The format of an OpenPGP V3 key is as follows. Entries in square brackets are optional and ellipses indicate repetition.

Microsoft Exhibit 1019

```
RSA Public Key
   [Revocation Self Signature]
    User ID [Signature ...]
   [User ID [Signature ...] ...]
```

Each signature certifies the RSA public key and the preceding user ID. The RSA public key can have many user IDs and each user ID can have many signatures.

The format of an OpenPGP V4 key that uses two public keys is similar except that the other keys are added to the end as 'subkeys' of the primary key.

```
Primary-Key
   [Revocation Self Signature]
   [Direct Key Self Signature...]
    User ID [Signature ...]
   [User ID [Signature ...] ...]
   [[Subkey [Binding-Signature-Revocation]
           Primary-Key-Binding-Signature] ...]
```

A subkey always has a single signature after it that is issued using the primary key to tie the two keys together. This binding signature may be in either V3 or V4 format, but V4 is preferred, of course.

In the above diagram, if the binding signature of a subkey has been revoked, the revoked binding signature may be removed, leaving only one signature.

In a key that has a main key and subkeys, the primary key MUST be a key capable of signing. The subkeys may be keys of any other type. There may be other constructions of V4 keys, too. For example, there may be a single-key RSA key in V4 format, a DSA primary key with an RSA encryption key, or RSA primary key with an Elgamal subkey, etc.

It is also possible to have a signature-only subkey. This permits a primary key that collects certifications (key signatures) but is used only used for certifying subkeys that are used for encryption and signatures.

11.2. Key IDs and Fingerprints

For a V3 key, the eight-octet key ID consists of the low 64 bits of the public modulus of the RSA key.

The fingerprint of a V3 key is formed by hashing the body (but not the two-octet length) of the MPIs that form the key material (public modulus n, followed by exponent e) with MD5.

Microsoft Exhibit 1019

A V4 fingerprint is the 160-bit SHA-1 hash of the one-octet Packet Tag, followed by the two-octet packet length, followed by the entire Public Key packet starting with the version field. The key ID is the low order 64 bits of the fingerprint. Here are the fields of the hash material, with the example of a DSA key:

a.1) 0x99 (1 octet)

a.2) high order length octet of (b)-(f) (1 octet)

a.3) low order length octet of (b)-(f) (1 octet)

b) version number = 4 (1 octet);

c) time stamp of key creation (4 octets);

d) algorithm (1 octet): 17 = DSA (example);

e) Algorithm specific fields.

Algorithm Specific Fields for DSA keys (example):

e.1) MPI of DSA prime p;

e.2) MPI of DSA group order q (q is a prime divisor of p-1);

e.3) MPI of DSA group generator g;

e.4) MPI of DSA public key value y (= g**x where x is secret).

Note that it is possible for there to be collisions of key IDs -- two different keys with the same key ID. Note that there is a much smaller, but still non-zero probability that two different keys have the same fingerprint.

Also note that if V3 and V4 format keys share the same RSA key material, they will have different key ids as well as different fingerprints.

12. Notes on Algorithms

12.1. Symmetric Algorithm Preferences

The symmetric algorithm preference is an ordered list of algorithms that the keyholder accepts. Since it is found on a self-signature, it is possible that a keyholder may have different preferences. For example, Alice may have TripleDES only specified for "alice@work.com" but CAST5, Blowfish, and TripleDES specified for "alice@home.org".

Microsoft Exhibit 1019

Note that it is also possible for preferences to be in a subkey's binding signature.

Since TripleDES is the MUST-implement algorithm, if it is not explicitly in the list, it is tacitly at the end. However, it is good form to place it there explicitly. Note also that if an implementation does not implement the preference, then it is implicitly a TripleDES-only implementation.

An implementation MUST not use a symmetric algorithm that is not in the recipient's preference list. When encrypting to more than one recipient, the implementation finds a suitable algorithm by taking the intersection of the preferences of the recipients. Note that the MUST-implement algorithm, TripleDES, ensures that the intersection is not null. The implementation may use any mechanism to pick an algorithm in the intersection.

If an implementation can decrypt a message that a keyholder doesn't have in their preferences, the implementation SHOULD decrypt the message anyway, but MUST warn the keyholder than protocol has been violated. (For example, suppose that Alice, above, has software that implements all algorithms in this specification. Nonetheless, she prefers subsets for work or home. If she is sent a message encrypted with IDEA, which is not in her preferences, the software warns her that someone sent her an IDEA-encrypted message, but it would ideally decrypt it anyway.)

An implementation that is striving for backward compatibility MAY consider a V3 key with a V3 self-signature to be an implicit preference for IDEA, and no ability to do TripleDES. This is technically non-compliant, but an implementation MAY violate the above rule in this case only and use IDEA to encrypt the message, provided that the message creator is warned. Ideally, though, the implementation would follow the rule by actually generating two messages, because it is possible that the OpenPGP user's implementation does not have IDEA, and thus could not read the message. Consequently, an implementation MAY, but SHOULD NOT use IDEA in an algorithm conflict with a V3 key.

12.2. Other Algorithm Preferences

Other algorithm preferences work similarly to the symmetric algorithm preference, in that they specify which algorithms the keyholder accepts. There are two interesting cases that other comments need to be made about, though, the compression preferences and the hash preferences.

Microsoft Exhibit 1019

12.2.1. Compression Preferences

Compression has been an integral part of PGP since its first days. OpenPGP and all previous versions of PGP have offered compression. And in this specification, the default is for messages to be compressed, although an implementation is not required to do so. Consequently, the compression preference gives a way for a keyholder to request that messages not be compressed, presumably because they are using a minimal implementation that does not include compression. Additionally, this gives a keyholder a way to state that it can support alternate algorithms.

Like the algorithm preferences, an implementation MUST NOT use an algorithm that is not in the preference vector. If the preferences are not present, then they are assumed to be [ZIP(1), UNCOMPRESSED(0)].

12.2.2. Hash Algorithm Preferences

Typically, the choice of a hash algorithm is something the signer does, rather than the verifier, because a signer does not typically know who is going to be verifying the signature. This preference, though, allows a protocol based upon digital signatures ease in negotiation.

Thus, if Alice is authenticating herself to Bob with a signature, it makes sense for her to use a hash algorithm that Bob's software uses. This preference allows Bob to state in his key which algorithms Alice may use.

12.3. Plaintext

Algorithm 0, "plaintext", may only be used to denote secret keys that are stored in the clear. Implementations must not use plaintext in Symmetrically Encrypted Data Packets; they must use Literal Data Packets to encode unencrypted or literal data.

12.4. RSA

There are algorithm types for RSA-signature-only, and RSA-encrypt-only keys. These types are deprecated. The "key flags" subpacket in a signature is a much better way to express the same idea, and generalizes it to all algorithms. An implementation SHOULD NOT create such a key, but MAY interpret it.

An implementation SHOULD NOT implement RSA keys of size less than 768 bits.

Microsoft Exhibit 1019

It is permissible for an implementation to support RSA merely for backward compatibility; for example, such an implementation would support V3 keys with IDEA symmetric cryptography. Note that this is an exception to the other MUST-implement rules. An implementation that supports RSA in V4 keys MUST implement the MUST-implement features.

12.5. Elgamal

If an Elgamal key is to be used for both signing and encryption, extra care must be taken in creating the key.

An ElGamal key consists of a generator g, a prime modulus p, a secret exponent x, and a public value y = g^x mod p.

The generator and prime must be chosen so that solving the discrete log problem is intractable. The group g should generate the multiplicative group mod p-1 or a large subgroup of it, and the order of g should have at least one large prime factor. A good choice is to use a "strong" Sophie-Germain prime in choosing p, so that both p and (p-1)/2 are primes. In fact, this choice is so good that implementors SHOULD do it, as it avoids a small subgroup attack.

In addition, a result of Bleichenbacher [BLEICHENBACHER] shows that if the generator g has only small prime factors, and if g divides the order of the group it generates, then signatures can be forged. In particular, choosing g=2 is a bad choice if the group order may be even. On the other hand, a generator of 2 is a fine choice for an encryption-only key, as this will make the encryption faster.

While verifying Elgamal signatures, note that it is important to test that r and s are less than p. If this test is not done then signatures can be trivially forged by using large r values of approximately twice the length of p. This attack is also discussed in the Bleichenbacher paper.

Details on safe use of Elgamal signatures may be found in [MENEZES], which discusses all the weaknesses described above.

If an implementation allows Elgamal signatures, then it MUST use the algorithm identifier 20 for an Elgamal public key that can sign.

An implementation SHOULD NOT implement Elgamal keys of size less than 768 bits. For long-term security, Elgamal keys should be 1024 bits or longer.

Microsoft Exhibit 1019

12.6. DSA

An implementation SHOULD NOT implement DSA keys of size less than 768 bits. Note that present DSA is limited to a maximum of 1024 bit keys, which are recommended for long-term use.

12.7. Reserved Algorithm Numbers

A number of algorithm IDs have been reserved for algorithms that would be useful to use in an OpenPGP implementation, yet there are issues that prevent an implementor from actually implementing the algorithm. These are marked in the Public Algorithms section as "(reserved for)".

The reserved public key algorithms, Elliptic Curve (18), ECDSA (19), and X9.42 (21) do not have the necessary parameters, parameter order, or semantics defined.

The reserved symmetric key algorithm, DES/SK (6), does not have semantics defined.

The reserved hash algorithms, TIGER192 (6), and HAVAL-5-160 (7), do not have OIDs. The reserved algorithm number 4, reserved for a double-width variant of SHA1, is not presently defined.

We have reserver three algorithm IDs for the US NIST’s Advanced Encryption Standard. This algorithm will work with (at least) 128, 192, and 256-bit keys. We expect that this algorithm will be selected from the candidate algorithms in the year 2000.

12.8. OpenPGP CFB mode

OpenPGP does symmetric encryption using a variant of Cipher Feedback Mode (CFB mode). This section describes the procedure it uses in detail. This mode is what is used for Symmetrically Encrypted Data Packets; the mechanism used for encrypting secret key material is similar, but described in those sections above.

OpenPGP CFB mode uses an initialization vector (IV) of all zeros, and prefixes the plaintext with ten octets of random data, such that octets 9 and 10 match octets 7 and 8. It does a CFB "resync" after encrypting those ten octets.

Note that for an algorithm that has a larger block size than 64 bits, the equivalent function will be done with that entire block. For example, a 16-octet block algorithm would operate on 16 octets, and then produce two octets of check, and then work on 16-octet blocks.

Microsoft Exhibit 1019

Step by step, here is the procedure:

1. The feedback register (FR) is set to the IV, which is all zeros.

2. FR is encrypted to produce FRE (FR Encrypted). This is the encryption of an all-zero value.

3. FRE is xored with the first 8 octets of random data prefixed to the plaintext to produce C1-C8, the first 8 octets of ciphertext.

4. FR is loaded with C1-C8.

5. FR is encrypted to produce FRE, the encryption of the first 8 octets of ciphertext.

6. The left two octets of FRE get xored with the next two octets of data that were prefixed to the plaintext. This produces C9-C10, the next two octets of ciphertext.

7. (The resync step) FR is loaded with C3-C10.

8. FR is encrypted to produce FRE.

9. FRE is xored with the first 8 octets of the given plaintext, now that we have finished encrypting the 10 octets of prefixed data. This produces C11-C18, the next 8 octets of ciphertext.

10. FR is loaded with C11-C18

11. FR is encrypted to produce FRE.

12. FRE is xored with the next 8 octets of plaintext, to produce the next 8 octets of ciphertext. These are loaded into FR and the process is repeated until the plaintext is used up.

13. Security Considerations

As with any technology involving cryptography, you should check the current literature to determine if any algorithms used here have been found to be vulnerable to attack.

This specification uses Public Key Cryptography technologies. Possession of the private key portion of a public-private key pair is assumed to be controlled by the proper party or parties.

Certain operations in this specification involve the use of random numbers. An appropriate entropy source should be used to generate these numbers. See RFC 1750.

Microsoft Exhibit 1019

The MD5 hash algorithm has been found to have weaknesses (pseudo-collisions in the compress function) that make some people deprecate its use. They consider the SHA-1 algorithm better.

Many security protocol designers think that it is a bad idea to use a single key for both privacy (encryption) and integrity (signatures). In fact, this was one of the motivating forces behind the V4 key format with separate signature and encryption keys. If you as an implementor promote dual-use keys, you should at least be aware of this controversy.

The DSA algorithm will work with any 160-bit hash, but it is sensitive to the quality of the hash algorithm, if the hash algorithm is broken, it can leak the secret key. The Digital Signature Standard (DSS) specifies that DSA be used with SHA-1. RIPEMD-160 is considered by many cryptographers to be as strong. An implementation should take care which hash algorithms are used with DSA, as a weak hash can not only allow a signature to be forged, but could leak the secret key. These same considerations about the quality of the hash algorithm apply to Elgamal signatures.

If you are building an authentication system, the recipient may specify a preferred signing algorithm. However, the signer would be foolish to use a weak algorithm simply because the recipient requests it.

Some of the encryption algorithms mentioned in this document have been analyzed less than others. For example, although CAST5 is presently considered strong, it has been analyzed less than Triple-DES. Other algorithms may have other controversies surrounding them.

Some technologies mentioned here may be subject to government control in some countries.

14. Implementation Nits

This section is a collection of comments to help an implementer, particularly with an eye to backward compatibility. Previous implementations of PGP are not OpenPGP-compliant. Often the differences are small, but small differences are frequently more vexing than large differences. Thus, this list of potential problems and gotchas for a developer who is trying to be backward-compatible.

* PGP 5.x does not accept V4 signatures for anything other than key material.

* PGP 5.x does not recognize the "five-octet" lengths in new-format headers or in signature subpacket lengths.

Microsoft Exhibit 1019

* PGP 5.0 rejects an encrypted session key if the keylength differs from the S2K symmetric algorithm. This is a bug in its validation function.

* PGP 5.0 does not handle multiple one-pass signature headers and trailers. Signing one will compress the one-pass signed literal and prefix a V3 signature instead of doing a nested one-pass signature.

* When exporting a private key, PGP 2.x generates the header "BEGIN PGP SECRET KEY BLOCK" instead of "BEGIN PGP PRIVATE KEY BLOCK". All previous versions ignore the implied data type, and look directly at the packet data type.

* In a clear-signed signature, PGP 5.0 will figure out the correct hash algorithm if there is no "Hash:" header, but it will reject a mismatch between the header and the actual algorithm used. The "standard" (i.e. Zimmermann/Finney/et al.) version of PGP 2.x rejects the "Hash:" header and assumes MD5. There are a number of enhanced variants of PGP 2.6.x that have been modified for SHA-1 signatures.

* PGP 5.0 can read an RSA key in V4 format, but can only recognize it with a V3 keyid, and can properly use only a V3 format RSA key.

* Neither PGP 5.x nor PGP 6.0 recognize Elgamal Encrypt and Sign keys. They only handle Elgamal Encrypt-only keys.

* There are many ways possible for two keys to have the same key material, but different fingerprints (and thus key ids). Perhaps the most interesting is an RSA key that has been "upgraded" to V4 format, but since a V4 fingerprint is constructed by hashing the key creation time along with other things, two V4 keys created at different times, yet with the same key material will have different fingerprints.

* If an implementation is using zlib to interoperate with PGP 2.x, then the "windowBits" parameter should be set to -13.

Microsoft Exhibit 1019

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Microsoft Exhibit 1019

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Microsoft Exhibit 1019

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Microsoft Exhibit 1019

Network Working Group
Request for Comments: 2660
Category: Experimental

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The Secure HyperText Transfer Protocol

Status of this Memo

This memo defines an Experimental Protocol for the Internet community. It does not specify an Internet standard of any kind. Discussion and suggestions for improvement are requested. Distribution of this memo is unlimited.

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Abstract

This memo describes a syntax for securing messages sent using the Hypertext Transfer Protocol (HTTP), which forms the basis for the World Wide Web. Secure HTTP (S-HTTP) provides independently applicable security services for transaction confidentiality, authenticity/integrity and non-repudiability of origin.

The protocol emphasizes maximum flexibility in choice of key management mechanisms, security policies and cryptographic algorithms by supporting option negotiation between parties for each transaction.

Table of Contents

1. Introduction 3
1.1. Summary of Features 3
1.2. Changes 4
1.3. Processing Model 5
1.4. Modes of Operation 6
1.5. Implementation Options 7
2. Message Format 7
2.1. Notational Conventions 8
2.2. The Request Line 8
2.3. The Status Line 8
2.4. Secure HTTP Header Lines 8
2.5. Content 12
2.6. Encapsulation Format Options 13

Microsoft Exhibit 1019

2.6.1. Content-Privacy-Domain: CMS 13
2.6.2. Content-Privacy-Domain: MOSS 14
2.6.3. Permitted HTTP headers 14
2.6.3.2. Host 15
2.6.3.3. Connection 15
3. Cryptographic Parameters 15
3.1. Options Headers 15
3.2. Negotiation Options 16
3.2.1. Negotiation Overview 16
3.2.2. Negotiation Option Format 16
3.2.3. Parametrization for Variable-length Key Ciphers 18
3.2.4. Negotiation Syntax 18
3.3. Non-Negotiation Headers 23
3.3.1. Encryption-Identity 23
3.3.2. Certificate-Info 23
3.3.3. Key-Assign 24
3.3.4. Nonces 25
3.4. Grouping Headers With SHTTP-Cryptopts 26
3.4.1. SHTTP-Cryptopts 26
4. New Header Lines for HTTP 26
4.1. Security-Scheme 26
5. (Retriable) Server Status Error Reports 27
5.1. Retry for Option (Re)Negotiation 27
5.2. Specific Retry Behavior 28
5.3. Limitations On Automatic Retries 29
6. Other Issues 30
6.1. Compatibility of Servers with Old Clients 30
6.2. URL Protocol Type 30
6.3. Browser Presentation 31
7. Implementation Notes 32
7.1. Preenhanced Data 32
7.2. Note:Proxy Interaction 34
7.2.1. Client-Proxy Authentication 34
8. Implementation Recommendations and Requirements 34
9. Protocol Syntax Summary 35
10. An Extended Example 36
Appendix: A Review of CMS 40
Bibliography and References 41
Security Considerations 43
Authors' Addresses 44
Full Copyright Statement.....45

Microsoft Exhibit 1019

1. Introduction

The World Wide Web (WWW) is a distributed hypermedia system which has gained widespread acceptance among Internet users. Although WWW browsers support other, preexisting Internet application protocols, the native and primary protocol used between WWW clients and servers is the HyperText Transfer Protocol (HTTP) [RFC-2616]. The ease of use of the Web has prompted its widespread employment as a client/server architecture for many applications. Many such applications require the client and server to be able to authenticate each other and exchange sensitive information confidentially. The original HTTP specification had only modest support for the cryptographic mechanisms appropriate for such transactions.

Secure HTTP (S-HTTP) provides secure communication mechanisms between an HTTP client-server pair in order to enable spontaneous commercial transactions for a wide range of applications. Our design intent is to provide a flexible protocol that supports multiple orthogonal operation modes, key management mechanisms, trust models, cryptographic algorithms and encapsulation formats through option negotiation between parties for each transaction.

1.1. Summary of Features

Secure HTTP is a secure message-oriented communications protocol designed for use in conjunction with HTTP. It is designed to coexist with HTTP's messaging model and to be easily integrated with HTTP applications.

Secure HTTP provides a variety of security mechanisms to HTTP clients and servers, providing the security service options appropriate to the wide range of potential end uses possible for the World-Wide Web. The protocol provides symmetric capabilities to both client and server (in that equal treatment is given to both requests and replies, as well as for the preferences of both parties) while preserving the transaction model and implementation characteristics of HTTP.

Several cryptographic message format standards may be incorporated into S-HTTP clients and servers, particularly, but in principle not limited to, [CMS] and [MOSS]. S-HTTP supports interoperation among a variety of implementations, and is compatible with HTTP. S-HTTP aware clients can communicate with S-HTTP oblivious servers and vice-versa, although such transactions obviously would not use S-HTTP security features.

S-HTTP does not require client-side public key certificates (or public keys), as it supports symmetric key-only operation modes.

Microsoft Exhibit 1019

This is significant because it means that spontaneous private transactions can occur without requiring individual users to have an established public key. While S-HTTP is able to take advantage of ubiquitous certification infrastructures, its deployment does not require it.

S-HTTP supports end-to-end secure transactions, in contrast with the original HTTP authorization mechanisms which require the client to attempt access and be denied before the security mechanism is employed. Clients may be "primed" to initiate a secure transaction (typically using information supplied in message headers); this may be used to support encryption of fill-out forms, for example. With S-HTTP, no sensitive data need ever be sent over the network in the clear.

S-HTTP provides full flexibility of cryptographic algorithms, modes and parameters. Option negotiation is used to allow clients and servers to agree on transaction modes (e.g., should the request be signed or encrypted or both -- similarly for the reply?); cryptographic algorithms (RSA vs. DSA for signing, DES vs. RC2 for encrypting, etc.); and certificate selection (please sign with your "Block-buster Video certificate").

S-HTTP attempts to avoid presuming a particular trust model, although its designers admit to a conscious effort to facilitate multiply-rooted hierarchical trust, and anticipate that principals may have many public key certificates.

S-HTTP differs from Digest-Authentication, described in [RFC-2617] in that it provides support for public key cryptography and consequently digital signature capability, as well as providing confidentiality.

1.2. Changes

This document describes S-HTTP/1.4. It differs from the previous memo in that it differs from the previous memo in its support of the Cryptographic Message Syntax (CMS) [CMS], a successor to PKCS-7; and hence now supports the Diffie-Hellman and the (NIST) Digital Signature Standard cryptosystems. CMS used in RSA mode is bits on the wire compatible with PKCS-7.

Microsoft Exhibit 1019

1.3. Processing Model

1.3.1. Message Preparation

The creation of an S-HTTP message can be thought of as a a function with three inputs:

1. The cleartext message. This is either an HTTP message or some other data object. Note that since the cleartext message is carried transparently, headers and all, any version of HTTP can be carried within an S-HTTP wrapper.
2. The receiver's cryptographic preferences and keying material. This is either explicitly specified by the receiver or subject to some default set of preferences.
3. The sender's cryptographic preferences and keying material. This input to the function can be thought of as implicit since it exists only in the memory of the sender.

In order to create an S-HTTP message, then, the sender integrates the sender's preferences with the receiver's preferences. The result of this is a list of cryptographic enhancements to be applied and keying material to be used to apply them. This may require some user intervention. For instance, there might be multiple keys available to sign the message. (See Section 3.2.4.9.3 for more on this topic.) Using this data, the sender applies the enhancements to the message clear-text to create the S-HTTP message.

The processing steps required to transform the cleartext message into the S-HTTP message are described in Sections 2 and 3. The processing steps required to merge the sender's and receiver's preferences are described in Sections 3.2.

1.3.2. Message Recovery

The recovery of an S-HTTP message can be thought of as a function of four distinct inputs:

1. The S-HTTP message.
2. The receiver's stated cryptographic preferences and keying material. The receiver has the opportunity to remember what cryptographic preferences it provided in order for this document to be dereferenced.
3. The receiver's current cryptographic preferences and keying material.
4. The sender's previously stated cryptographic options. The sender may have stated that he would perform certain cryptographic operations in this message. (Again, see sections 4 and 5 for details on how to do this.)

Microsoft Exhibit 1019

In order to recover an S-HTTP message, the receiver needs to read the headers to discover which cryptographic transformations were performed on the message, then remove the transformations using some combination of the sender's and receiver's keying material, while taking note of which enhancements were applied.

The receiver may also choose to verify that the applied enhancements match both the enhancements that the sender said he would apply (input 4 above) and that the receiver requested (input 2 above) as well as the current preferences to see if the S-HTTP message was appropriately transformed. This process may require interaction with the user to verify that the enhancements are acceptable to the user. (See Section 6.4 for more on this topic.)

1.4. Modes of Operation

Message protection may be provided on three orthogonal axes: signature, authentication, and encryption. Any message may be signed, authenticated, encrypted, or any combination of these (including no protection).

Multiple key management mechanisms are supported, including password-style manually shared secrets and public-key key exchange. In particular, provision has been made for prearranged (in an earlier transaction or out of band) symmetric session keys in order to send confidential messages to those who have no public key pair.

Additionally, a challenge-response ("nonce") mechanism is provided to allow parties to assure themselves of transaction freshness.

1.4.1. Signature

If the digital signature enhancement is applied, an appropriate certificate may either be attached to the message (possibly along with a certificate chain) or the sender may expect the recipient to obtain the required certificate (chain) independently.

1.4.2. Key Exchange and Encryption

In support of bulk encryption, S-HTTP defines two key transfer mechanisms, one using public-key enveloped key exchange and another with externally arranged keys.

In the former case, the symmetric-key cryptosystem parameter is passed encrypted under the receiver's public key.

Microsoft Exhibit 1019

In the latter mode, we encrypt the content using a prearranged session key, with key identification information specified on one of the header lines.

1.4.3. Message Integrity and Sender Authentication

Secure HTTP provides a means to verify message integrity and sender authenticity for a message via the computation of a Message Authentication Code (MAC), computed as a keyed hash over the document using a shared secret -- which could potentially have been arranged in a number of ways, e.g.: manual arrangement or 'inband' key management. This technique requires neither the use of public key cryptography nor encryption.

This mechanism is also useful for cases where it is appropriate to allow parties to identify each other reliably in a transaction without providing (third-party) non-repudiability for the transactions themselves. The provision of this mechanism is motivated by our bias that the action of "signing" a transaction should be explicit and conscious for the user, whereas many authentication needs (i.e., access control) can be met with a lighter-weight mechanism that retains the scalability advantages of public-key cryptography for key exchange.

1.4.4. Freshness

The protocol provides a simple challenge-response mechanism, allowing both parties to insure the freshness of transmissions. Additionally, the integrity protection provided to HTTP headers permits implementations to consider the Date: header allowable in HTTP messages as a freshness indicator, where appropriate (although this requires implementations to make allowances for maximum clock skew between parties, which we choose not to specify).

1.5. Implementation Options

In order to encourage widespread adoption of secure documents for the World-Wide Web in the face of the broad scope of application requirements, variability of user sophistication, and disparate implementation constraints, Secure HTTP deliberately caters to a variety of implementation options. See Section 8 for implementation recommendations and requirements.

2. Message Format

Syntactically, Secure HTTP messages are the same as HTTP, consisting of a request or status line followed by headers and a body. However, the range of headers is different and the bodies are typically

Microsoft Exhibit 1019

cryptographically enhanced.

2.1. Notational Conventions

This document uses the augmented BNF from HTTP [RFC-2616]. You should refer to that document for a description of the syntax.

2.2. Request Line

In order to differentiate S-HTTP messages from HTTP messages and allow for special processing, the request line should use the special Secure" method and use the protocol designator "Secure-HTTP/1.4". Consequently, Secure-HTTP and HTTP processing can be intermixed on the same TCP port, e.g. port 80. In order to prevent leakage of potentially sensitive information Request-URI should be "*". For example:

```
Secure * Secure-HTTP/1.4
```

When communicating via a proxy, the Request-URI should be consist of the AbsoluteURI. Typically, the rel path section should be replaced by "*" to minimize the information passed to in the clear. (e.g. http://www.terisa.com/*); proxies should remove the appropriate amount of this information to minimize the threat of traffic analysis. See Section 7.2.2.1 for a situation where providing more information is appropriate.

2.3. The Status Line

S-HTTP responses should use the protocol designator "Secure-HTTP/1.4". For example:

```
Secure-HTTP/1.4 200 OK
```

Note that the status in the Secure HTTP response line does not indicate anything about the success or failure of the unwrapped HTTP request. Servers should always use 200 OK provided that the Secure HTTP processing is successful. This prevents analysis of success or failure for any request, which the correct recipient can determine from the encapsulated data. All case variations should be accepted.

2.4. Secure HTTP Header Lines

The header lines described in this section go in the header of a Secure HTTP message. All except ’Content-Type’ and ’Content-Privacy-Domain’ are optional. The message body shall be separated from the header block by two successive CRLFs.

Microsoft Exhibit 1019

All data and fields in header lines should be treated as case insensitive unless otherwise specified. Linear whitespace [RFC-822] should be used only as a token separator unless otherwise quoted. Long header lines may be line folded in the style of [RFC-822].

This document refers to the header block following the S-HTTP request/response line and preceding the successive CRLFs collectively as "S-HTTP headers".

2.4.1. Content-Privacy-Domain

The two values defined by this document are 'MOSS' and 'CMS'. CMS refers to the privacy enhancement specified in section 2.6.1. MOSS refers to the format defined in [RFC-1847] and [RFC-1848].

2.4.2. Content-Type for CMS

Under normal conditions, the terminal encapsulated content (after all privacy enhancements have been removed) would be an HTTP message. In this case, there shall be a Content-Type line reading:

```
Content-Type: message/http
```

The message/http content type is defined in RFC-2616.

If the inner message is an S-HTTP message, then the content type shall be 'application/s-http'. (See Appendix for the definition of this.)

It is intended that these types be registered with IANA as MIME content types.

The terminal content may be of some other type provided that the type is properly indicated by the use of an appropriate Content-Type header line. In this case, the header fields for the encapsulation of the terminal content apply to the terminal content (the 'final headers'). But in any case, final headers should themselves always be S-HTTP encapsulated, so that the applicable S-HTTP/HTTP headers are never passed unenhanced.

S-HTTP encapsulation of non-HTTP data is a useful mechanism for passing pre-enhanced data (especially presigned data) without requiring that the HTTP headers themselves be pre-enhanced.

Microsoft Exhibit 1019

2.4.3. Content-Type for MOSS

The Content-Type for MOSS shall be an acceptable MIME content type describing the cryptographic processing applied. (e.g. multipart/signed). The content type of the inner content is described in the content type line corresponding to that inner content, and for HTTP messages shall be 'message/http'.

2.4.4. Prearranged-Key-Info

This header line is intended to convey information about a key which has been arranged outside of the internal cryptographic format. One use of this is to permit in-band communication of session keys for return encryption in the case where one of the parties does not have a key pair. However, this should also be useful in the event that the parties choose to use some other mechanism, for instance, a one-time key list.

This specification defines two methods for exchanging named keys, Inband, Outband. Inband indicates that the session key was exchanged previously, using a Key-Assign header of the corresponding method. Outband arrangements imply that agents have external access to key materials corresponding to a given name, presumably via database access or perhaps supplied immediately by a user from keyboard input. The syntax for the header line is:

```
Prearranged-Key-Info =
 "Prearranged-Key-Info" ":" Hdr-Cipher "," CoveredDEK "," CoverKey-ID
CoverKey-ID = method ":" key-name
CoveredDEK = *HEX
method = "inband" |  "outband"
```

While chaining ciphers require an Initialization Vector (IV) [FIPS-81] to start off the chaining, that information is not carried by this field. Rather, it should be passed internal to the cryptographic format being used. Likewise, the bulk cipher used is specified in this fashion.

<Hdr-Cipher> should be the name of the block cipher used to encrypt the session key (see section 3.2.4.7)

<CoveredDEK> is the protected Data Encryption Key (a.k.a. transaction key) under which the encapsulated message was encrypted. It should be appropriately (randomly) generated by the sending agent, then encrypted under the cover of the negotiated key (a.k.a. session key) using the indicated header cipher, and then converted into hex.

Microsoft Exhibit 1019

In order to avoid name collisions, cover key namespaces must be maintained separately by host and port.

Note that some Content-Privacy-Domains, notably likely future revisions of MOSS and CMS may have support for symmetric key management.

The Prearranged-Key-Info field need not be used in such circumstances. Rather, the native syntax is preferred. Keys exchanged with Key-Assign, however, may be used in this situation.

2.4.5. MAC-Info

This header is used to supply a Message Authenticity Check, providing both message authentication and integrity, computed from the message text, the time (optional -- to prevent replay attack), and a shared secret between client and server. The MAC should be computed over the encapsulated content of the S-HTTP message. S-HTTP/1.1 defined that MACs should be computed using the following algorithm ('||' means concatenation):

```
MAC = hex(H(Message||[<time>]||<shared key>))
```

The time should be represented as an unsigned 32 bit quantity representing seconds since 00:00:00 GMT January 1, 1970 (the UNIX epoch), in network byte order. The shared key format is a local matter.

Recent research [VANO95] has demonstrated some weaknesses in this approach, and this memo introduces a new construction, derived from [RFC-2104]. In the name of backwards compatibility, we retain the previous constructions with the same names as before. However, we also introduce a new series of names (See Section 3.2.4.8 for the names) that obey a different (hopefully stronger) construction. (^ means bitwise XOR)

```
HMAC = hex(H(K' ^ pad2 || H(K' ^ pad1 ||[<time>]|| Message)))
pad1 = the byte 0x36 repeated enough times to fill out a
            hash input block. (I.e. 64 times for both MD5 and SHA-1)
pad2 = the byte 0x5c repeated enough times to fill out a
            hash input block.
K' = H(<shared key>)
```

The original HMAC construction is for the use of a key with length equal to the length of the hash output. Although it is considered safe to use a key of a different length (Note that strength cannot be increased past the length of the hash function itself, but can be reduced by using a shorter key.) [KRAW96b] we hash the original key

Microsoft Exhibit 1019

to permit the use of shared keys (e.g. passphrases) longer than the length of the hash. It is noteworthy (though obvious) that this technique does not increase the strength of short keys.

The format of the MAC-Info line is:

```
MAC-Info =
"MAC-Info" ":"  [hex-time],
hash-alg, hex-hash-data, key-spec
hex-time = <unsigned seconds since Unix epoch represented as HEX>
hash-alg = <hash algorithms from section 3.2.4.8>
hex-hash-data = <computation as described above represented as HEX>
Key-Spec = "null" | "dek" | Key-ID
```

Key-Ids can refer either to keys bound using the Key-Assign header line or those bound in the same fashion as the Outband method described later. The use of a 'Null' key-spec implies that a zero length key was used, and therefore that the MAC merely represents a hash of the message text and (optionally) the time. The special key-spec 'DEK' refers to the Data Exchange Key used to encrypt the following message body (it is an error to use the DEK key-spec in situations where the following message body is unencrypted).

If the time is omitted from the MAC-Info line, it should simply not be included in the hash.

Note that this header line can be used to provide a more advanced equivalent of the original HTTP Basic authentication mode in that the user can be asked to provide a username and password. However, the password remains private and message integrity can be assured. Moreover, this can be accomplished without encryption of any kind.

In addition, MAC-Info permits fast message integrity verification (at the loss of non-repudiability) for messages, provided that the participants share a key (possibly passed using Key-Assign in a previous message).

Note that some Content-Privacy-Domains, notably likely future revisions of MOSS and CMS may have support for symmetric integrity protection The MAC-Info field need not be used in such circumstances. Rather, the native syntax is preferred. Keys exchanged with Key-Assign, however, may be used in this situation.

2.5. Content

The content of the message is largely dependent upon the values of the Content-Privacy-Domain and Content-Transfer-Encoding fields.

Microsoft Exhibit 1019

For a CMS message, with '8BIT' Content-Transfer-Encoding, the content should simply be the CMS message itself.

If the Content-Privacy-Domain is MOSS, the content should consist of a MOSS Security Multipart as described in RFC1847.

It is expected that once the privacy enhancements have been removed, the resulting (possibly protected) contents will be a normal HTTP request. Alternately, the content may be another Secure-HTTP message, in which case privacy enhancements should be unwrapped until clear content is obtained or privacy enhancements can no longer be removed. (This permits embedding of enhancements, such as sequential Signed and Enveloped enhancements.) Provided that all enhancements can be removed, the final de-enhanced content should be a valid HTTP request (or response) unless otherwise specified by the Content-Type line.

Note that this recursive encapsulation of messages potentially permits security enhancements to be applied (or removed) for the benefit of intermediaries who may be a party to the transaction between a client and server (e.g., a proxy requiring client authentication). How such intermediaries should indicate such processing is described in Section 7.2.1.

2.6. Encapsulation Format Options

2.6.1. Content-Privacy-Domain: CMS

Content-Privacy-Domain 'CMS' follows the form of the CMS standard (see Appendix).

Message protection may proceed on two orthogonal axes: signature and encryption. Any message may be either signed, encrypted, both, or neither. Note that the 'auth' protection mode of S-HTTP is provided independently of CMS coding via the MAC-Info header of section 2.3.6 since CMS does not support a 'KeyDigestedData' type, although it does support a 'DigestedData' type.

2.6.1.1. Signature

This enhancement uses the 'SignedData' type of CMS. When digital signatures are used, an appropriate certificate may either be attached to the message (possibly along with a certificate chain) as specified in CMS or the sender may expect the recipient to obtain its certificate (and/or chain) independently. Note that an explicitly allowed instance of this is a certificate signed with the private component corresponding to the public component being attested to. This shall be referred to as a self-signed certificate. What, if any, weight to give to such a certificate is a purely local matter. In

Microsoft Exhibit 1019

either case, a purely signed message is precisely CMS compliant.

2.6.1.2. Encryption

2.6.1.2.1. Encryption -- normal, public key

This enhancement is performed precisely as enveloping (using either ’ EnvelopedData’ types) under CMS. A message encrypted in this fashion, signed or otherwise, is CMS compliant. To have a message which is both signed and encrypted, one simply creates the CMS SignedData production and encapsulates it in EnvelopedData as described in CMS.

2.6.1.2.2. Encryption -- prearranged key

This uses the ’EncryptedData’ type of CMS. In this mode, we encrypt the content using a DEK encrypted under cover of a prearranged session key (how this key may be exchanged is discussed later), with key identification information specified on one of the header lines. The IV is in the EncryptedContentInfo type of the EncryptedData element. To have a message which is both signed and encrypted, one simply creates the CMS SignedData production and encapsulates it in EncryptedData as described in CMS.

2.6.2. Content-Privacy-Domain: MOSS

The body of the message should be a MIME compliant message with content type that matches the Content-Type line in the S-HTTP headers. Encrypted messages should use multipart/encrypted. Signed messages should use multipart/signed. However, since multipart/signed does not convey keying material, is is acceptable to use multipart/mixed where the first part is application/mosskey-data and the second part is multipart/mixed in order to convey certificates for use in verifying the signature.

Implementation Note: When both encryption and signature are applied by the same agent, signature should in general be applied before encryption.

2.6.3. Permitted HTTP headers

2.6.3.1. Overview

In general, HTTP [RFC-2616] headers should appear in the inner content (i.e. the message/http) of an S-HTTP message but should not appear in the S-HTTP message wrapper for security reasons. However, certain headers need to be visible to agents which do not have access to the encapsulated data. These headers may appear in the S-HTTP headers as well.

Microsoft Exhibit 1019

Please note that although brief descriptions of the general purposes of these headers are provided for clarity, the definitive reference is [RFC-2616].

2.6.3.2. Host

The host header specificies the internet host and port number of the resource being requested. This header should be used to disambiguate among multiple potential security contexts within which this message could be interpreted. Note that the unwrapped HTTP message will have it's own Host field (assuming it's an HTTP/1.1 message). If these fields do not match, the server should respond with a 400 status code.

2.6.3.3. Connection

The Connection field has precisely the same semantics in S-HTTP headers as it does in HTTP headers. This permits persistent connections to be used with S-HTTP.

3. Cryptographic Parameters

3.1. Options Headers

As described in Section 1.3.2, every S-HTTP request is (at least conceptually) preconditioned by the negotiation options provided by the potential receiver. The two primary locations for these options are

1. In the headers of an HTTP Request/Response.
2. In the HTML which contains the anchor being dereferenced.

There are two kinds of cryptographic options which may be provided: Negotiation options, as discussed in Section 3.2 convey a potential message recipient's cryptographic preferences. Keying options, as discussed in Section 3.3 provide keying material (or pointers to keying material) which may be of use to the sender when enhancing a message.

Binding cryptographic options to anchors using HTML extensions is the topic of the companion document [SHTML] and will not be treated here.

Microsoft Exhibit 1019

3.2. Negotiation Options

3.2.1. Negotiation Overview

Both parties are able to express their requirements and preferences regarding what cryptographic enhancements they will permit/require the other party to provide. The appropriate option choices depend on implementation capabilities and the requirements of particular applications.

A negotiation header is a sequence of specifications each conforming to a four-part schema detailing:

Property -- the option being negotiated, such as bulk encryption algorithm.

Value -- the value being discussed for the property, such as DES-CBC

Direction -- the direction which is to be affected, namely: during reception or origination (from the perspective of the originator).

Strength -- strength of preference, namely: required, optional, refused

As an example, the header line:

```
SHTTP-Symmetric-Content-Algorithms: recv-optional=DES-CBC,RC2
```

could be thought to say: "You are free to use DES-CBC or RC2 for bulk encryption for encrypting messages to me."

We define new headers (to be used in the encapsulated HTTP header, not in the S-HTTP header) to permit negotiation of these matters.

3.2.2. Negotiation Option Format

The general format for negotiation options is:

```
Option = Field ":" Key-val ";" *(Key-val)
Key-val = Key "=" Value *("," Value)
Key = Mode"-"Action             ; This is represented as one
                                ; token without whitespace
Mode = "orig" | "recv"
Action = "optional" | "required" | "refused"
```


Microsoft Exhibit 1019

The <Mode> value indicates whether this <Key-val> refers to what the agent's actions are upon sending privacy enhanced messages as opposed to upon receiving them. For any given mode-action pair, the interpretation to be placed on the enhancements (<Value>s) listed is:

'recv-optional:' The agent will process the enhancement if the other party uses it, but will also gladly process messages without the enhancement.

'recv-required:' The agent will not process messages without this enhancement.

'recv-refused:' The agent will not process messages with this enhancement.

'orig-optional:' When encountering an agent which refuses this enhancement, the agent will not provide it, and when encountering an agent which requires it, this agent will provide it.

'orig-required:' The agent will always generate the enhancement.

'orig-refused:' The agent will never generate the enhancement.

The behavior of agents which discover that they are communicating with an incompatible agent is at the discretion of the agents. It is inappropriate to blindly persist in a behavior that is known to be unacceptable to the other party. Plausible responses include simply terminating the connection, or, in the case of a server response, returning 'Not implemented 501'.

Optional values are considered to be listed in decreasing order of preference. Agents are free to choose any member of the intersection of the optional lists (or none) however.

If any <Key-Val> is left undefined, it should be assumed to be set to the default. Any key which is specified by an agent shall override any appearance of that key in any <Key-Val> in the default for that field.

Microsoft Exhibit 1019

3.2.3. Parametrization for Variable-length Key Ciphers

For ciphers with variable key lengths, values may be parametrized using the syntax <cipher>'['<length>']'

For example, 'RSA[1024]' represents a 1024 bit key for RSA. Ranges may be represented as

<cipher>'['<bound1>'-'<bound2>']'

For purposes of preferences, this notation should be treated as if it read (assuming x and y are integers)

<cipher>[x], <cipher>[x+1],...<cipher>[y] (if x<y)

and

<cipher>[x], <cipher>[x-1],...<cipher>[y] (if x>y)

The special value 'inf' may be used to denote infinite length.

Using simply <cipher> for such a cipher shall be read as the maximum range possible with the given cipher.

3.2.4. Negotiation Syntax

3.2.4.1. SHTTP-Privacy-Domains

This header refers to the Content-Privacy-Domain type of section 2.3.1. Acceptable values are as listed there. For instance,

```
SHTTP-Privacy-Domains: orig-required=cms;
                       recv-optional=cms,MOSS
```

would indicate that the agent always generates CMS compliant messages, but can read CMS or MOSS (or, unenhanced messages).

3.2.4.2. SHTTP-Certificate-Types

This indicates what sort of Public Key certificates the agent will accept. Currently defined values are 'X.509' and 'X.509v3'.

3.2.4.3. SHTTP-Key-Exchange-Algorithms

This header indicates which algorithms may be used for key exchange. Defined values are 'DH', 'RSA', 'Outband' and 'Inband'. DH refers to Diffie-Hellman X9.42 style enveloping. [DH] RSA refers to RSA enveloping. Outband refers to some sort of external key agreement.

Microsoft Exhibit 1019

Inband refers to section 3.3.3.1.

The expected common configuration of clients having no certificates and servers having certificates would look like this (in a message sent by the server):

```
SHTTP-Key-Exchange-Algorithms: orig-optional=Inband, DH;
                                    recv-required=DH
```

3.2.4.4. SHTTP-Signature-Algorithms

This header indicates what Digital Signature algorithms may be used. Defined values are 'RSA' [PKCS-1] and 'NIST-DSS' [FIPS-186] Since NIST-DSS and RSA use variable length moduli the parametrization syntax of section 3.2.3 should be used. Note that a key length specification may interact with the acceptability of a given certificate, since keys (and their lengths) are specified in public-key certificates.

3.2.4.5. SHTTP-Message-Digest-Algorithms

This indicates what message digest algorithms may be used. Previously defined values are 'RSA-MD2' [RFC-1319], 'RSA-MD5' [RFC-1321], 'NIST-SHS' [FIPS-180].

3.2.4.6. SHTTP-Symmetric-Content-Algorithms

This header specifies the symmetric-key bulk cipher used to encrypt message content. Defined values are:

```
DES-CBC -- DES in Cipher Block Chaining (CBC) mode [FIPS-81]
DES-EDE-CBC -- 2 Key 3DES using Encrypt-Decrypt-Encrypt in outer
               CBC mode
DES-EDE3-CBC -- 3 Key 3DES using Encrypt-Decrypt-Encrypt in outer
                CBC mode
DESX-CBC -- RSA's DESX in CBC mode
IDEA-CBC -- IDEA in CBC mode
RC2-CBC -- RSA's RC2 in CBC mode
CDMF-CBC -- IBM's CDMF (weakened key DES) [JOHN93] in CBC mode
```

Since RC2 keys are variable length, the syntax of section 3.2.3 should be used.

Microsoft Exhibit 1019

3.2.4.7. SHTTP-Symmetric-Header-Algorithms

This header specifies the symmetric-key cipher used to encrypt message headers.

DES-ECB -- DES in Electronic Codebook (ECB) mode [FIPS-81]
DES-EDE-ECB -- 2 Key 3DES using Encrypt-Decrypt-Encrypt in ECB mode
DES-EDE3-ECB -- 3 Key 3DES using Encrypt-Decrypt-Encrypt in ECB mode
DESX-ECB -- RSA's DESX in ECB mode
IDEA-ECB -- IDEA
RC2-ECB -- RSA's RC2 in ECB mode
CDMF-ECB -- IBM's CDMF in ECB mode

Since RC2 is variable length, the syntax of section 3.2.3 should be used.

3.2.4.8. SHTTP-MAC-Algorithms

This header indicates what algorithms are acceptable for use in providing a symmetric key MAC. 'RSA-MD2', 'RSA-MD5' and 'NIST-SHS' persist from S-HTTP/1.1 using the old MAC construction. The tokens 'RSA-MD2-HMAC', 'RSA-MD5-HMAC' and 'NIST-SHS-HMAC' indicate the new HMAC construction of 2.3.6 with the MD2, MD5, and SHA-1 algorithms respectively.

3.2.4.9. SHTTP-Privacy-Enhancements

This header indicates security enhancements to apply. Possible values are 'sign', 'encrypt' and 'auth' indicating whether messages are signed, encrypted, or authenticated (i.e., provided with a MAC), respectively.

3.2.4.10. Your-Key-Pattern

This is a generalized pattern match syntax to describe identifiers for a large number of types of keying material. The general syntax is:

```
Your-Key-Pattern =
        "Your-Key-Pattern" ":" key-use "," pattern-info
key-use = "cover-key" | "auth-key" | "signing-key"
```


Microsoft Exhibit 1019

3.2.4.10.1. Cover Key Patterns

This header specifies desired values for key names used for encryption of transaction keys using the Prearranged-Key-Info syntax of section 2.3.5. The pattern-info syntax consists of a series of comma separated regular expressions. Commas should be escaped with backslashes if they appear in the regexps. The first pattern should be assumed to be the most preferred.

3.2.4.10.2. Auth key patterns

Auth-key patterns specify name forms desired for use for MAC authenticators. The pattern-info syntax consists of a series of comma separated regular expressions. Commas should be escaped with backslashes if they appear in the regexps. The first pattern should be assumed to be the most preferred.

3.2.4.10.3. Signing Key Pattern

This parameter describes a pattern or patterns for what keys are acceptable for signing for the digital signature enhancement. The pattern-info syntax for signing-key is:

pattern-info = name-domain "," pattern-data

The only currently defined name-domain is 'DN-1779'. This parameter specifies desired values for fields of Distinguished Names. DNs are considered to be represented as specified in RFC1779, the order of fields and whitespace between fields is not significant.

All RFC1779 values should use ',' as a separator rather than ';', since ';' is used as a statement separator in S-HTTP.

Pattern-data is a modified RFC1779 string, with regular expressions permitted as field values. Pattern match is performed field-wise, unspecified fields match any value (and therefore leaving the DN-Pattern entirely unspecified allows for any DN). Certificate chains may be matched as well (to allow for certificates without name subordination). DN chains are considered to be ordered left-to-right with the issuer of a given certificate on its immediate right, although issuers need not be specified. A trailing '.' indicates that the sequence of DNs is absolute. I.e. that the one furthest to the right is a root.

Microsoft Exhibit 1019

The syntax for the pattern values is,

```
Value = DN-spec *("," Dn-spec)["."]
Dn-spec = "/" *(Field-spec) "/"
Field-spec := Attr = "Pattern"
Attr = "CN" | "L" | "ST" | "O" |
           "OU" | "C" | <or as appropriate>
Pattern = <POSIX 1003.2 regular expressions>
```

For example, to request that the other agent sign with a key certified by the RSA Persona CA (which uses name subordination) one could use the expression below. Note the use of RFC1779 quoting to protect the comma (an RFC1779 field separator) and the POSIX 1003.2 quoting to protect the dot (a regular expression metacharacter).

```
   Your-Key-Pattern: signing-key, DN-1779,
               /OU=Persona Certificate, O="RSA Data Security,
Inc\."/
```

3.2.4.11. Example

A representative header block for a server follows.

```
SHTTP-Privacy-Domains: recv-optional=MOSS, CMS;
      orig-required=CMS
SHTTP-Certificate-Types: recv-optional=X.509;
      orig-required=X.509
SHTTP-Key-Exchange-Algorithms: recv-required=DH;
      orig-optional=Inband,DH
SHTTP-Signature-Algorithms: orig-required=NIST-DSS;
      recv-required=NIST-DSS
SHTTP-Privacy-Enhancements: orig-required=sign;
      orig-optional=encrypt
```

3.2.4.12. Defaults

Explicit negotiation parameters take precedence over default values. For a given negotiation option type, defaults for a given mode-action pair (such as 'orig-required') are implicitly merged unless explicitly overridden.

The default values (these may be negotiated downward or upward) are:

```
SHTTP-Privacy-Domains: orig-optional=CMS;
                       recv-optional=CMS
SHTTP-Certificate-Types: orig-optional=X.509;
                         recv-optional=X.509
SHTTP-Key-Exchange-Algorithms: orig-optional=DH,Inband,Outband;
```


Microsoft Exhibit 1019

```
                                    recv-optional=DH,Inband,Outband
      SHTTP-Signature-Algorithms: orig-optional=NIST-DSS;
                                  recv-optional=NIST-DSS
      SHTTP-Message-Digest-Algorithms: orig-optional=RSA-MD5;
                                    recv-optional=RSA-MD5
      SHTTP-Symmetric-Content-Algorithms: orig-optional=DES-CBC;
                                    recv-optional=DES-CBC
      SHTTP-Symmetric-Header-Algorithms: orig-optional=DES-ECB;
                                   recv-optional=DES-ECB
      SHTTP-Privacy-Enhancements: orig-optional=sign,encrypt, auth;
                                    recv-required=encrypt;
                                    recv-optional=sign, auth
```

3.3. Non-Negotiation Headers

There are a number of options that are used to communicate or identify the potential recipient's keying material.

3.3.1. Encryption-Identity

This header identifies a potential principal for whom the message described by these options could be encrypted; Note that this explicitly permits return encryption under (say) public key without the other agent signing first (or under a different key than that of the signature). The syntax of the Encryption-Identity line is:

```
Encryption-Identity =
        "Encryption Identity" ":" name-class,key-sel,name-arg
name-class = "DN-1779" | MOSS name forms
```

The name-class is an ASCII string representing the domain within which the name is to be interpreted, in the spirit of MOSS. In addition to the MOSS name forms of RFC1848, we add the DN-1779 name form to represent a more convenient form of distinguished name.

3.3.1.1. DN-1779 Name Class

The argument is an RFC-1779 encoded DN.

3.3.2. Certificate-Info

In order to permit public key operations on DNs specified by Encryption-Identity headers without explicit certificate fetches by the receiver, the sender may include certification information in the Certificate-Info option. The format of this option is:

```
Certificate-Info: <Cert-Fmt>',<Cert-Group>
```

<Cert-Fmt> should be the type of <Cert-Group> being presented.

Microsoft Exhibit 1019

Defined values are 'PEM' and 'CMS'. CMS certificate groups are provided as a base-64 encoded CMS SignedData message containing sequences of certificates with or without the SignerInfo field. A PEM format certificate group is a list of comma-separated base64-encoded PEM certificates.

Multiple Certificate-Info lines may be defined.

3.3.3. Key-Assign

This option serves to indicate that the agent wishes to bind a key to a symbolic name for (presumably) later reference.

The general syntax of the key-assign header is:

```
Key-Assign =
        "Key-Assign" ":" Method "," Key-Name ","
        Lifetime "," Ciphers ";" Method-args

Key-name = string
Lifetime = "this" | "reply" | ""
Method ="inband"
Ciphers = "null" | Cipher+
Cipher" = <Header cipher from section 3.2.4.7>
kv = "4" | "5"
```

Key-Name is the symbolic name to which this key is to be bound. Ciphers is a list of ciphers for which this key is potentially applicable (see the list of header ciphers in section 3.2.4.7). The keyword 'null' should be used to indicate that it is inappropriate for use with ANY cipher. This is potentially useful for exchanging keys for MAC computation.

Lifetime is a representation of the longest period of time during which the recipient of this message can expect the sender to accept that key. 'this' indicates that it is likely to be valid only for reading this transmission. 'reply' indicates that it is useful for a reply to this message. If a Key-Assign with the reply lifetime appears in a CRYPTOPTS block, it indicates that it is good for at least one (but perhaps only one) dereference of this anchor. An unspecified lifetime implies that this key may be reused for an indefinite number of transactions.

Method should be one of a number of key exchange methods. The only currently defined value is 'inband' referring to Inband keys (i.e., direct assignment).

Microsoft Exhibit 1019

This header line may appear either in an unencapsulated header or in an encapsulated message, though when an uncovered key is being directly assigned, it may only appear in an encrypted encapsulated content. Assigning to a key that already exists causes that key to be overwritten.

Keys defined by this header are referred to elsewhere in this specification as Key-IDs, which have the syntax:

```
Key-ID = method ":" key-name
```

3.3.3.1. Inband Key Assignment

This refers to the direct assignment of an uncovered key to a symbolic name. Method-args should be just the desired session key encoded in hexidecimal as in:

```
Key-Assign: inband,akey,reply,DES-ECB;0123456789abcdef
```

Short keys should be derived from long keys by reading bits from left to right.

Note that inband key assignment is especially important in order to permit confidential spontaneous communication between agents where one (but not both) of the agents have key pairs. However, this mechanism is also useful to permit key changes without public key computations. The key information is carried in this header line must be in the inner secured HTTP request, therefore use in unencrypted messages is not permitted.

3.3.4. Nonces

Nonces are opaque, transient, session-oriented identifiers which may be used to provide demonstrations of freshness. Nonce values are a local matter, although they are might well be simply random numbers generated by the originator. The value is supplied simply to be returned by the recipient.

3.3.4.1. Nonce

This header is used by an originator to specify what value is to be returned in the reply. The field may be any value. Multiple nonces may be supplied, each to be echoed independently.

The Nonce should be returned in a Nonce-Echo header line. See section 4.1.1.

Microsoft Exhibit 1019

3.4. Grouping Headers With SHTTP-Cryptopts

In order for servers to bind a group of headers to an HTML anchor, it is possible to combine a number of headers on a single S-HTTP Cryptopts header line. The names of the anchors to which these headers apply is indicated with a 'scope' parameter.

3.4.1. SHTTP-Cryptopts

This option provides a set of cryptopts and a list of references to which it applies. (For HTML, these references would be named using the NAME tag). The names are provided in the scope attribute as a comma separated list and separated from the next header line by a semicolon. The format for the SHTTP-Cryptopts line is:

```
SHTTP-Cryptopts =
                  "SHTTP-Cryptopts" ":" scope ";" cryptopt-list
scope = "scope="<tag-spec>    ; This is all one token without whitespace
tag-spec = tag *("," tag) | ""
cryptopt-list = cryptopt *(";" cryptopt)
cryptopt = <S-HTTP cryptopt lines described below>
tag = <value used in HTML anchor NAME attribute>
```

For example:

```
SHTTP-Cryptopts: scope=tag1,tag2;
                  SHTTP-Privacy-Domains:
                  orig-required=cms; recv-optional=cms,MOSS
```

If a message contains both S-HTTP negotiation headers and headers grouped on SHTTP-Cryptopts line(s), the other headers shall be taken to apply to all anchors not bound on the SHTTP-Cryptopts line(s). Note that this is an all-or-nothing proposition. That is, if a SHTTP-Cryptopts header binds options to a reference, then none of these global options apply, even if some of the options headers do not appear in the bound options. Rather, the S-HTTP defaults found in Section 3.2.4.11 apply.

4. New Header Lines for HTTP

Two non-negotiation header lines for HTTP are defined here.

4.1. Security-Scheme

All S-HTTP compliant agents must generate the Security-Scheme header in the headers of all HTTP messages they generate. This header permits other agents to detect that they are communicating with an S-HTTP compliant agent and generate the appropriate cryptographic

Microsoft Exhibit 1019

options headers.

For implementations compliant with this specification, the value must be 'S-HTTP/1.4'.

4.1.1. Nonce-Echo

The header is used to return the value provided in a previously received Nonce: field. This has to go in the encapsulated headers so that it an be cryptographically protected.

5. (Retriable) Server Status Error Reports

We describe here the special processing appropriate for client retries in the face of servers returning an error status.

5.1. Retry for Option (Re)Negotiation

A server may respond to a client request with an error code that indicates that the request has not completely failed but rather that the client may possibly achieve satisfaction through another request. HTTP already has this concept with the 3XX redirection codes.

In the case of S-HTTP, it is conceivable (and indeed likely) that the server expects the client to retry his request using another set of cryptographic options. E.g., the document which contains the anchor that the client is dereferencing is old and did not require digital signature for the request in question, but the server now has a policy requiring signature for dereferencing this URL. These options should be carried in the header of the encapsulated HTTP message, precisely as client options are carried.

The general idea is that the client will perform the retry in the manner indicated by the combination of the original request and the precise nature of the error and the cryptographic enhancements depending on the options carried in the server response.

The guiding principle in client response to these errors should be to provide the user with the same sort of informed choice with regard to dereference of these anchors as with normal anchor dereference. For instance, in the case above, it would be inappropriate for the client to sign the request without requesting permission for the action.

Microsoft Exhibit 1019

5.2. Specific Retry Behavior

5.2.1. Unauthorized 401, PaymentRequired 402

The HTTP errors 'Unauthorized 401', 'PaymentRequired 402' represent failures of HTTP style authentication and payment schemes. While S-HTTP has no explicit support for these mechanisms, they can be performed under S-HTTP while taking advantage of the privacy services offered by S-HTTP. (There are other errors for S-HTTP specific authentication errors.)

5.2.2. 420 SecurityRetry

This server status reply is provided so that the server may inform the client that although the current request is rejected, a retried request with different cryptographic enhancements is worth attempting. This header shall also be used in the case where an HTTP request has been made but an S-HTTP request should have been made. Obviously, this serves no useful purpose other than signalling an error if the original request should have been encrypted, but in other situations (e.g. access control) may be useful.

5.2.2.1. SecurityRetries for S-HTTP Requests

In the case of a request that was made as an SHTTP request, it indicates that for some reason the cryptographic enhancements applied to the request were unsatisfactory and that the request should be repeated with the options found in the response header. Note that this can be used as a way to force a new public key negotiation if the session key in use has expired or to supply a unique nonce for the purposes of ensuring request freshness.

5.2.2.2. SecurityRetries for HTTP Requests

If the 420 code is returned in response to an HTTP request, it indicates that the request should be retried using S-HTTP and the cryptographic options indicated in the response header.

5.2.3. 421 BogusHeader

This error code indicates that something about the S-HTTP request was bad. The error code is to be followed by an appropriate explanation, e.g.:

421 BogusHeader Content-Privacy-Domain must be specified

Microsoft Exhibit 1019

5.2.4. 422 SHTTP Proxy Authentication Required

This response is analagous to the 420 response except that the options in the message refer to enhancements that the client must perform in order to satisfy the proxy.

5.2.5. 320 SHTTP Not Modifed

This response code is specifically for use with proxy-server interaction where the proxy has placed the If-Modified-Since header in the S-HTTP headers of its request. This response indicates that the following S-HTTP message contains sufficient keying material for the proxy to forward the cached document for the new requestor.

In general, this takes the form of an S-HTTP message where the actual enhanced content is missing, but all the headers and keying material are retained. (I.e. the optional content section of the CMS message has been removed.) So, if the original response was encrypted, the response contains the original DEK re-covered for the new recipient. (Notice that the server performs the same processing as it would have in the server side caching case of 7.1 except that the message body is elided.)

5.2.6. Redirection 3XX

These headers are again internal to HTTP, but may contain S-HTTP negotiation options of significance to S-HTTP. The request should be redirected in the sense of HTTP, with appropriate cryptographic precautions being observed.

5.3. Limitations On Automatic Retries

Permitting automatic client retry in response to this sort of server response permits several forms of attack. Consider for the moment the simple credit card case:

> The user views a document which requires his credit card. The user verifies that the DN of the intended recipient is acceptable and that the request will be encrypted and dereferences the anchor. The attacker intercepts the server’s reply and responds with a message encrypted under the client’s public key containing the Moved 301 header. If the client were to automatically perform this redirect it would allow compromise of the user’s credit card.

Microsoft Exhibit 1019

5.3.1. Automatic Encryption Retry

This shows one possible danger of automatic retries -- potential compromise of encrypted information. While it is impossible to consider all possible cases, clients should never automatically reencrypt data unless the server requesting the retry proves that he already has the data. So, situations in which it would be acceptable to reencrypt would be if:

1. The retry response was returned encrypted under an inband key freshly generated for the original request.
2. The retry response was signed by the intended recipient of the original request.
3. The original request used an outband key and the response is encrypted under that key.

This is not an exhaustive list, however the browser author would be well advised to consider carefully before implementing automatic reencryption in other cases. Note that an appropriate behavior in cases where automatic reencryption is not appropriate is to query the user for permission.

5.3.2. Automatic Signature Retry

Since we discourage automatic (without user confirmation) signing in even the usual case, and given the dangers described above, it is prohibited to automatically retry signature enchancement.

5.3.3. Automatic MAC Authentication Retry

Assuming that all the other conditions are followed, it is permissible to automatically retry MAC authentication.

6. Other Issues

6.1. Compatibility of Servers with Old Clients

Servers which receive requests in the clear which should be secured should return ’SecurityRetry 420’ with header lines set to indicate the required privacy enhancements.

6.2. URL Protocol Type

We define a new URL protocol designator, ’shttp’. Use of this designator as part of an anchor URL implies that the target server is S-HTTP capable, and that a dereference of this URL should undergo S-HTTP processing.

Microsoft Exhibit 1019

Note that S-HTTP oblivious agents should not be willing to dereference a URL with an unknown protocol specifier, and hence sensitive data will not be accidentally sent in the clear by users of non-secure clients.

6.3. Browser Presentation

6.3.1. Transaction Security Status

While preparing a secure message, the browser should provide a visual indication of the security of the transaction, as well as an indication of the party who will be able to read the message. While reading a signed and/or enveloped message, the browser should indicate this and (if applicable) the identity of the signer. Self-signed certificates should be clearly differentiated from those validated by a certification hierarchy.

6.3.2. Failure Reporting

Failure to authenticate or decrypt an S-HTTP message should be presented differently from a failure to retrieve the document. Compliant clients may at their option display unverifiable documents but must clearly indicate that they were unverifiable in a way clearly distinct from the manner in which they display documents which possessed no digital signatures or documents with verifiable signatures.

6.3.3. Certificate Management

Clients shall provide a method for determining that HTTP requests are to be signed and for determining which (assuming there are many) certificate is to be used for signature. It is suggested that users be presented with some sort of selection list from which they may choose a default. No signing should be performed without some sort of explicit user interface action, though such action may take the form of a persistent setting via a user preferences mechanism (although this is discouraged.)

6.3.4. Anchor Dereference

Clients shall provide a method to display the DN and certificate chain associated with a given anchor to be dereferenced so that users may determine for whom their data is being encrypted. This should be distinct from the method for displaying who has signed the document containing the anchor since these are orthogonal pieces of encryption information.

Microsoft Exhibit 1019

7. Implementation Notes

7.1. Preenhanced Data

While S-HTTP has always supported preenhanced documents, in previous versions it was never made clear how to actually implement them. This section describes two methods for doing so: preenhancing the HTTP request/response and preenhancing the underlying data.

7.1.1. Motivation

The two primary motivations for preenhanced documents are security and performance. These advantages primarily accrue to signing but may also under special circumstances apply to confidentiality or repudiable (MAC-based) authentication.

Consider the case of a server which repeatedly serves the same content to multiple clients. One such example would be a server which serves catalogs or price lists. Clearly, customers would like to be able to verify that these are actual prices. However, since the prices are typically the same to all comers, confidentiality is not an issue. (Note: see Section 7.1.5 below for how to deal with this case as well).

Consequently, the server might wish to sign the document once and simply send the cached signed document out when a client makes a new request, avoiding the overhead of a private key operation each time. Note that conceivably, the signed document might have been generated by a third party and placed in the server’s cache. The server might not even have the signing key! This illustrates the security benefit of presigning: Untrusted servers can serve authenticated data without risk even if the server is compromised.

7.1.2. Presigned Requests/Responses

The obvious implementation is simply to take a single request/response, cache it, and send it out in situations where a new message would otherwise be generated.

7.1.3. Presigned Documents

It is also possible using S-HTTP to sign the underlying data and send it as an S-HTTP messsage. In order to do this, one would take the signed document (a CMS or MOSS message) and attach both S-HTTP headers (e.g. the S-HTTP request/response line, the Content-Privacy-Domain) and the necessary HTTP headers (including a Content-Type that reflects the inner content).

Microsoft Exhibit 1019

```
SECURE * Secure-HTTP/1.4
Content-Type: text/html
Content-Privacy-Domain: CMS

Random signed message here...
```

This message itself cannot be sent, but needs to be recursively encapsulated, as described in the next section.

7.1.4. Recursive Encapsulation

As required by Section 7.3, the result above needs to be itself encapsulated to protect the HTTP headers. the obvious case [and the one illustrated here] is when confidentiality is required, but the auth enhancement or even the null transform might be applied instead. That is, the message shown above can be used as the inner content of a new S-HTTP message, like so:

```
SECURE * Secure-HTTP/1.4
Content-Type: application/s-http
Content-Privacy-Domain: CMS

Encrypted version of the message above...
```

To unfold this, the receiver would decode the outer S-HTTP message, reenter the (S-)HTTP parsing loop to process the new message, see that that too was S-HTTP, decode that, and recover the inner content.

Note that this approach can also be used to provide freshness of server activity (though not of the document itself) while still providing nonrepudiation of the document data if a NONCE is included in the request.

7.1.5. Preencrypted Messages

Although preenhancement works best with signature, it can also be used with encryption under certain conditions. Consider the situation where the same confidential document is to be sent out repeatedly. The time spent to encrypt can be saved by caching the ciphertext and simply generating a new key exchange block for each recipient. [Note that this is logically equivalent to a multi- recipient message as defined in both MOSS and CMS and so care must be taken to use proper PKCS-1 padding if RSA is being used since otherwise, one may be open to a low encryption exponent attack [HAST96].

Microsoft Exhibit 1019

7.2. Proxy Interaction

The use of S-HTTP presents implementation issues to the use of HTTP proxies. While simply having the proxy blindly forward responses is straightforward, it would be preferable if S-HTTP aware proxies were still able to cache responses in at least some circumstances. In addition, S-HTTP services should be usable to protect client-proxy authentication. This section describes how to achieve those goals using the mechanisms described above.

7.2.1. Client-Proxy Authentication

When an S-HTTP aware proxy receives a request (HTTP or S-HTTP) that (by whatever access control rules it uses) it requires to be S-HTTP authenticated (and if it isn't already so), it should return the 422 response code (5.7.4).

When the client receives the 422 response code, it should read the cryptographic options that the proxy sent and determine whether or not it is willing to apply that enhancement to the message. If the client is willing to meet these requirements, it should recursively encapsulate the request it previously sent using the appropriate options. (Note that since the enhancement is recursively applied, even clients which are unwilling to send requests to servers in the clear may be willing to send the already encrypted message to the proxy without further encryption.) (See Section 7.1 for another example of a recursively encapsulated message)

When the proxy receives such a message, it should strip the outer encapsulation to recover the message which should be sent to the server.

8. Implementation Recommendations and Requirements

All S-HTTP agents must support the MD5 message digest and MAC authentication. As of S-HTTP/1.4, all agents must also support the RSA-MD5-HMAC construction.

All S-HTTP agents must support Outband, Inband, and DH key exchange.

All agents must support encryption using DES-CBC.

Agents must support signature generation and verification using NIST-DSS.

Microsoft Exhibit 1019

9. Protocol Syntax Summary

We present below a summary of the main syntactic features of S-HTTP/1.4, excluding message encapsulation proper.

9.1. S-HTTP (Unencapsulated) Headers

```
Content-Privacy-Domain: ('CMS' | 'MOSS')
Prearranged-Key-Info: <Hdr-Cipher>,<Key>,<Key-ID>
Content-Type: 'message/http'
MAC-Info: [hex(timeofday)','']<hash-alg>','hex(<hash-data>)','
        <key-spec>
```

9.2. HTTP (Encapsulated) Non-negotiation Options

```
Key-Assign: <Method>','<Key-Name>','<Lifetime>','
        <Ciphers>';'<Method-args>
Encryption-Identity: <name-class>','<key-sel>','<name-args>
Certificate-Info: <Cert-Fmt>','<Cert-Group>
Nonce: <string>
Nonce-Echo: <string>
```

9.3. Encapsulated Negotiation Options

```
SHTTP-Cryptopts: <scope>';'<string>(,<string>)*
SHTTP-Privacy-Domains: ('CMS' | 'MOSS')
SHTTP-Certificate-Types: ('X.509')
SHTTP-Key-Exchange-Algorithms: ('DH', 'RSA' | 'Inband' | 'Outband')
SHTTP-Signature-Algorithms: ('RSA' | 'NIST-DSS')
SHTTP-Message-Digest-Algorithms:  ('RSA-MD2' | 'RSA-MD5' | 'NIST-SHS'
        'RSA-MD2-HMAC', 'RSA-MD5-HMAC', 'NIST-SHS-HMAC')
SHTTP-Symmetric-Content-Algorithms: ('DES-CBC' | 'DES-EDE-CBC' |
        'DES-EDE3-CBC' | 'DESX-CBC' | 'CDMF-CBC' | 'IDEA-CBC' |
        'RC2-CBC' )
SHTTP-Symmetric-Header-Algorithms: ('DES-ECB' | 'DES-EDE-ECB' |
        'DES-EDE3-EBC' | 'DESX-ECB' | 'CDMF-ECB' | 'IDEA-ECB' |
        'RC2-ECB')
SHTTP-Privacy-Enhancements: ('sign' | 'encrypt' | 'auth')
Your-Key-Pattern: <key-use>','<pattern-info>
```

9.4. HTTP Methods

```
Secure * Secure-HTTP/1.4
```


Microsoft Exhibit 1019

9.5. Server Status Reports

```
Secure-HTTP/1.4 200 OK
SecurityRetry 420
BogusHeader 421 <reason>
```

10. An Extended Example

We provide here a contrived example of a series of S-HTTP requests and replies. Rows of equal signs are used to set off the narrative from sample message traces. Note that the actual encrypted or signed message bodies would normally be binary garbage. In an attempt to preserve readability while still using (mostly) genuine messages, the bodies of the requests have been base64 encoded. To regenerate actual S-HTTP messages, it is necessary to remove the base64 encoding from the message body.

10.1. A request using RSA key exchange with Inband key reply

Alice, using an S-HTTP-capable client, begins by making an HTTP request which yields the following response page:

```
==========================================================
200 OK HTTP/1.0
Server-Name: Navaho-0.1.3.3alpha
Certificate-Info: CMS,MIAGCSqGSIb3DQEHAqCAMIACAQExADCABgkqh
        kiG9w0BBwEAAKCAM
        IIBrTCCAUkCAgC2MA0GCSqGSIb3DQEBAgUAME0xCzAJBgNVBAYTAlVTMSAwH
        gYDVQQKExdSU0EgRGF0YSBTZWN1cml0eSwgSW5jLjEcMBoGA1UECxMTUGVyc
        29uYSBDZXJ0aWZpY2F0ZTAeFw05NDA0MDkwMDUwMzdaFw05NDA4MDIxODM4N
        TdaMGcxCzAJBgNVBAYTAlVTMSAwHgYDVQQKExdSU0EgRGF0YSBTZWN1cml0e
        SwgSW5jLjEcMBoGA1UECxMTUGVyc29uYSBDZXJ0aWZpY2F0ZTEYMBYGA1UEA
        xMPU2V0ZWMgQXN0cm9ub215MFwwDQYJKoZIhvcNAQEBBQADSwAwSAJBAMy8Q
        cW7RMrB4sTdQ8Nmb2DFmJmkWn+el+NdeamIDElX/qw9mIQu4xNj1FfepfJNx
        zPvA0OtMKhy6+bkrlyMEU8CAwEAATANBgkqhkiG9w0BAQIFAANPAAYn7jDgi
        rhiIL4wnP8nGzUisGSpsFsF4/7z2P2wqne6Qk8Cg/Dstu3RyaN78vAMGP8d8
        2H5+Ndfhi2mRp4YHiGHz0HlK6VbPfnyvS2wdjCCAccwggFRAgUCQAAAFDANB
        gkqhkiG9w0BAQIFADBfMQswCQYDVQQGEwJVUzEgMB4GA1UEChMXUlNBIERhd
        GEgU2VjdXJpdHksIEluYy4xLjAsBgNVBAsTJUxvdyBBc3N1cmFuY2UgQ2Vyd
        GlmaWNhdGlvbiBBdXRob3JpdHkwHhcNOTQwMTA3MDAwMDAwWhcNOTYwMTA3M
        jM1OTU5WjBNMQswCQYDVQQGEwJVUzEgMB4GA1UEChMXUlNBIERhdGEgU2Vjd
        XJpdHksIEluYy4xHDAaBgNVBAsTE1BlcnNvbmEgQ2VydGlmaWNhdGUwaTANB
        gkqhkiG9w0BAQEFAANYADBVAk4GqghQDa9Xi/2zAdYEqJVIcYhlLN1FpI9tX
        Q1m6zZ39PYXK8Uhoj0Es7kWRv8hC04vqkOKwndWbzVtvoHQOmP8nOkkuBi+A
        QvgFoRcgOUCAwEAATANBgkqhkiG9w0BAQIFAANhAD/5Uo7xDdp49oZm9GoNc
        PhZcW1e+nojLvHXWAU/CBkwfcR+FSf4hQ5eFu1AjYv6Wqf430Xe9Et5+jgnM
        Tiq4LnwgTdA8xQX4elJz9QzQobkE3XVOjVAtCFcmiin80RB8AAAMYAAAAAAA
        AAAAA==
```


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```
   Encryption-Identity: DN-1779, null, CN=Setec Astronomy, OU=Persona
           Certificate,O="RSA Data Security, Inc.", C=US;
   SHTTP-Privacy-Enhancements: recv-required=encrypt

   <A name=tag1 HREF="shttp://www.setec.com/secret">
   Don't read this. </A>
   ============================================================

   An appropriate HTTP request to dereference this URL would be:

   ============================================================
   GET /secret HTTP/1.0
   Security-Scheme: S-HTTP/1.4
   User-Agent: Web-O-Vision 1.2beta
   Accept: *.*
   Key-Assign: Inband,1,reply,des-ecb;7878787878787878

   ============================================================

   The added Key-Assign line that would not have been in an ordinary
   HTTP request permits Bob (the server) to encrypt his reply to Alice,
   even though Alice does not have a public key, since they would share
   a key after the request is received by Bob.  This request has the
   following S-HTTP encapsulation:

   ============================================================
   Secure * Secure-HTTP/1.4
   Content-Type: message/http
   Content-Privacy-Domain: CMS

   MIAGCSqGSIb3DQEHA6CAMIACAQAxgDCBqQIBADBTME0xCzAJBgNVBAYTAlVTMSAw
   HgYDVQQKExdSU0EgRGF0YSBTZWN1cml0eSwgSW5jLjEcMBoGA1UECxMTUGVyc29u
   YSBDZXJ0aWZpY2F0ZQICALYwDQYJKoZIhvcNAQEBBQAEQCU/R+YCJSUsV6XLilHG
   cNVzwqKcWzmT/rZ+duOv8Ggb7oO/d8H3xUVGQ2LsX4kYGq2szwj8Q6eWhsmhf4oz
   lvMAADCABgkqhkiG9w0BBwEwEQYFKw4DAgcECFif7BadXlw3oIAEgZBNcMexKe16
   +mNxx8YQPukBCL0bWqS86lvws/AgRkKPELmysBi5lco8MBCsWK/fCyrnxIRHs1oK
   BXBVlsAhKkkusk1kCf/GbXSAphdSgG+d6LxrNZwHbBFOX6A2hYS63Iczd5bOVDDW
   Op2gcgUtMJq6k2LFrs4L7HHqRPPlqNJ6j5mFP4xkzOCNIQynpD1rV6EECMIk/T7k
   1JLSAAAAAAAAAAAAAA==
   ============================================================

   The data between the delimiters is a CMS message, RSA enveloped for
   Setec Astronomy.

   Bob decrypts the request, finds the document in question, and is
   ready to serve it back to Alice.
```


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An appropriate HTTP server response would be:

```
============================================================
HTTP/1.0 200 OK
Security-Scheme: S-HTTP/1.4
Content-Type: text/html

Congratulations, you've won.
<A href="/prize.html"
 CRYPTOPTS="Key-Assign: Inband,alice1,reply,des-ecb;020406080a0c0e0f;
 SHTTP-Privacy-Enhancements: recv-required=auth">Click here to
claim your prize</A>
============================================================
```

This HTTP response, encapsulated as an S-HTTP message becomes:

```
============================================================
Secure * Secure-HTTP/1.4
Content-Type: message/http
Prearranged-Key-Info: des-ecb,697fa820df8a6e53,inband:1
Content-Privacy-Domain: CMS

MIAGCSqGSIb3DQEHBqCAMIACAQAwgAYJKoZIhvcNAQcBMBEGBSsOAwIHBAifqtdy
x6uIMYCCARgvFzJtOZBn773DtmXlx037ck3giqnV0WC0QAx5f+fesAiGaxMqWcir
r9XvT0nT0LgSQ/8tiLCDBEKdyCNgdcJAduy3D0r2sb5sNTT0TyL9uydG3w55vTnW
aPbCPCWLudArI1UHDZbnoJICrVehxG/sYX069M8v6VO8PsJS7//hh1yM+0nekzQ5
l1p0j7uWKu4W0csrlGqhLvEJanj6dQAGSTNCOoH3jzEXGQXntgesk8poFPfHdtj0
5RH4MuJRajDmoEjlrNcnGl/BdHAd2JaCo6uZWGcnGAgVJ/TVfSVSwN5nlCK87tXl
nL7DJwaPRYwxb3mnPKNq7ATiJPf5u162MbwxrddmiE7e3sST7naSN+GS0ateY5X7
AAAAAAAAAAA=
============================================================
```

The data between the delimiters is a CMS message encrypted under a randomly-chosen DEK which can be recovered by computing:

```
        DES-DECRYPT(inband:1,697fa820df8a6e53)
```

where 'inband:1' is the key exchanged in the Key-Assign line in the original request.

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10.2. A request using the auth enhancement

There is a link on the HTML page that was just returned, which Alice dereferences, creating the HTTP message:

```
============================================================
GET /prize.html HTTP/1.0
Security-Scheme: S-HTTP/1.4
User-Agent: Web-O-Vision 1.1beta
Accept: *.*

============================================================
```

Which, when encapsulated as an S-HTTP message, becomes:

```
============================================================
Secure * Secure-HTTP/1.4
Content-Type: message/http
MAC-Info:31ff8122,rsa-md5,b3ca4575b841b5fc7553e69b0896c416,inband:alice1
Content-Privacy-Domain: CMS

MIAGCSqGSIb3DQEHAaCABGNHRVQgL3ByaXplLmh0bWwgSFRUUC8xLjAKU2VjdXJp
dHktU2NoZW1lOiBTLUhUVFAvMS4xClVzZXItQWdlbnQ6IFdlYi1PLVZpc2lvbiAx
LjFiZXRhCkFjY2VwdDogKi4qCgoAAAAA
============================================================
```

The data between the delimiters is a CMS 'Data' representation of the request.

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Appendix: A Review of CMS

CMS ("Cryptographic Message Syntax Standard") is a cryptographic message encapsulation format, similar to PEM, based on RSA's PKCS-7 cryptographic messaging syntax.

CMS is only one of two encapsulation formats supported by S-HTTP, but it is to be preferred since it permits the least restricted set of negotiable options, and permits binary encoding. In the interest of making this specification more self-contained, we summarize CMS here.

CMS is defined in terms of OSI's Abstract Syntax Notation (ASN.1, defined in X.208), and is concretely represented using ASN.1's Basic Encoding Rules (BER, defined in X.209). A CMS message is a sequence of typed content parts. There are six content types, recursively composable:

Data -- Some bytes, with no enhancement.

SignedData -- A content part, with zero or more signature blocks, and associated keying materials. Keying materials can be transported via the degenerate case of no signature blocks and no data.

EnvelopedData -- One or more (per recipient) key exchange blocks and an encrypted content part.

DigestedData -- A content part with a single digest block.

EncryptedData -- An encrypted content part, with key materials externally provided.

Here we will dispense with convention for the sake of ASN.1-impaired readers, and present a syntax for CMS in informal BNF (with much gloss). In the actual encoding, most productions have explicit tag and length fields.

```
Message = *Content
Content = Data | SignedData | EnvelopedData |
                DigestedData | EncryptedData
Data = Bytes
SignedData = *DigestAlg Content *Certificates
                 *CRLs SignerInfo*
EnvelopedData = *RecipientInfo BulkCryptAlg
                Encrypted(Content)
```


Microsoft Exhibit 1019

```
   DigestedData = DigestAlg Content DigestBytes
   EncryptedData = BulkCryptAlg Encrypted(Bytes)
   SignerInfo = CertID ... Encrypted(DigestBytes) ...
   RecipientInfo = CertID KeyCryptAlg Encrypted(DEK)
```

Appendix: Internet Media Type message/s-http

In addition to defining the S-HTTP/1.4 protocol, this document serves as the specification for the Internet media type "message/s-http". The following is to be registered with IANA.

```
        Media Type name:        message
        Media subtype name:     s-http
        Required parameters:    none
        Optional parameters:    version, msgtype

          version: The S-HTTP version number of the enclosed message
          (e.g. "1.4"). If not present, the version can be
                determined from the first line of the body.

          msgtype: The message type -- "request" or "response".
                If not present, the type can be determined from the
                first line of the body.

          Encoding considerations: only "7bit", "8bit", or "binary"
                are permitted.

          Security considerations: this is a security protocol.
```

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Security Considerations

This entire document is about security.

Microsoft Exhibit 1019

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