Operating Systems and Middleware: Supporting Controlled Interaction

Max Hailperin
Gustavus Adolphus College

Revised Edition 1.1
July 27, 2011

Chapter 9

Networking

9.1 Introduction

The overarching theme of this book is how computations interact with one another with support from operating systems and middleware. In Chapter 8 you saw that computations running at different times can interact by using persistent storage. In this chapter, you will see that computations can be distributed in space as well as time by sending messages across networks.

Networking is a topic that merits its own courses and textbooks. My goal here is to give an overview or review, depending on whether you have studied the topic before. I particularly highlight the ways in which networking ties in with the division of responsibilities between operating systems, middleware, and application software. Chapter 10 goes into more depth on the middleware commonly used to support distributed applications.

Because black boxes are inimical to education, I provide more detail about networking than is absolutely necessary for the development of distributed systems. However, my objective is not to make you a network engineer, capable of monitoring congestion and configuring routers, or even to give you a start towards that goal. (Though if you do pursue that goal, my overview of networking should not hurt you.) Instead, I am trying to explain the foundations that underlie distributed systems. In this chapter, not only do I spend a lot of time on the foundations, but also some time on such higher-level structures as the web and distributed file systems. Chapter 10 moves completely away from networking per se and into the middleware most commonly used to build distributed systems.

9.1.1 Networks and Internets

Before going any further, I should be clear about the meaning of three closely related terms: "a network," "an internet," and "the Internet." I will start by describing what networks and internets have in common and then describe the essential difference. Once you understand the general concept of an internet, I will be able to define the Internet as one specific internet.

A network is a collection of *links* and *switches*; so is an internet. Links are communication channels, such as wires, optical fibers, or radio channels. Switches are devices that connect links together and forward data between them. Some switches are known by more specific names; for example, those that connect radio links to wired links are known as *access points*, and those that connect the constituent networks of an internet are known as *routers*, as I will discuss subsequently.

Both networks and internets have computers interfaced to some of the links, as shown in in Figure 9.1, with each interface identified by an address. Any interfaced computer can transmit data tagged with a destination address, and under normal circumstances the data will make its way through the appropriate links and switches so as to arrive at the specified destination. (As a simplification, I will ignore *multicast*, in which a single message can be delivered to more than one destination interface.) A chunk of data tagged with an address is informally called a *packet*; later I will introduce a variety of more specific terms (*datagram*, *segment*, and *frame*), each of which is synonymous with *packet* but implies a particular context.

For a single *network*, as opposed to an internet, the preceding description is essentially the entire story. Data injected into the network is tagged only with the destination address, not with any information about the route leading to that address. Even the word "address" may be misleading; addresses on a network do not convey any information about physical location. If you move a computer to somewhere else on the same network, it will still have the same address.

Thus, a packet of data with a network address is not like an envelope addressed to "800 W. College Ave., St. Peter, MN 56082, USA," but rather like one addressed to "Max Hailperin." The network needs to figure out where I am, as well as what path through the links and switches leads to that location. As such, the switches need to take considerable responsibility.

In part, switches within networks shoulder their responsibility for delivering data by keeping track of each interface's last known location. In part, the switches take the simpler approach of forwarding data every which way, so that it is sure to run into the destination interface somewhere. (However,

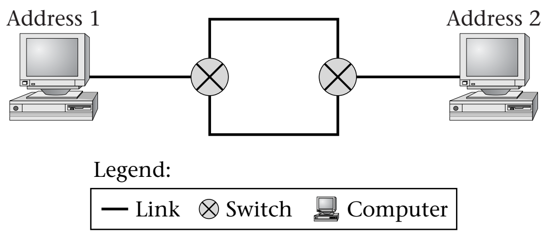


Figure 9.1: This network (or internet) contains four links, two switches, and two interfaced computers. Two alternative paths connect the two computers. As described in the text, more information would be needed to determine whether this is a picture of a single network or an interconnected group of networks, that is, an internet.

the forwarding must not be so comprehensive as to cause data to flow in cycles.) Neither approach scales well. As such, networks are normally confined to a limited number of interfaces, such as one workgroup within a company. When the network's scale is small geographically as well as in number of interfaces, it is called a *local area network* (*LAN*). Conversely, a *wide area network* (*WAN*) ties together interfaces that are far apart, though they are generally still few in number, perhaps even just two.

Multiple networks can be linked together into an *internet* by using *routers*, which are switches that connect to more than one network, as shown in Figure 9.2. In order to distinguish internets from networks, I still need to explain why linking networks together doesn't just result in a single larger network.

The distinguishing feature of an internet is that the destination addresses for the the data it conveys are two-part internet addresses, identifying both the destination network and the specific computer interface on that network. Returning to my real-world analogy, a packet of data with an internet address is like an envelope addressed to "Max Hailperin, Gustavus Adolphus College." There are people all over the world (analogous to routers) who could figure out how to forward the envelope to Gustavus Adolphus College. Once the envelope was on my college campus, people (analogous to switches within my local network) could forward the envelope to me.

Internets work similarly. Each router figures out what the next router should be in order to reach the destination network, independent of the specific computer on that network. The data is forwarded from each router to the next using the ordinary mechanisms of each constituent network, and likewise is forwarded from the last router to the destination computer using

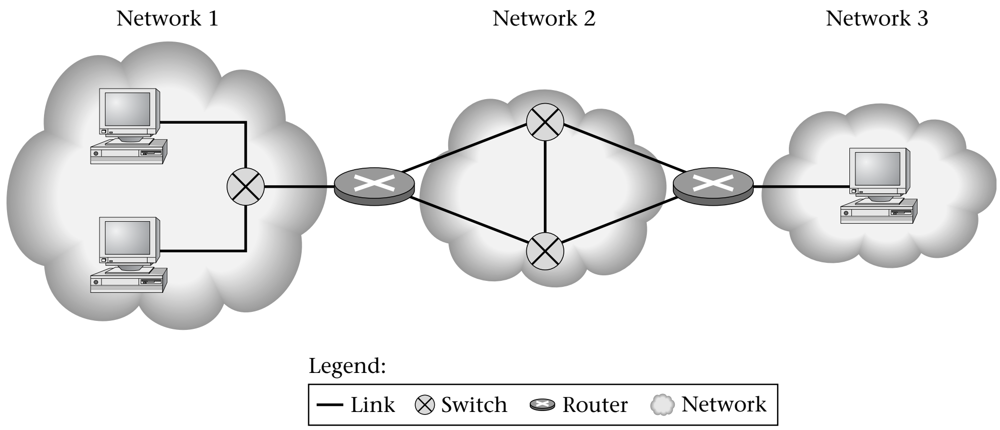


Figure 9.2: This internet was formed by connecting three networks. Each connection between networks is provided by a router, which is a switch interfaced to links belonging to two or more networks.

the destination network's mechanisms.

The two-part structure of internet addresses comes with a cost; when a computer is moved from one network to another, it must be assigned a new internet address. However, in return for this cost, an internet can scale to a much larger size than an individual network. In particular, one internet now connects such a large fraction of the world's computers that it is simply known as "the Internet."

9.1.2 Protocol Layers

Network communication is governed by sets of rules, known as *protocols*, which specify the legal actions for each partner in the dialog at each step along the way. For example, web browsers communicate with web servers using HTTP (Hypertext Transfer Protocol), which specifies the messages the browser and server can legally send at each step. In Section 9.2.1, I will show what those messages actually look like. For now, however, I will paraphrase the messages in plain English in order to illustrate the notion of a protocol.

When a web browser asks a web server to download a particular web page only if it has changed since the version the browser has cached, the server may legitimately respond in several different ways, including:

- "It changed; here is the new version."

- "No change, so I won't bother sending the page."
- "I have no clue what page you are talking about."

However, the web server is not allowed to give any of those responses until the question is asked, and it is also not allowed to give other responses that might be legal in other circumstances, such as "I created that new page per your upload." Not surprisingly, HTTP also forbids the web server from responding with something like "mailbox full" that would be appropriate in a different protocol, the one used to deliver email.

When humans converse, they talk not only about the subject of the conversation ("We could sure use some rain.") but also about the conversation itself ("I didn't catch that, could you say it again?"). Similarly, computers use not only *application protocols*, like the ones for downloading web pages and sending email messages, but also *transport protocols*, which control such matters as retransmitting any portions of the message that get lost.

An application protocol can be viewed as layered on top of a transport protocol, because the designers of the application protocol take for granted the services provided by the transport protocol. With the most common transport protocol, TCP (Transmission Control Protocol), the application protocol designers assume the transport protocol will take care of reliably sending a stream of bytes and having them arrive in order, without duplication or loss. All that need concern the application protocol designer is what the bytes should be to encode the various messages and responses. Meanwhile, the transport protocol designer doesn't worry about what bytes need streaming from one computer to another, just about the mechanisms for packaging chunks of bytes with sequence numbers, retransmitting lost chunks, and assembling the chunks together at the receiving end based on their sequence numbers. Thus, the layering of the two protocols results in a separation of concerns; each protocol can be designed without concern for the details of the other.

The transport layer is also responsible for allowing each pair of computers to engage in more than one conversation, a feature known as *multiplexing*. For example, a web browser on my desktop computer can be requesting web pages from the college's main server at the same time as my email program is delivering outgoing email to that same server. Each transport-layer connection is identified not only by the internet addresses of the two computers, but also by a *port number* on each end, which identifies a specific communication endpoint. My web browser connects to one port number on the server while my email program connects to another. The transport-layer software on the receiving computer delivers the data for each port number to the appropriate application-layer software, that is, it *demultiplexes* the arriving data.

The transport protocol can in turn be simplified by assuming that it has a *network protocol* under it, which makes its best effort to deliver a chunk of data to an internet address. The transport protocol may use this service for sending fresh chunks of application data, handed to it from the application layer, or for sending retransmissions. It may also use it for its own internal purposes, such as sending acknowledgments indicating what data has been received versus what needs retransmission. Regardless, from the perspective of the network protocol, these are all just packets to deliver. Meanwhile, from the perspective of the transport layer, delivery just happens; details like routing need not concern it.

The network layer is actually something of a misnomer, it that it is responsible for routing data through an internet. In fact, the most common network protocol is called the Internet Protocol (IP). This protocol is used to attempt to deliver data to any internet address, possibly by way of intermediate routers. Underneath it are two more layers, which are genuinely concerned with individual networks: the link and physical layers. I'll say more about these layers in Section 9.5. For now, it suffices to say that these are the layers implemented by networking hardware, such as Ethernet or Wi-Fi network cards, for wired or wireless LANs, respectively.

Counting up from the bottom of the stack, the physical, link, network, and transport layers are frequently referred to as layers 1, 2, 3, and 4. You might think that the application layer is 5, but in fact there are two layers I omitted, the session and presentation layers, which are layers 5 and 6. Therefore, the application layer is layer 7. The only reason you need to know these numbers is because they frequently show up in discussions of networking devices such as firewalls. For example, someone may tell you that their firewall does "filtering based on level 7 content." What this says is that the firewall looks at the specific contents of web page requests or email transmissions.

The listing of seven layers, illustrated in Figure 9.3, is known as the *OSI (Open Systems Interconnection) reference model*. I omit layers 5 and 6 from my subsequent discussions because they are not part of the architecture of the Internet, which was developed prior to the OSI reference model. In the Internet architecture, the application layer takes on the additional responsibilities, such as character set encoding and the establishment of network connections, that are assigned to the presentation and session layers in the OSI reference model. I will also largely fold together layers 1 and 2, because the difference doesn't matter unless you are engineering network hardware.

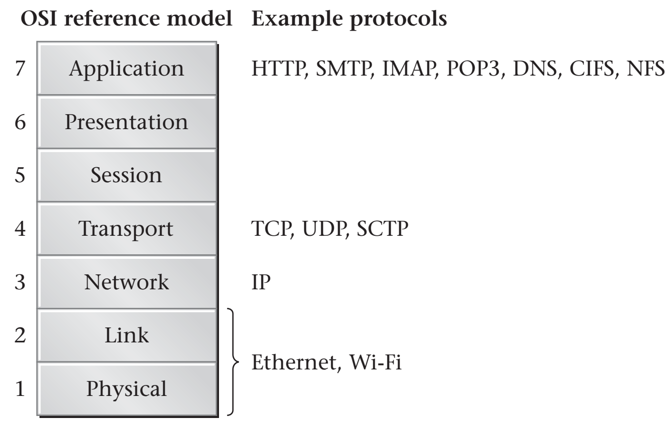


Figure 9.3: This diagram of the seven protocol layers in the OSI reference model provides examples for the layers I discuss.

As such, the bulk of this chapter is divided into four sections, one each for the application layer (9.2), the transport layer (9.3), the network layer (9.4), and the combination of link and physical layers (9.5). Those four sections are followed by my usual section on security (9.6), and by exercises, projects, and notes.

9.1.3 The End-to-End Principle

Traditionally, the Internet has been based on the *end-to-end principle*, which states that considerable control and responsibility should be in the hands of the endpoint computers interfaced to the Internet's periphery, with the routers and other devices interior to the Internet providing very simple packet delivery service. In terms of the protocol layering, this means that only end computers have traditionally concerned themselves with the transport and application protocols.

One virtue of the end-to-end principle is that two users can agree upon a new application protocol without needing the cooperation of anyone else. The ability to try new application protocols at the grassroots, and see whether they become popular, was very important in the evolution of the Internet up through the introduction of the web.

However, the Internet has been progressively moving away from the end-to-end principle. I already alluded to one example: firewalls that filter at the application layer. I will mention firewalls again in Section 9.6.2. However, there have also been other non-security-related forces leading away from the end-to-end principle; I will examine one in Section 9.4.3. One upshot of this is that today it may no longer be possible to just start using a new application protocol with its associated port number. Traffic on the new port number might well be blocked as it traverses the Internet.

This helps explain a popular use of web services, which I explain in Chapter 10. This form of communications middleware is often configured to package application programs' messages into web traffic, in effect layering yet another protocol on top of the web's application-layer protocol. This approach helps circumvent obstacles to new application-layer protocols within the Internet. For this chapter, I will stick with the traditional layers, topping out at the application layer.

9.1.4 The Networking Roles of Operating Systems, Middleware, and Application Software

Just as network protocols are layered, so too is the software that communicates using those protocols. However, the layering of networking software does not always correspond directly to the major divisions that I focus on in this book, between application software, optional middleware, and an operating system.

The most common division of roles in systems without middleware has application software responsible for the application-layer protocol, while the operating system handles everything from transport layer on down. That is, the API that operating systems present to application programs usually corresponds to the services of the transport layer. This transport-layer API is normally described as providing a socket abstraction; I will discuss socket APIs in Section 9.3.1.

In keeping with this division of roles, most application-layer protocols are the responsibility of application software. (For example, web browsers and email programs take responsibility for their respective application protocols.) There are a few interesting exceptions, however:

- The *Domain Name System* (*DNS*) maps names such as *www.gustavus.edu* into numerical internet addresses such as 138.236.128.22 using an application-layer protocol. Although it uses an application-layer protocol, it plays a critical supporting role for many different applications. In most systems, you can best think of the DNS software as a form of middleware, because it runs outside of the operating system kernel but supports application software.

- Distributed file systems run at the application protocol layer but need to be visible through the operating system's normal support for file systems. Often this means that at least some of the distributed file system software is part of the operating system kernel itself, contrary to the norm for application-layer protocols.

- In Chapter 10, you will see that many applications are expressed in terms of more sophisticated communication services than the socket API. For example, application programmers may want to send messages that are queued until received, with the queuing and dequeuing operations treated as part of atomic transactions. As another example, application programmers may want to invoke higher-level operations on objects, rather than just sending streams of bytes. In either case, middleware provides the necessary communication abstractions at the application layer, above the transport services provided by operating systems.

9.2 The Application Layer

Typical application-layer protocols include HTTP, which is used for browsing the web, SMTP (Simple Mail Transfer Protocol), which is used for sending email, POP3 (Post Office Protocol–Version 3), which is used for retrieving email, and IMAP (Internet Message Access Protocol), which is also used for accessing email. Rather than examining each of these, I'll present HTTP as an example in Section 9.2.1. Then I'll turn to some less-typical application protocols that play important roles behind the scenes: the Domain Name System, which I explain in Section 9.2.2, and various distributed file systems, which I explain in Section 9.2.3.

9.2.1 The Web as a Typical Example

When you use a web browser to view a web page, the browser contacts the web server using an application-layer protocol known as *HTTP* (*Hypertext Transfer Protocol*). This protocol has a request-response form; that is, after the browser connects to the appropriate port on the server (normally port number 80), it sends a request and then awaits a response from the server. Both the request and the response have the same overall format:

1. An initial line stating the general nature of the request or response

2. Any number of header lines providing more detailed information

3. A blank line to separate the header from the body
4. Any number of lines of message body

The message body is where the actual web page being downloaded (or uploaded) appears. For ordinary web browsing, it is empty in the request and non-empty in the response. A common case where a request has a non-empty body is when you fill in a form and submit it.

To take a concrete example, let's see how you could retrieve my home page, *http://www.gustavus.edu/+max/*, without the benefit of a web browser. You can use the program called `telnet` to connect to the web server's port 80 using the command

```
telnet www.gustavus.edu 80
```

Then you can type in the following three lines, the last of which is blank:

```
GET /+max/ HTTP/1.1
Host: www.gustavus.edu

```

The first of these is the request line stating that you want to get my home page using version 1.1 of the protocol. The second is a header line, indicating which web host you want to get the page from. This is necessary because some web servers have different aliases and may serve different content depending on which host name you are using. The blank line indicates that no more header lines are being specified.

At this point, the server should respond with a large number of lines of output, of which the first ones will look something like

```
HTTP/1.1 200 OK
Date: Sun, 16 Jan 2005 01:18:19 GMT
Server: Apache
Last-Modified: Sun, 16 Jan 2005 01:18:25 GMT
ETag: W/"30ba07-b94-21857f40"
Accept-Ranges: bytes
Content-Length: 2964
Connection: close
Content-Type: text/html; charset=UTF-8

<!DOCTYPE HTML PUBLIC "-//W3C//DTD HTML 4.01 Transitional//EN"
       "http://www.w3.org/TR/html4/loose.dtd">
```

```
<html lang="en">
<head>
<title>Max Hailperin’s home page</title>
</head>
<body>
<h1>Max Hailperin</h1>
```

and the last two will be

```
</body>
</html>
```

The first line of the response says that the request was OK and will be satisfied using HTTP version 1.1. (The number 200 is a status code, which indicates that the request was successful.) The server then sends quite a few header lines; you can probably figure out what several of them mean. For example, the Content-Length header indicates that my home page contained 2964 bytes at the time I tried this example. The Content-Type line describes how the web browser should interpret the message body. In this case, it is a text file written using *HTML* (*HyperText Markup Language*) and with the character set being an international standard known as UTF-8 (Unicode Transformation Format 8). The boundary between the headers and the message body is formed by the blank line. If you are familiar with the syntax of HTML, you can see that the body is indeed written in HTML. The HTML format is independent of the HTTP protocol, which can be used for transferring any kind of file; the most familiar other formats on the web are those used for images.

The HTTP standard includes many features beyond those shown in this one simple example. To illustrate just one more, consider sending another request, similar to the first but with one additional header:

```
GET /+max/ HTTP/1.1
Host: www.gustavus.edu
If-none-match: W/"30ba07-b94-21857f40"

```

This time, the reply from the web server is much shorter:

```
HTTP/1.1 304 Not Modified
Date: Sun, 16 Jan 2005 01:19:55 GMT
Server: Apache
Connection: close
```

```
ETag: W/"30ba07-b94-21857f40"
```

This corresponds with the scenario described in Section 9.1.2. The browser (or a human using `telnet` to simulate a browser) is asking "please send this web page only if it has changed since the version I previously downloaded." The version is identified using the *ETag* (*entity tag*) the server provided when it sent the previous version. In this case, the version on the server still is the same (matches the provided tag), so the server just sends a short reply to that effect. A browser could use this to validate continuing to use a cached copy.

9.2.2 The Domain Name System: Application Layer as Infrastructure

The network layer takes responsibility for routing a packet of data to a specified internet address. However, the internet addresses that it understands are numbers, encoding the destination network and the specific interface on that network. Humans don't generally want to use these numeric addresses; instead, they prefer to use names such as *www.gustavus.edu*. Thus, no matter whether you are using HTTP to browse the web or SMTP to send email, you are probably also using an additional application-layer protocol behind the scenes, to translate names into numerical addresses. This protocol is known as the *Domain Name System* (*DNS*), because the hierarchically structured names such as *www.gustavus.edu* are known as *domain names*.

The Domain Name System is actually a general facility that allows machines distributed around the Internet to maintain any arbitrary mappings of domain names to values, not just mappings of computers' names to their numerical internet addresses. However, for the sake of this overview, I will concentrate on how DNS is used in this one particularly important context.

The use of domain names to refer to internet addresses is quite analogous to the use of pathnames to refer to files, a topic I addressed in Section 8.6.3. In the following paragraphs, I will describe four aspects of this analogy. First, both kinds of names are hierarchical. Second, both kinds of names can be either absolute or relative. Third, both naming systems allow one object to directly have multiple names. And fourth, both naming systems also allow a name to indirectly refer to whatever some other name refers to.

A domain name such as *www.gustavus.edu* specifies that *www* should be found as a subdomain of *gustavus*, which is in turn a subdomain of *edu*. Thus, the structure of the name is similar to a pathname from a POSIX file system, which might be `edu/gustavus/www` for the file `www` within the subdirectory `gustavus` of the directory `edu`. The only two differences are that the components of a domain name are separated with dots instead of slashes, and that they are listed from most specific to least specific, rather than the other way around.

In POSIX pathnames, the difference between `edu/gustavus/www` and `/edu/gustavus/www` (with an initial slash) is that the former starts by looking for `edu` in the current working directory, whereas the latter starts from the root directory of the file system. These two options are called relative and absolute pathnames. One little-known fact about the DNS is that domain names also come in these two varieties. The familiar domain name *www.gustavus.edu* is relative, and so may or may not refer to my college's web server, depending on the context in which it is used. If you want to be absolutely sure what you are talking about, you need to use the absolute domain name *www.gustavus.edu.* complete with the dot on the end. On the other hand, only a cruel system administrator would set up a system where *www.gustavus.edu* was interpreted as *www.gustavus.edu.horrible.com.* rather than the expected site. The real reason for relative domain names is to allow shorter names when referring to computers within your own local domain.

My discussion of file linking in Section 8.6.3 explained that the simplest form of linking is when two names directly refer to the same file. Similarly, two domain names can directly refer to the same internet address. In the DNS, a domain name can have multiple kinds of information, or *resource records*, each with an associated type. The domain name has a directly specified internet address if it has a resource record of type A. (The letter A is short for address.) As an example, the domain names *gustavus.edu.* and *ns1.gustavus.edu.* both have type A resource records containing the address 138.236.128.18, so both of these domain names are referring directly to the same internet address.

Recall that symbolic links (or soft links) are pathnames that do not refer directly to a file, but rather indirectly to whatever another pathname refers to. Similarly, the DNS supports domain names that are aliases for other domain names. As an example, the domain name *www.gustavus.edu.* currently has no type A resource record. Instead, it has a type CNAME resource record, showing that it is an alias for *www.gac.edu.* Looking this second name up in the DNS, I find that it too is an alias, with a CNAME record referring to *charlotte.gac.edu.* Only this third domain name has the actual type A record, specifying the internet address 138.236.128.22. This internet address will be returned by a lookup operation on any of the three alternative domain names. The domain name at the end of a chain of aliases is known as the *canonical name*, which explains why the resource record type is called CNAME.

In order to translate a name into an address, an application program such as a web browser uses a system component known as the *resolver*. The resolver communicates using the DNS application-layer protocol with a name server, which provides the requested information. In most systems, the resolver is not part of the operating system kernel. Instead, it is linked into each application program as part of a shared library. From the operating system's perspective, the application program is engaged in network communication with some remote system; the fact that the communication constitutes DNS lookups is invisible. From the perspective of the application programmer, however, the resolver is part of the supporting infrastructure, not something that needs programming. As such, the resolver constitutes middleware in the technical sense of that word. However, it is conventionally marketed as part of the same product as the operating system, not as a separate middleware product.

The protocol used between the resolver and name server is a request-response protocol. The resolver indicates what information it is looking for, such as an internet address (type A resource record) for a particular domain name. The name server responds with the requested information, an error report, or a referral to another name server better able to answer the question.

The details of the DNS protocol are somewhat complicated for three reasons. One is that the system is designed to be general, not just suitable for internet address lookups. The second is that the system is designed to reliably serve the entire Internet. Therefore, it contains provisions for coordinating multiple name servers, as I outline in the next paragraph. The third is that the DNS protocol does not use ordinary lines of text, unlike the HTTP example I showed earlier. Instead, DNS messages are encoded in a compact binary format. As such, you cannot experiment with DNS using `telnet`. Exploration Projects 9.1 and 9.2 suggest some alternate ways you can experiment with DNS.

No one name server contains all the information for the complete DNS, nor is any given piece of information stored in only a single name server, under normal circumstances. Instead, the information is both partitioned and replicated, in the following three ways:

- The hierarchical tree is divided into zones of control that are stored independently. For example, my college maintains the information about all domain names ending in *gustavus.edu.* and *gac.edu.* on name servers we control. Additional resource records within the DNS itself indicate where the dividing lines are between zones.

- Authoritative information about names in each zone is stored on multiple name servers to provide failure tolerance and higher performance. Secondary servers for the zone periodically check with a master server for updates. Resource records within the DNS itself list all the authoritative name servers for each zone.

- Name servers cache individual pieces of information they receive from other name servers in the course of normal operation. Thus, when I repeatedly access *www.nytimes.com.*, I don't have to keep sending DNS queries all the way to the *New York Times*'s name server. Instead, my local name server acquires a non-authoritative copy of the information, which it can continue using for a specified period of time before it expires.

9.2.3 Distributed File Systems: An Application Viewed Through Operating Systems

Using HTTP, you can download a copy of a file from a remote server. Depending on how the server is configured, you may also be able to upload the file back to the server after editing it. Given that the file I am currently editing (containing this chapter) is stored on a centralized server, I could be making use of this download-edit-upload process. Instead, I am taking advantage of a more convenient, more subtle, kind of application-layer protocol, a distributed file system. In order to edit this chapter, or any other file stored on the central server, I simply access it by pathname, just the same way I would access a file on the local disk drive. Through some behind-the-scenes magic, certain parts of the file system directory tree are accessed over the network from the server. Ordinary file system operations, such as reading and writing, turn into network messages using the appropriate application-layer protocol.

Distributed file systems are most commonly used within the boundaries of a particular organization, unlike the protocols previously discussed. Perhaps for this reason, several different distributed file system protocols have remained viable, rather than a single standard dominating. Two of the most popular are *CIFS* (*Common Internet File System*) and *NFS* (*Network File System*). CIFS has primarily been championed by Microsoft and is commonly found in organizations with a substantial number of Microsoft

Windows systems. It frequently is still referred to by its previous name, the *SMB* (*Server Message Block*) protocol. (The individual messages sent by CIFS continue to be called Server Message Blocks.) NFS was developed by Sun Microsystems and is primarily used at sites where UNIX and Linux systems dominate. To confuse nomenclature further, one specific feature of CIFS is called DFS, for Distributed File System. I won't discuss that feature here and will continue to use the the phrase with lower-case letters to refer to distributed file systems in general.

As I will describe shortly, the designs of CIFS and NFS differ in some important regards. However, they also have quite a bit in common. In particular, in each case the client software needs to be at least partially located within the operating system kernel. When you use a pathname that extends into a directory tree supported by CIFS or NFS, the operating system kernel needs to recognize this fact and transfer control to the appropriate network client code, rather than the code that handles local file systems. The kernel can do this using a general purpose VFS (virtual file system) mechanism, as described in Section 8.8. The VFS mechanism delegates responsibility for file operations (such as reading or writing) to kernel code specific to the distributed file system. That kernel code may itself carry out the appropriate application-layer protocol exchange with a remote server, or it may just capture the details of the attempted file operation and pass them up to a specialized process outside the kernel, which actually does the network communication.

NFS is a pure request-response protocol, in the same sense as HTTP and DNS are: each interaction between client and server consists of the client sending a request first, then the server sending a response. CIFS, on the other hand, has a more complicated communication structure. Ordinary operations (such as reading from a file) are accomplished through messages paired in request-response form. However, the server can also spontaneously send a message to the client, without any request, to notify the client of some event of interest, such as that a file has changed or that another client wishes to access the same file. These notifications allow CIFS clients to cache file contents locally for performance, without needing to sacrifice correctness.

Another key difference in the two systems' designs concerns the amount of information servers maintain about ongoing client operations. The difference is most clear if you consider reading a file. In CIFS, the client invokes an operation to open the file for reading, then invokes individual read operations, and then invokes a close operation. These operations are much like the `open`, `pread`, and `close` operations described in Section 8.3. By contrast, NFS has no open and close operations; each read operation stands completely on its own, specifying the file that should be read as well as the position within it. One virtue of this "stateless" design is that the interaction between client and server can naturally tolerate either party crashing and being rebooted. On the other hand, a stateless design cannot readily support file locking or keeping client-side file caches up to date.

9.3 The Transport Layer

As mentioned earlier, the transport layer provides port numbers so that multiple communication channels can share (be multiplexed on) each internet address. Of the two transport-layer protocols common on the Internet, one provides essentially no services other than this multiplexing. This primitive transport protocol is called *UDP* (*User Datagram Protocol*). Like the underlying Internet Protocol, it makes an effort to deliver a chunk of data to a destination anywhere on the Internet, but does not guarantee reliability or that ordering will be preserved.

The other major transport-layer protocol—the one at work every time you browse a web page or send an email message—is the *Transmission Control Protocol* (*TCP*). This protocol does far more than provide port numbers; it provides the application layer with the ability to open reliable connections through which bytes can be streamed. A program using TCP opens a connection from a local port number to a remote port number at a specified internet address. Once the connection is open, each side can transmit bytes of data into its end of the connection and know that they will be received at the other end in order, without duplications or omissions. When the two parties are done communicating, they close the connection. In the HTTP examples of Section 9.2.1, the `telnet` program was used to open a TCP connection to the web server's port 80. The characters typed in for the request were streamed over the connection and hence received intact by the web server. Similarly, the web server's response was received by `telnet` and displayed.

The services provided by these transport-layer protocols are not so convenient for application programming as the higher-level messaging and distributed-object services I will present in Chapter 10. However, they are convenient enough to be widely used in application programming, and they are generally what operating systems provide. Therefore, in Section 9.3.1, I will present an overview of the socket application programming interfaces used to take advantage of these services. Thereafter, in Section 9.3.2, I will explain the basics of how TCP works. Finally, in Section 9.3.3 I will sketch the evolution of TCP into more modern versions, proposed future versions, and possible outright replacements.

9.3.1 Socket APIs

A *socket* is an object used as an endpoint for communication. Several different APIs revolve around the socket abstraction, each forming a variation on a common theme. The most important three are the POSIX socket API, the Windows socket API (known as Winsock), and the Java socket API. I will discuss all three briefly, but will give programming examples only for Java, as it is the easiest to use.

Ordinarily, each socket is associated with a local internet address and port number; that is, the socket knows its own computer's address and its own port number. If the socket is being used for a TCP communication stream, it will also be associated with a remote internet address and port number, identifying the communication partner. The local association is known as a *binding*; the socket is bound to its own address and port number. The remote association is known as a *connection*; the socket is connected to a partner.

Sockets used for UDP are not connected to partners; each time a packet of data, known as a *datagram*, is communicated using the socket, a remote internet address and port number need to be provided specifically for that datagram. As a convenience, if the same remote internet address and port number are to be used repeatedly, socket APIs generally allow the information to be provided once and for all using the connect operation, even though no real connection is formed. The address and port number are simply stored as default values for further datagram operations.

Each socket can be in any of several different states. The diagrams in Figures 9.4, 9.5, and 9.6 show three different life cycles through the states: one for datagram sockets (used with the UDP protocol), one for client-side stream sockets (initiating TCP connections), and one for server-side stream sockets (accepting incoming TCP connections). Several of the transitions, marked in the diagrams with dashed lines, do not require explicit operations in the Java API. The states are as follows:

- When freshly created, the socket may be *unbound*, with no address or port number. In this state, the socket does not yet represent a genuine communication endpoint but is just a hollow shell that has the potential to become an endpoint once bound. In the POSIX and Winsock APIs, all sockets are created unbound and are then bound

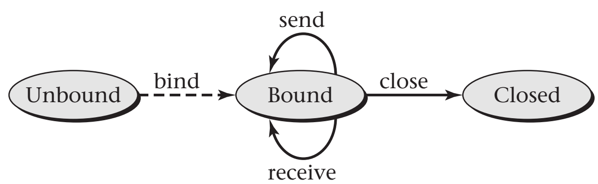


Figure 9.4: This state diagram shows the life cycle of datagram sockets used for sending or receiving UDP datagrams. In the Java API, the class `java.net.DatagramSocket` is used for this purpose, and binding happens automatically as part of the constructor.

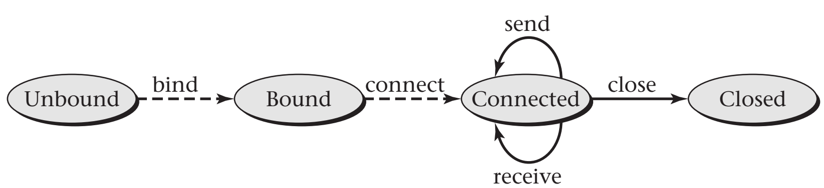


Figure 9.5: This state diagram shows the life cycle of client-side stream sockets used to initiate TCP connections. In the Java API, the class `java.net.Socket` is used for this purpose, and binding and connection ordinarily both happen automatically as part of the constructor.

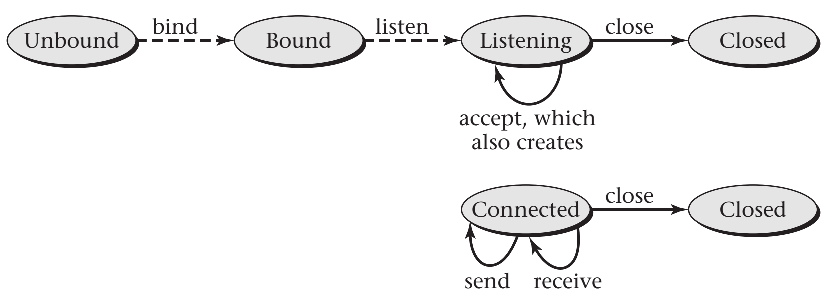


Figure 9.6: This state diagram shows the life cycle of server-side stream sockets used to accept TCP connections. In the Java API, the class `java.net.ServerSocket` is used for this purpose, and the bind and listen operations ordinarily are performed automatically as part of the constructor. Each time the accept operation succeeds, a new connected socket is returned, which in the Java API is an instance of `java.net.Socket`.

using a separate operation. In the Java API, you can create an unbound socket if you really want to (and then later bind it), but the normal constructors for the socket classes do the binding at the time the socket is created, saving a step.

- A socket can be *bound* but neither connected nor listening for incoming connection attempts. For UDP, datagrams can be sent or received in this state. For stream sockets, this state is only used as a stepping stone to the connected or listening state. In the Java API, the transition to the connected or listening state is generally accomplished at the time the socket is created, whereas in the POSIX and Winsock APIs, the connect and listen operations are explicit.

- A bound socket can be *connected* to a remote address and port number, forming a TCP connection over which bytes can be streamed in each direction.

- Alternatively, a bound socket can be *listening* for incoming connection attempts. Each time the application program accepts an incoming connection attempt, the socket remains in the listening state. A new socket is spawned off, bound to the same local address and port number as the original listening socket, but in the connected state rather than the listening state. The new connected socket can then be used to communicate with the client that initiated the accepted connection.

  A server program can in this way wind up with lots of sockets associated with the same local port number—one listening socket and any number of connected sockets. The TCP connections are still kept distinct, because each TCP connection is identified by four numbers: the internet addresses and port numbers on both ends of the connection.

- Finally, a socket should be *closed* when it is no longer needed. The socket data structure becomes a vestige of the former communication endpoint, and no operations can be legally performed on it.

To illustrate how a TCP server accepts incoming connections and then communicates using the resulting connected sockets, consider the Java program in Figure 9.7. This server contains an infinite loop that accepts only one connection at a time, reading from that connection, writing back to it, and then closing it before accepting the next connection. This would not be acceptable in a performance-critical setting such as a web server, because a slow client could hold all others up, as you can demonstrate in Exploration

```
import java.io.*;
import java.net.*;

class Server {
  public static void main(String argv[]) throws Exception {
    String storedMessage = "nothing yet";
    ServerSocket listeningSocket = new ServerSocket(2718);
    while(true) {
      Socket connectedSocket = listeningSocket.accept();
      BufferedReader fromClient = new BufferedReader
        (new InputStreamReader(connectedSocket.getInputStream()));
      PrintWriter toClient = new PrintWriter
        (connectedSocket.getOutputStream());
      String newMessage = fromClient.readLine();
      toClient.println(storedMessage);
      storedMessage = newMessage;
      toClient.close();
      fromClient.close();
      connectedSocket.close();
    }
  }
}
```

Figure 9.7: This message-storage server listens on port 2718 for connections. Each time it gets one, it reads a line of text from the connection to use as a new message to store. The server then writes the previous message out to the connection. For the first connection, the message sent out is `nothing yet`, because there is no previous message to deliver.

Project 9.5. In Programming Project 9.1, you will modify the server to spawn off a concurrent thread for each incoming client. Even sticking with the unmodified code, though, you can see that there may be many sockets associated with port 2718 as the program runs: one listening socket (of class `ServerSocket`) that exists the whole time the server is running, and a whole succession of connected sockets (of class `Socket`), one for each time a client connects. In a multithreaded version, several connected sockets could be in existence at the same time, all on port 2718.

If you compile and run the Java code from Figure 9.7, you can test out the server in the same way as shown in Section 9.2.1 for HTTP. That is, you can use the `telnet` program to connect to port 2718 on whatever machine is running the server, just as there I connected to port 80 on *www.gustavus.edu*. Once you connect with `telnet`, type in a line of text. You should see the `nothing yet` response and then see the connection close. Connect again (from the same or a different machine) and repeat the procedure. This time you should see the line of text you previously entered come back to you. If you find you can't connect to port 2718, there is probably a security firewall blocking your connection. The simplest workaround would be to limit yourself to testing connections from the same machine that is running the server program; connect using the special hostname *localhost*.

Rather than using `telnet` for the client side of this interaction, you could use a program written specifically for the purpose. This would demonstrate the other way TCP sockets are used, to connect from within a client program to a server's port. The program in Figure 9.8 directly forms a connected socket, bound to an arbitrary system-chosen port on the local host but connected to the specified host's port 2718. To try this program out, you could compile it and then run a command like

```
java Client localhost test-message
```

You should see in response whatever previous message was stored in the server. Moreover, repeating the command with a new message should retrieve `test-message`.

The preceding Java examples send only a single line of text in each direction over each connected socket. However, this is just a feature of the example I chose; in effect, it defines the nonstandard application-layer protocol being used. The same TCP transport layer (accessed through Java's socket API) could equally well carry any number of lines of text, or other sequences of bytes, in each direction. You would just need to insert a loop at the point in the program that reads from or writes to the connected socket.

```
import java.io.*;
import java.net.*;

class Client {
  public static void main(String argv[]) throws Exception {
    if(argv.length != 2){
      System.err.println("usage: java Client hostname msgToSend");
      System.exit(1);
    }
    String hostname = argv[0];
    String messageToSend = argv[1];
    Socket connectedSocket = new Socket(hostname, 2718);
    BufferedReader fromServer = new BufferedReader
      (new InputStreamReader(connectedSocket.getInputStream()));
    PrintWriter toServer = new PrintWriter
      (connectedSocket.getOutputStream(), true);
    toServer.println(messageToSend);
    String retrievedMessage = fromServer.readLine();
    System.out.println(retrievedMessage);
    toServer.close();
    fromServer.close();
    connectedSocket.close();
  }
}
```

Figure 9.8: This client program receives a hostname and a textual message string as command line arguments. It connects to the server running on the specified host's port 2718 and sends it a line of text containing the message. It then reads a reply line back and prints it out for the user to see.

For example, you could write an HTTP client or server in Java using this sort of code.

9.3.2 TCP, the Dominant Transport Protocol

You now understand how TCP can be used, through a socket API, to provide reliable transport of a byte stream in each direction over a connection between ports. Now I can take you behind the scenes and give you a brief overview of some of the techniques TCP uses to support reliable ordered byte streams. This will help you appreciate some of the difficult performance-critical issues. In this subsection, I will sketch TCP in its most well-established form; these TCP mechanisms are generally implemented within each operating system's kernel. Recent enhancements, as well as proposals for further change, are the topic of Section 9.3.3.

As the application program uses the kernel's socket API to send bytes, the kernel stores those bytes away in an internal buffer. From time to time, it takes a group of consecutive bytes from the buffer, adds a header of identifying information to the beginning, and sends it over the network to the receiver using the network layer, that is, IP. The chunk of bytes with a header on the front is called a *segment*. Each connection has a maximum segment size, typically no larger than 1460 bytes, exclusive of header. Thus, if the application program is trying to send a large number of bytes at once, the kernel will break it into several segments and send each. If the application program is sending only a few bytes, however, the kernel will wait only a little while for more bytes, and failing to get any, will send a small segment. One performance bottleneck is the copying of bytes from the application program to the kernel's buffer, and generally at least once more before reaching the network interface card. Systems optimized for network performance go to great lengths to reduce the number of times data is copied.

The header on each segment provides the port number for each end of the connection. It also specifies the position the segment's bytes occupy within the overall sequence being transmitted. For example, the first segment header might say "these are bytes 1 through 1000," and then the second segment header would say "these are bytes 1001 through 2000." The receiving code (also part of an operating system kernel) needs to pay attention to these sequence numbers and use them to deliver the bytes correctly to the application program that is using the socket API to read the data. Segments may arrive over the network out of order, for example, by taking two different routes. Thus, the kernel needs to store the arriving data in a buffer and return it to the application program in the order of sequence numbers, not in the order it arrives. As on the sending side, the trick is to do this without spending too much time copying data from one place to another.

In addition to arriving out of order, some segments may not arrive at all, because the network layer may occasionally lose a packet. To overcome that problem, TCP has mechanisms for retransmitting segments. The sender must continue to buffer each segment until its receipt is acknowledged, in case it needs to be retransmitted. Also, a segment believed to be lost may be retransmitted, and then turn out to not have been lost after all, but only delayed. Thus, the receiver needs to cope with duplicate segments.

Performance would be unacceptable if TCP transmitters sent only one segment at a time, waiting for acknowledgment before sending another. However, it would not be a good idea to allow arbitrarily many segments to be sent without waiting for acknowledgment. If a fast computer were sending to a slow computer, the receive buffer space could easily be overwhelmed. Thus, one of the many features TCP provides behind the scenes is *flow control*, which is to say, a receiver-controlled limit on how much unacknowledged data the sender is allowed to have outstanding at any time.

In traditional TCP, each acknowledgment contains a single number, n, to indicate that bytes 1 through n have been successfully received and that byte $n+1$ hasn't been. This style of acknowledgment, known as *cumulative acknowledgment*, is rather limited. Suppose the sender transmits seven segments of 1000 bytes each and only the first, third, fifth, and seventh arrive. The receiver will see four incoming segments and will send four acknowledgments, all saying bytes 1 through 1000 have been received. The sender will know that those bytes were received and have a pretty good clue that bytes 1001 through 2000 were not. It will also have a clue that three of the subsequent five segments were received, but it will have no idea which three.

The preceding example illustrates one scenario under which a TCP sender will retransmit a segment. Having received an acknowledgment of the first 1000 bytes and then three duplicates of that same acknowledgment, the sender is justified in assuming the second segment was lost and retransmitting it. The rules of TCP specify waiting for three duplicate acknowledgments, because one or two can easily occur simply from segments arriving out of order. That is, any duplicate acknowledgment indicates a hole has opened up in the sequence number order, but if segments are arriving out of order, the hole may quickly get filled in without needing retransmission.

Unfortunately, to provoke the triple duplicate acknowledgment, subsequent segments need to be transmitted. If the sender has no more segments to transmit, or is not allowed to send any more due to flow control restrictions or the congestion control restrictions I will describe shortly, then no duplicate acknowledgments will be triggered. Thus, TCP senders need to fall back on some other means of detecting lost segments; they use a timer. If no acknowledgment is received in a conservatively long time, then the segment is assumed lost. This conservative waiting period can cause substantial performance degradation.

A final challenge for TCP is controlling congestion that occurs at the switches (including routers) within the Internet. Each link leading out from a switch has a particular rate at which it can receive new data. Data destined for a particular outbound link may be coming into the switch from any number of the inbound links. If the total rate at which that data is flowing into the switch exceeds the rate at which it can be sent on the outbound link, then the switch will start to build up a queue of data awaiting forwarding. If the imbalance is only temporary, the queue will build up a little, then drain back down. However, if the imbalance persists, then the queue will grow long, creating lengthy delays, and then eventually get so full that the switch starts discarding packets. This phenomenon is known as *congestion*.

Congestion is not good, because it causes packets of data to be delayed or lost. Because TCP interprets unreasonably long delays as packet losses, either delays or outright losses can cause TCP to retransmit segments. This might even make the problem worse by sending more data to the already congested switch. Thus, TCP contains congestion-control features, which act to throttle back the rate at which it sends segments (new or retransmitted) when it detects packet losses. The theory is that most packet loss is caused by switches with full queues and therefore can be interpreted as a sign of congestion.

The details of congestion control are somewhat complicated. The most important facts to know are that it occurs independently in each TCP connection, and that newly opened connections start with a very low transmission rate, ramping up until the rate that causes congestion is detected. Thus, application performance can be improved by using multiple TCP connections in parallel and by sending a lot of data over each connection rather than repeatedly opening new connections for a little data apiece. Modern web browsers obey both these rules, using parallel and persistent connections. Parallel connections are a mixed blessing, because they constitute an attempt to unfairly compete with other Internet users, creating the potential for an arms race.

9.3.3 Evolution Within and Beyond TCP

Traditional TCP detects data loss through a combination of timeouts and triple duplicate cumulative acknowledgments. This detected data loss serves as the sign of congestion. TCP also responds to the detected data loss with retransmissions in order to ensure that all data is reliably delivered. Every one of these three design decisions has been challenged by networking researchers. That is, there are systems that detect loss in other ways, that detect congestion other than through loss, and that ensure reliability other than through retransmission. Some of the results are already partially deployed, whereas others remain research proposals. Some innovations also discard TCP's basic service model of the bidirectional byte stream. In this subsection, I will briefly overview a few of these trends in order to make the point that network protocols are not timeless truths, but rather are designs that are subject to change.

As network hardware has improved, the rate at which bits can be transmitted has greatly increased. However, the time needed for those bits to travel to the other side of the world and for acknowledgment bits to travel back has not shrunk. The consequence is that a computer may now transmit quite a few bits before getting any acknowledgment back. As a result, it is now common to have large numbers of unacknowledged TCP segments. In this situation, the weakness I mentioned for cumulative acknowledgment starts to become significant. There may well be more than one lost segment, and it would be nice to know exactly which ones were lost. For this reason, a *selective acknowledgment* feature was added to TCP, in which the receiver can provide the sender more complete information about which bytes have been received. This provides a new way to detect data loss.

In whatever manner data loss is detected, it likely stems from congestion. That does not mean, however, that the TCP sender needs to wait for a lost segment in order to sense congestion. If it could sense the congestion sooner, it could avoid the loss entirely. One way this can be done, deployed in some parts of the Internet, is for the routers to provide *Explicit Congestion Notification* (*ECN*). That is, they send an overt signal to the TCP transmitters to throttle back, rather than needing to implicitly code that signal by discarding a packet. Another approach, which has been experimented with, is for the TCP sender to notice that acknowledgments are starting to take longer and infer that a queue must be building up. This is called *TCP Vegas*.

Lost segments don't just signal congestion; they also prevent data from being delivered, necessitating retransmissions. However, there is another approach to ensuring that all data is delivered, even if some packets are lost. Namely, the data can be encoded into a redundant format, in which any sufficiently large subset of the packets contains enough information to allow all the data to be reconstructed. This concept is best explained by a highly simplified example, shown in Figure 9.9. This general approach is known as *forward error correction* using an *erasure code*. More sophisticated versions have been used to build high-performance systems for streaming large files, such as videos, using UDP.

One final area of change is in the service provided by TCP. An alternative transport protocol known as *SCTP* (*Stream Control Transmission Protocol*) is a proposed Internet standard that would offer similar reliable delivery and congestion control to TCP, but would go beyond the single bidirectional byte stream. An SCTP sender can transmit a stream of messages, that is, entire chunks of data akin to UDP datagrams, and have them delivered not only reliably and in their correct order, but also with the boundaries between them preserved, rather than all run together into an undifferentiated byte stream. Moreover, the SCTP connection can carry more than one stream in each direction. The messages on each individual stream are delivered in order, but a lost message on one stream doesn't hold up the delivery of messages on other streams. Unlike the trends mentioned previously, which affect only low-level details, if SCTP becomes popular, it will be necessary to rethink the APIs that constitute the interface between application programs and operating systems.

9.4 The Network Layer

The network layer delivers a packet of data to the appropriate destination computer on an internet. In this section, I will highlight a few aspects of this layer. First, I will explain the difference between the two versions of IP and explain how addresses are structured for the currently dominant version, IPv4. Second, I will give an overview of how routers forward packets along appropriate paths to reach their destinations. Finally, I will explain Network Address Translation (NAT), a technology that has considerable utility, but which is at odds with the original end-to-end architectural principle of the Internet.

9.4.1 IP, Versions 4 and 6

Each packet of data sent on the Internet has a header formatted in accordance with the *Internet Protocol* (*IP*). If the packet contains a TCP segment

Four data segments
10110110
00100111
10100010
01001011
Parity segment
01111000
Each column has an even number of 1s, so if any segment is lost, it can be reconstructed to fit that pattern.

Figure 9.9: Sending redundant data allows loss to be tolerated. Suppose four segments of data are to be sent; for simplicity, here each segment is only 1 byte long. Suppose the loss rate is low enough that it is unlikely two segments will be lost. Rather than waiting to see which one segment is lost, and then retransmitting it, a sender can transmit the four data segments and a parity segment, each with a sequence number in the header. Any one of the five can be lost, and yet all four data segments will be deliverable, because the lost segment can be reconstructed.

or UDP datagram, the TCP or UDP header follows the IP header. Each packet starting with an IP header is known as an IP datagram. Thus, an IP datagram can contain a UDP datagram. Because this is confusing, I will stick with the word "packet" when discussing IP and reserve "datagram" for UDP.

The most important pieces of information in the IP header are as follows:

- The version of IP being used, which governs the format of the remaining header fields; currently version 4 is dominant and version 6 is in limited use (version 5 does not exist)
- The internet address from which the packet was sent
- The internet address to which the packet should be delivered
- A code number for the transport-layer protocol, indicating whether the IP header is followed by a TCP header, UDP header, or whatever else

Among the other header fields I will not discuss are some that support optional extensions to the basic protocol.

The next-generation protocol, IPv6, differs from the currently dominant IPv4 in two principle ways. First, the source and destination internet addresses are much larger, 128 bits instead of 32. This should greatly ease assigning internet addresses to ubiquitous devices such as cell phones. Second, IPv6 was designed to support security features, protecting packets from interception, alteration, or forgery. However, these features have in the meantime become available as an optional extension to IPv4, known as *IPsec*.

Partially for this reason, the transition to IPv6 is happening exceedingly slowly, and IPv4 is still by far the dominant version.

As I explained in Section 9.1.1, an internet address contains two components: an identifier for a particular network and an identifier for a specific interface on that network. (My analogy in that section was with "Max Hailperin, Gustavus Adolphus College.") Given that IPv4 addresses are 32 bits long, you might ask how many of these bits are devoted to each purpose. For example, does the address contain a 16-bit network number and a 16-bit interface number? Unfortunately, the answer to this question is not so simple.

Each IPv4 address has a prefix, some number of bits long, that identifies the network, with the remainder of the 32 bits identifying the interface within the network. However, the length of the network prefix varies from network to network, so that internet addresses are not partitioned into their two components in a uniform way. The motivation for this awkward design is that the Internet needs both to support a large number of networks (more than 2^{16}, for example) and also some large networks (some containing more than 2^{16} interfaces, for example).

The conceptually simple solution to this problem would be to use larger fixed-format addresses, perhaps containing a 32-bit network number and a 32-bit interface number. However, the designers of IPv4 decided to stick with a total of 32 bits, because this address size was already in place from an early fixed-format version of internet addressing, in which the network identifier was always 8 bits long and the interface identifier 24 bits. The designers considered it more important to stick with their prior address size, 32 bits, than with their prior design concept, that the bits be partitioned in a uniform way. Thus, they made the design choice to cram all addresses into 32 bits by allowing a flexible division. This allows both for a small number of large networks (with short network prefixes) and a large number of small networks (with long network prefixes).

IPv4 addresses are conventionally written in *dotted decimal* format, in which the 32-bit address is divided into four 8-bit components, and each of the 8-bit components is translated into a decimal number. The four decimal numbers are written with dots between them. As an example, my computer's internet address is 138.236.64.64. Translating 138, 236, 64, and 64 from decimal to binary produces 10001010, 11101100, 01000000, and 01000000. Thus, my internet address in binary is 10001010111011000100000001000000.

Of these 32 bits, the first 21 identify the network for my college's department of mathematics and computer science, whereas the remaining 11 identify my specific computer on that network. My computer's operating system kernel is aware not only of its own internet address, but also of this division into 21 bits and 11. The latter fact is stored as a *mask*, which in my case is 255.255.248.0. If you translate that from dotted decimal to binary, you will see that the first 21 bits are 1s, whereas the last 11 bits are 0s.

The kernel uses this information whenever it sends out an internet packet. It compares the destination address to its own address, paying attention only to the prefix specified by the mask. Thus, in my case, the kernel checks whether the first 21 bits of the destination address are the same as my own. If so, the destination is on my own network, and my computer's kernel should send the data directly, using my network's own link-layer addressing mechanism.

If, on the other hand, the destination is outside my network, then the kernel should send the packet to the *gateway router* leading from my local network to the outside world. At the link layer, the kernel will send the packet out with the gateway router's network address, though it will still have the ultimate destination's internet address within the IP header.

9.4.2 Routing and Label Switching

In the ordinary functioning of the Internet, no entity actually selects a route for a packet of data to follow, in the sense of planning the entire route in advance. Instead, each time the packet arrives at a router, that router decides which neighboring router to forward the packet to. The overall route emerges as the composite result of all these local decisions.

When a router needs to forward a packet, it decide which neighboring router should be next by consulting its forwarding table. Each entry in the forwarding table specifies an internet address prefix and what the next router should be for that prefix. Given this large table, the router's forwarding decision can be made rather rapidly, because it just requires a table lookup. The one problem is that the entries in the table are keyed by variable-length prefixes, making the lookup operation more complicated than would be the case with fixed-length keys.

One other limitation of traditional internet routing, beyond the need to look up variable-length prefixes, is that all traffic for the same destination network gets forwarded to the same next router. Large service providers in the core of the Internet would prefer more flexible traffic engineering with the ability to send some of the traffic through each of several alternative routes. These same core providers are also the ones for whom expensive lookup operations on variable-length prefixes are most burdensome, because their routers need to switch traffic at a very high rate.

In order to address both these issue, some Internet service providers, particularly in the core of the Internet, are moving away from traditional IP routers to *label switching routers* using *Multiprotocol Label Switching* (*MPLS*). A label switching router looks up the next router, not using a variable-length prefix of the destination address, but instead using a fixed-length label that has been attached to the packet; MPLS specifies the standard format for this labeling.

When an IP packet first reaches a label switching router, the router attaches a label to it based on both the destination address and any traffic engineering considerations. Once the packet is labeled, it can be forwarded from label switching router to label switching router any number of times, based only on the label. When the packet is finally passed to a traditional router, the label gets stripped off.

With either approach, the performance-critical task of a router is forwarding individual packets based on a table lookup operation. However, the routers are also engaged in another less time-sensitive activity. Namely, the routers are constantly rebuilding their forwarding tables to reflect the most recent information they have about the Internet. They exchange information with one another using routing protocols. The study of routing protocols, and of approaches to generating forwarding table entries, is quite interesting, but I will leave it for networking texts.

9.4.3 Network Address Translation: An End to End-to-End?

Like many people, I have a network within my home, which uses the Internet Protocol. Moreover, from any of the computers on this network, I can open a TCP connection to any computer on the Internet. For example, I can browse any web site. However, my home network is not a constituent network of the Internet. This situation, which is actually quite typical for home networks, results from my use of *Network Address Translation* (*NAT*) within the router that connects my home network to my service provider's network. NAT is a technology that allows an entire network (or even an entire private internet) to pose as a single computer on the Internet. No matter which of my home computers connects to an external site, the external site will see the same source internet address, the one address that represents my whole network.

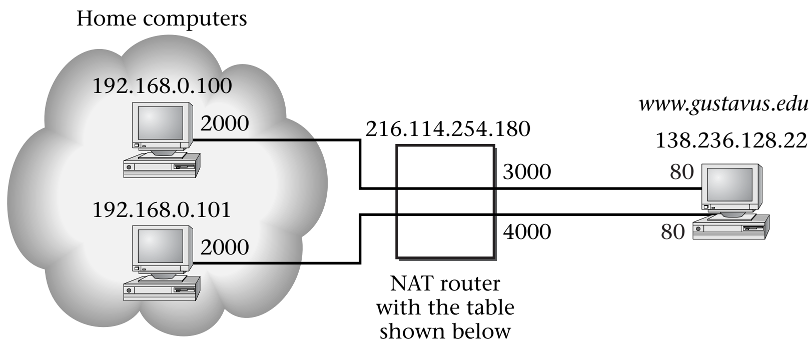
Each computer on my home network has its own private IP address; for example, one is 192.168.0.100 and another is 192.168.0.101. All packets on my home network use these addresses. However, as the router forwards packets out from the home network to the service provider, it modifies the packets, replacing these private IP addresses with the single public internet address my service provider assigned me, which is 216.114.254.180.

If all the NAT router did was to change the packets' source addresses, chaos would result. Recall that each TCP connection is uniquely identified by the combination of four values: the source and destination addresses and port numbers. Suppose I start browsing *www.gustavus.edu* on one home computer at the same time my wife does so on another of our home computers. Each of us is therefore opening a TCP connection with destination address 138.236.128.22 and destination port number 80. My source address starts out as 192.168.0.100, and my computer picks a source port number it is not otherwise using, perhaps 2000. My wife's source address starts out as 192.168.0.101, and her computer also picks a source port. Perhaps by coincidence it also picks 2000. Within our home network, our two TCP connections are distinguishable because of our two different IP addresses. However, outside the home network is another issue. If the NAT box rewrites our packets so both have source address 216.114.254.180 and leaves both our port numbers as 2000, then it will have combined what should have been two separate TCP connections into one.

To get around this problem, the NAT router rewrites our packets' source port numbers (in the TCP headers) as well as the source addresses (in the IP headers). Internally to our network, we have distinct addresses but may have coincidentally identical port numbers. Externally, we share a common address, but the NAT router makes sure we use different port numbers. For example, it might assign me port number 3000 and my wife port number 4000. Thus, the router would make two entries in a table for its own later reference. One entry would show that it should map all internal packets destined for the web server from address 192.168.0.100 and port number 2000 into external packets with address 216.114.254.180 and port number 3000. The other entry would show that traffic to the web server from internal address 192.168.0.101 with port number 2000 maps into external address 216.114.254.180 with port number 4000. Figure 9.10 illustrates this scenario.

Luckily, port numbers are not a scarce resource. The TCP header fields for source and destination port numbers are each 16 bits in size, yet computers do not ordinarily use anywhere near 2^{16} ports apiece. Therefore, the NAT router has no problem assigning distinct external port numbers for all the ports in use by any of the computers on the home network.

The NAT router has one further essential function. It must also rewrite in the reverse manner each packet coming in from the Internet at large to the private network. For example, if a packet arrives with source address 138.236.128.22, source port 80, destination address 216.114.254.180, and destination port 3000, then the NAT's table will show that this belongs to



Internal address	Internal port	External address	External port	Remote address	Remote port
192.168.0.100	2000	216.114.254.180	3000	138.236.128.22	80
192.168.0.101	2000	216.114.254.180	4000	138.236.128.22	80

Figure 9.10: A NAT router rewrites port numbers as well as addresses so two computers can share a single public address.

my connection, and so the NAT will modify the packet to show destination 192.168.0.100 with port 2000. By the time the packet reaches my computer, it will look as though the web server was directly communicating with me using my private address.

What happens if an external computer wants to initiate a TCP connection to one of my home computers? In the particular case of my home, it is out of luck. My NAT router is configured to forward only inbound packets that come as part of a connection initiated from the private network. However, the answer might be somewhat different on another network using a NAT router. Consider, for example, a business that chooses to use a NAT router. The business would allow outgoing connections from all its computers, just as I do at home. These outgoing connections would create temporary entries in the router's table of rewriting rules, just like in my router. However, the business would also allow incoming connections to port 80 (the standard HTTP port) for its main web server and to port 25 (the standard SMTP port) for its main email server. It would do this by configuring the NAT router with two permanent rewriting rules. Any packets coming to the business's public address on port 80 should get rewritten to the web server's private address, while retaining port 80. Any packets coming to the public address on port 25 should get rewritten to the email server's private address, while retaining port 25.

This example illustrates one of the problems with NAT routing, stemming from its violation of the end-to-end principle. Suppose someone within the business wants to start offering a new kind of Internet service using a new application-layer protocol that listens on a new port number. For this to work, the corporate network administrator would need to make a new entry in the NAT router's table, like the two for the web and email servers. This is a significant stumbling block for introducing new services. This is one reason why many services today are packaged inside HTTP traffic, directed to the usual port 80.

NAT routing has other problems as well. One of the fundamental ones is that IPsec is designed to prevent packets from being altered in transit, but NAT relies on doing exactly that. A technique for working around this difficulty has recently been developed, but it does at least introduce additional complexity.

Despite these problems, NAT routing is heavily used and becomes more so every day. The principle reason is that internet addresses are expensive and so are worth sharing. The negative consequences are ameliorated by the fact that most network administrators would prefer to put most internal computers off limits to external access anyhow, for security reasons.

9.5 The Link and Physical Layers

When you plug an Ethernet cable into a socket, the plug snaps into the socket because it is the right size. When your computer starts sending high and low voltages over that cable, they are high enough and low enough to be recognized as 1 and 0 by the equipment on the other end, but not so extreme as to fry that equipment. These are examples of issues addressed by the physical layer. Various physical-layer standards exist for Ethernet over fiber optics, Ethernet over twisted pairs of copper wires, and so forth.

Even granted that the physical layer can get bits from one end of a link to the other, there are still problems to solve. For example, how do computers on the local network address data to each other? Presumably, each chunk of data (known as a *frame*) needs to have a header that specifies a source address and destination address. However, suppose you plug a computer into a network, and it starts immediately hearing 1s and 0s, having come in on the middle of a frame. It should start paying attention at the start of the next frame. How can it recognize the boundary between frames? Perhaps there is some definite minimum space of silence between frames or some recognizable signal at the beginning of each frame. Shared links (like radio channels) pose another problem: how can the various computers take turns, rather than all transmitting at once? These issues of addressing, framing, and taking turns are concerns at the link layer, which is also sometimes known as the data link layer.

Computer systems have hardware devices that support both the link and physical layers. The operating system kernel provides this hardware with a frame of data to deliver, complete with the address of the recipient on the local network. The hardware does the rest. Conversely, the hardware may interrupt the operating system to report the arrival of an incoming frame.

Most of the issues at the link and physical layer have no direct bearing on operating systems or other software. The biggest exception is addressing. The two common kinds of networks (Ethernet and Wi-Fi) use 48-bit addresses that are totally independent from the 32-bit internet addresses. Thus, whenever the operating system sends out a packet of data to an internet address, and the address's prefix shows that it is on the local network, the kernel needs some mechanism for looking up the corresponding 48-bit hardware address, commonly known as a *MAC (Media Access Control) address.*

The kernel discovers MAC addresses using a network protocol called *ARP* (*Address Resolution Protocol*). The kernel broadcasts a request to all machines on the local network asking if any of them knows the MAC address corresponding to a particular IP address. The operating system kernels on all the receiving machines compare the requested address to their own. The one that sees its own IP address responds, providing its own MAC address. The requesting kernel stores the information away in a table for later reuse, so that it won't need to bother all other computers repeatedly.

Lots of network technologies have been developed, but two account for the bulk of networks today. Most networks that use wires or fiber optics use some version of the Ethernet standard, whereas most networks that use radio signals use some version of the Wi-Fi standard. (Wi-Fi is also frequently known by the less-catchy name 802.11, which is the identifying number of the working group that standardizes it.) Because these two use the same high-level interface, they can be integrated into combined networks, in which any device can communicate with any other device by MAC address, even if one is on the Ethernet portion of the network and the other on the Wi-Fi portion. Internet routing is not required. The switching devices that link Ethernet and Wi-Fi in this way are known as *access points.*

9.6 Network Security

Just as networking is a large field that I can only survey in this chapter, network security is a large area. My purpose in addressing it here is twofold. First, I want to impress upon you how important it is; if I remained silent, you might think it was unimportant. Second, by scratching the surface, I can give you some feel for some of the constituent topics.

Data security must extend beyond security for the systems on which the data is persistently stored. In today's world, the data is frequently also in transit over networks. For example, my students' grades are not only stored on the college's computer, they are also transmitted over the Internet every time I do grading from home. Thus, to be comprehensive, data security must include network security.

There are two key differences between persistent storage and network communication, however:

- Large amounts of data are available for long periods of time in persistent storage. Networks, on the other hand, generally carry any particular piece of data very fleetingly. Contrast gaining access to a merchant's database, containing all its customers' credit card numbers, with tapping into the network connection and snagging the few numbers that pass through during the time your interception is in operation.

- Persistent storage is directly accessible only to a very limited number of people who have physical access and who are subject to the risks of being physically apprehended. The Internet, on the other hand, is accessible to an entire world worth of malefactors, many of whom may be beyond effective reach of law enforcement.

When the Internet was less pervasive, the first of these factors was the dominant one, and network security was not such a major concern. Today, the second factor must be considered the dominant one. Keep in mind also that network adversaries are not limited to eavesdropping on, or modifying, data already passing through the network. They can also send messages that might trigger additional data flows that would not otherwise occur. Many computers (typically in homes) today are "owned" by network intruders. That is, the intruder has obtained complete control and can remotely command the computer to carry out any action, as though it were his or her own computer, including accessing any of the persistently stored data. The only way organizations such as companies and government agencies prevent their computers from being similarly "owned" is by devoting large amounts of attention to network security.

9.6.1 Security and the Protocol Layers

Security vulnerabilities and threats exist at each layer of the protocol stack. Similarly, defensive measures are possible at each level, whether to protect the confidentiality of transmitted data, or to ensure the authenticity and integrity of arriving data.

Many of the most notorious network security problems have been at the application layer. Examples include forged email and the SQL Slammer worm, which propagated by overflowing the memory space a particular application program used for incoming messages. Some of the application-layer vulnerabilities stem from fundamental protocol design decisions (such as that email can claim to come from anyone), whereas others come from implementation flaws (such as a program not checking whether it was putting more data into a buffer than it had room for).

These vulnerabilities can be combated directly by using better designs and more careful programming. However, it is unrealistic to expect perfection in this area, any more than in other human endeavors. Therefore, indirect methods should also be used to minimize risk. Section 9.6.2 mentions the role well-configured firewall and intrusion detection systems can play. To take one example, there was essentially zero need for organizations to allow traffic to come in from the Internet at large to the particular port number used by the SQL Slammer worm. This application-layer vulnerability ought to have been shielded by a firewall.

The application layer also provides plenty of opportunity to actively enhance security. For example, the email protocols can be retrofitted with cryptographic techniques to ensure that messages really come from their stated sender, have not been modified in transit, and are read only by their intended recipient; *PGP* (*Pretty Good Privacy*) and *S/MIME* (*Secure/Multipurpose Internet Mail Extensions*) do exactly that. To take another example, there is no reason why web browsers and web servers need to directly send HTTP messages over vulnerable TCP connections. Instead, they can interpose a layer of encryption known as the *Secure Sockets Layer* (*SSL*). Every time you visit a secure web site and see the padlock icon click shut, it means that SSL is in use. Conceptually this is between the main HTTP application layer and the TCP transport layer, but strictly speaking it is an application-layer protocol.

At the transport layer, TCP is subject to its own set of vulnerabilities.

Many of the denial-of-service attacks on network servers take place at this level; the server under attack is flooded by initial connection-establishment requests that are never followed up on. Proposals for fixing that problem in fundamental ways run into the difficulty of changing any protocol that is so widely deployed.

One example of a security-enhancement technology at the transport layer is an optional TCP feature for message authentication. This feature is particularly used by routers in order to secure their communication with neighboring routers. If it were possible for an intruder to inject bogus routing information, the Internet could be rendered unusable. Therefore, routers "sign" their routing update messages, using the optional TCP feature, and check the signatures on the updates they receive from neighboring routers.

One of the biggest security vulnerabilities at the network layer is that packets may have incorrect source addresses. The typical response to this problem is filtering at routers. For example, no packets should be allowed out of my college campus onto the Internet at large if the source address is not a legitimate one from the range assigned to this college. That would prevent someone here from pretending to be elsewhere.

I already mentioned that IPsec is a security technology at the network layer. The most common application of IPsec is when an organization has computers at several physical locations (including, frequently, within workers' homes) and wants to allow them all to communicate securely with one another, even though the traffic between locations is carried on the public Internet. IPsec supports this kind of *virtual private network* (*VPN*) by making sure every packet of data sent is encrypted, so as to be completely opaque to eavesdroppers, and so as to stymie any active intruder who would attempt to modify or inject packets.

Finally, the lowest layers of the protocol stack, the link and physical layers, are not immune from security issues. I will mention just two. One is that the ARP protocol, used to translate internet addresses into MAC addresses, was designed without any serious consideration of security issues. As a result, it is easy for any computer on a local network to take over an internet address that ought to belong to another computer on the same network. This is an attack more readily detected and responded to than prevented. To take a second example, Wi-Fi signals for many organizations can be picked up from the street outside. Moreover, the encryption built into early versions of Wi-Fi was faulty and even in newer versions is frequently not turned on. If you use Wi-Fi, you should definitely read one of the widely available tutorials on Wi-Fi security. These systems can be configured much more securely than they usually are.

9.6.2 Firewalls and Intrusion Detection Systems

A *firewall* is a system that imposes some restriction on the Internet traffic crossing a border, for example, between a company and the outside world, or between a particular computer and the rest of the Internet. Security-conscious organizations deploy multiple firewalls, protecting not only the outer perimeter, but also the borders between internal groups and around individual systems. Every computer installation that is hooked up to the Internet, even as small as a single home computer, should have at least one firewall.

A firewall can be a computer (or special-purpose hardware unit) devoted to the purpose, a router that has been configured to filter traffic, or software installed directly on the computer being protected. If the firewall operates correctly, any of these approaches is valid. However, if the firewall software itself is buggy, the consequences are likely to be more severe if it is operating on the same computer that is being protected. The best practice is to use a reputable external firewall at the organizational and workgroup perimeters and then software firewalls on individual computers. Home users should ideally use the same approach. The external firewall in this case may be a NAT router.

The big problem with firewalls is configuring them to let through only traffic that has a good reason to exist while blocking all other traffic. Empirical studies have shown that a large percentage of firewalls are misconfigured. Security-conscious organizations have their firewall configuration files examined by auditors and also have penetration testing performed, in which the auditors make attempts to gain access to the protected network.

In an organizational setting, there is pressure on network administrators to not configure firewalls too restrictively. If traffic necessary to the organization's functioning is blocked, someone will complain. These complaints could cost the administrator a job. In a home setting, on the other hand, you are likely to be complaining to yourself and can presumably stand the heat. Therefore, you should set all firewall settings as restrictively as possible, and wait and see what harm it does you. Loosen up on only those settings that prove to get in your way. This approach compensates for your inability to hire security auditors.

One of the most important steps an organization can take to preserve overall security is to use firewalls to isolate machines that are exposed to attack, so that even if those particular machines are "owned" by attackers, the damage is limited. As an example, consider a web server that does not support interactive transactions (such as shopping), but rather just disseminates information about the organization. A security-conscious configuration is as shown in Figure 9.11.

Suppose that the web server software has some bug, such that by sending some clever, over-long message to the server's normal port 80, an outside attacker can overwrite some critical memory and come to "own" the server, executing arbitrary code. Depending on the access controls in place on the server, the attacker may be able to deface the web site, replacing the organization's web pages with others. However, the attacker cannot mount any attack from the server to other machines, whether on the internal network or the external, because the firewall prohibits any outbound connections from the server. When employees of the organization want to reconfigure the server or put new information on it, they do so using connections they initiate from within the internal network.

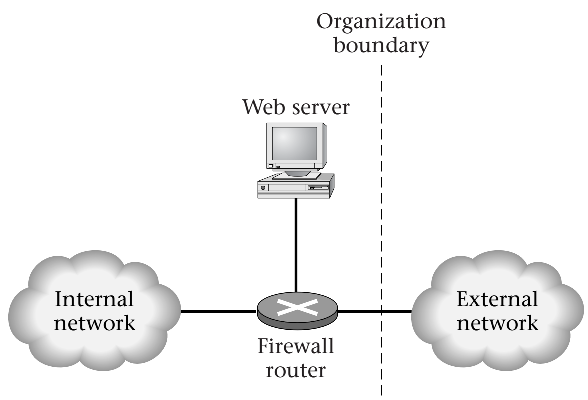
In addition to firewalls, all organizational networks should also include an *intrusion detection system* (*IDS*). This system monitors all network traffic looking for activity that does not fit the usual, legitimate patterns. The IDS can play two roles, both alerting the network administrators to the existence of a problem and capturing forensic evidence useful in crafting an appropriate response. The response can be both technical (such as removing an infected machine from the network) and non-technical (such as cooperating with law-enforcement officials). A properly configured IDS should protect not only against intrusions that breach the organizational perimeter, but also against attacks mounted by insiders.

9.6.3 Cryptography

Cryptography consists of mathematical techniques for transforming data in order to assure confidentiality or integrity and authenticity. Cryptography underlies much of network security, ranging from application-layer secure email and web browsing to link-layer encryption within the Wi-Fi protocol. Cryptography provides the means for legitimate communication to continue even as adversaries are thwarted. However, you should be aware that most practical security problems are outside the scope of cryptography. Rarely is there a report of encryption being broken, whereas misconfigured firewalls and systems vulnerable to buffer overflows are everyday occurrences.

Cryptographic techniques can be categorized in two independent ways:

- Some techniques rely on both the sender and the receiver knowing a *shared secret*, that is, a secret key that the two of them both know but intruders don't. Other techniques use a *key pair*, with one component



Configuration of the firewall router:

Initiator	**Target**	**Allowed ports**
external network	web server	80
internal network	web server	a few needed for operation
internal network	external network	none
external network	internal network	none
web server	any	none

Figure 9.11: This firewall configuration allows an organization's web server to provide static content to the outside world but allows for no other interaction. An organization with other needs would have other equally security-conscious modules added to this one.

known to the sender and the other to the receiver. These two options are known as *symmetric-key cryptography* and *asymmetric-key cryptography*. Because in many applications one half of a key pair can be made publicly known while the other is kept secret, asymmetric-key cryptography is also known as *public-key cryptography*.

- Some techniques *encrypt* the message, that is, transform the message so that it is not readable without the appropriate key, whereas other techniques leave the message itself alone but append a *Message Authentication Code* that allows the possessor of the appropriate key to verify that the message really comes from the legitimate sender and was not modified. Note that the abbreviation *MAC* is used in this context independently from its use in describing features of link-layer protocols, such as MAC addresses.

The more bits long a cryptographic key is, the more work the legitimate sender and receiver need to do, but also the more work any intruder needs to do. The goal in designing a cryptographic system is to make the legitimate parties' work scale up only modestly with key size, whereas the intruder's work scales up much more rapidly. That way, a key size can be chosen that is infeasible for an intruder to break, yet still practical for use. Unfortunately, none of the computational hardness results used in practical cryptosystems have been proved. Thus, the possibility remains that a sufficiently clever intruder could find a way to break the system that does not scale up so rapidly with key size.

Symmetric-key systems of reasonable security are more computationally efficient for the legitimate parties than asymmetric-key systems are. However, giving each potential pair of communicating parties a shared secret in advance is impractical. Thus, many practical systems (such as PGP and SSL) combine the two types of cryptography, using asymmetric-key cryptography to establish a secret key and then switching to symmetric-key cryptography for the bulk of the communication.

The present standard technique for symmetric-key encryption is *AES* (*Advanced Encryption Standard*), also known as *Rijndael*. Many applications still use the prior standard, the *Data Encryption Standard* (*DES*). However, DES is now considered not very secure, simply because the key size is too small. A more secure variant, *3DES*, uses the basic DES operation three times. Best practice for new applications is to use AES.

The most well-known technique for asymmetric-key encryption is the *RSA* system, named for the initials of its three developers, Rivest, Shamir, and Adleman. Data transformed with one half of the RSA key pair can be transformed back to the original using the other half of the key pair; the two specify inverse functions. Thus, a user who wants to receive encrypted messages can make one half the key pair public for any sender to use, while keeping the other half private so that no one else can read the messages.

The standard technique for computing a MAC using a shared secret is known as a *Hashed Message Authentication Code* (*HMAC*). The shared secret and the message are combined together and fed into a *cryptographic hash function*, also known as a *message digest function*. This function is designed so that adversaries cannot realistically hope to find another input that produces the same output. Thus, if the recipient's copy of the message and shared secret result in the same HMAC code, the recipient can be confident that the message is legitimate, because it came from someone else who knew the same two ingredients. As an additional safeguard against certain possible flaws in the cryptographic hash function, the standard HMAC technique (used in IPsec, for example) applies the hash function twice, as shown in Figure 9.12.

HMAC codes are commonly based on one of two cryptographic hash functions, *MD5* (*Message Digest 5*) and *SHA-1* (*Secure Hash Algorithm 1*). Unfortunately, neither of these widely deployed functions turns out to be as secure as previously thought. Unlike DES, which simply used an insufficiently long key, MD5 and SHA-1 have fallen prey to fundamental mathematical progress. The computational problem faced by adversaries does not scale up as rapidly with hash size as had been conjectured, especially for MD5. No one has yet found a way to exploit these functions' vulnerabilities within the context of HMAC. However, the fact that the underlying cryptographic hash functions are weaker than previously thought is worrisome enough that new systems should definitely at a minimum use SHA-1 rather than MD5, as MD5's vulnerabilities are more pronounced. System developers should monitor further news from the cryptography research community and should consider using successors to SHA-1, such as SHA-512. Existing systems using MD5 (particularly in non-HMAC contexts) should be reconsidered, and many of them should be converted to SHA-1 or successor functions with deliberate speed. Practical exploits have been found for MD5's vulnerabilities in some non-HMAC contexts; the same is not currently true for SHA-1.

The most common technique for creating an asymmetric-key MAC combines a cryptographic hash function with the RSA system. These MACs are also known as *digital signatures*, because they share some important features with real signatures, as I will discuss in the next paragraph. First, though, let me explain how they are computed. Recall that each RSA key

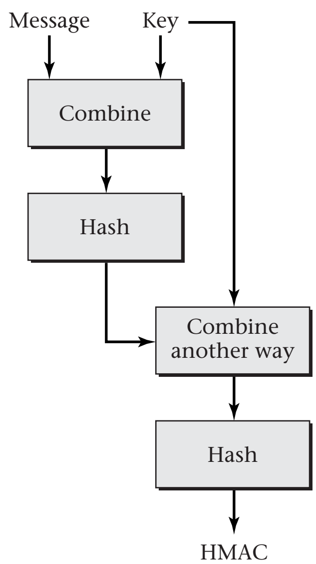


Figure 9.12: An HMAC can be computed as shown here. Both the sender and the receiver use this computation, each with its own copy of the shared secret key. The sender uses the message it is sending and transmits the resulting HMAC along with the message. The receiver does the computation using the (hopefully unchanged) message it received. If all is well, the receiver computes the same HMAC as it received along with the message.

pair specifies a pair of inverse functions. A sender can keep one half the key pair secret, for use in signing messages, and make the other public, for use in checking messages. Call the two inverse functions S and C, for signing and checking, and the cryptographic hash function H. Then a sender can use $S(H(m))$ as a signature for the message m. Any recipient who wants to check this signature runs it through the function C, producing $C(S(H(m)))$, which is the same as $H(m)$, because C is the inverse function of S. The recipient also runs the received message through H, and verifies that the same value of $H(m)$ results. This provides evidence that the message wasn't tampered with, and was signed by the one person who knew S. This system is summarized in Figure 9.13.

The key difference between a digital signature and an HMAC is that the recipient is in no better position to forge a digital signature than anyone else would be. Thus, digital signatures offer the feature known as *non-repudiation*. That is, if you have an embarrassing email signed by me, you could show it to a third party and I couldn't convincingly claim that you forged it yourself. An HMAC, on the other hand, would offer no evidence to a third party regarding which of the two of us wrote the message.

Exercises

9.1 Under the end-to-end principle, which protocol layers are processed by devices within the Internet, excluding the endpoint computers?

9.2 List at least five types of header lines that can be used in the HTTP protocol. What is the function of each?

9.3 What is one reason a domain name might take much longer to resolve the first time it is used than on subsequent uses?

9.4 Figure 9.9 on page 419 illustrates how parity can be used as a primitive form of erasure coding for forward error correction. Show how the third segment could be reconstructed if it were missing. Also, suppose the parity segment were 01100001. What would the missing third data segment then have been?

9.5 TCP and UDP headers contain port numbers used by the transport-layer software to demultiplex incoming data to the appropriate application-layer consumer. What analogous number is in the IP header, allowing the network layer to demultiplex to the appropriate transport-layer consumer?

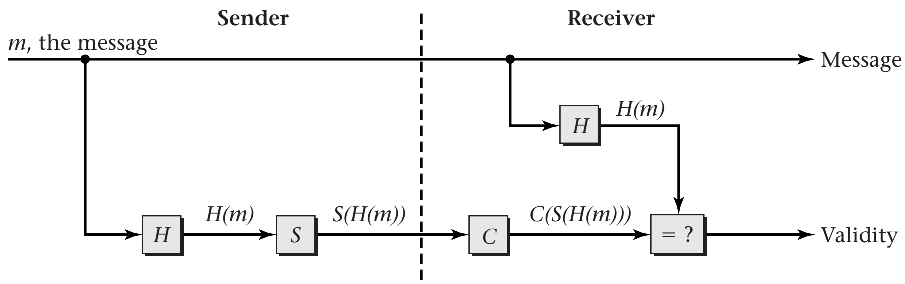


Figure 9.13: A digital signature is computed and verified as shown here. The signing and checking functions S and C are inverses, one kept private and the other publicly known. The role of the cryptographic hash function, H, is simply to efficiently reduce the amount of data that S and C need to process.

9.6 Express the IP address 10100011000101000110000100001111 in dotted decimal.

9.7 Express the IP address 194.79.127.10 in binary.

9.8 If a network mask is 255.255.192.0, how many bits long is the network prefix?

9.9 If a network mask is 255.255.224.0, how many bits long is the network prefix?

9.10 What would the network mask be for a 20-bit network prefix?

9.11 What would the network mask be for a 22-bit network prefix?

9.12 My computer has IP address 138.236.64.64 and mask 255.255.248.0. For which of the following destination IP addresses does my computer send the packet directly, and for which does it send by way of the gateway router?

(a) 138.236.71.64
(b) 138.236.72.64
(c) 138.236.64.72
(d) 216.114.254.180

9.13 Identify by name and number which layer of the OSI reference model corresponds to each of the following specific protocols, technologies, or functions:

(a) UDP
(b) retrieving email
(c) Ethernet MAC addresses
(d) IP
(e) congestion control
(f) fiber optics
(g) TCP
(h) routers
(i) delivering bytes in their proper sequence
(j) DNS
(k) end-to-end flow-control
(l) HTTP
(m) retransmitting data for which no acknowledgment is received
(n) CIFS
(o) verifying that a cached web page is up to date
(p) port numbers
(q) NFS

9.14 Section 9.3.2 explains how TCP recovers from lost data segments, but it doesn't consider lost acknowledgments. Nonetheless, the description in that section is sufficient that you could figure out how lost acknowledgments are tolerated.

(a) Explain why sometimes a lost acknowledgment is tolerated without any additional communication or delay.
(b) Explain why under other circumstances, TCP incurs extra delay and communication in recovering from a lost acknowledgment.

Programming Projects

9.1 Modify the message storage server from Figure 9.7 on page 411 so that each accepted connection is handled in a separate thread, with the main thread immediately looping back around to accept another connection. Using `telnet`, show that a client that is slow providing its line of text does not prevent other clients from being served. Your program should use a synchronized method that atomically stores a newly arrived message and retrieves the previously stored message.

9.2 Write a program that can retrieve a single file from a web server using HTTP, given the hostname of the server and the name of the file on the server. (For example, given *www.gustavus.edu* and `/+max/`, it would retrieve my home page.) You should directly use a socket API, rather than any higher-level mechanism that already implements HTTP for you. Your handling of errors and other exceptional circumstances can be very primitive, at least initially.

9.3 The Java class `java.net.ServerSocket` provides two methods called `getLocalAddress()` and `getInetAddress()` that can be used to find the local and remote internet addresses associated with a connected socket. Write a server program that accepts connection requests and, each time it receives a connection, writes out on the connection a line of text containing the string form of the remote address from which the connection was received. Write a client program that connects to the server, displays the string the server sends, and also displays the client's own local address. Show how you can use this pair of programs to test whether the client is connected to the server through a NAT router or not.

Exploration Projects

9.1 Most UNIX and Linux systems have a program named `dig` that can be used to send DNS queries and show the responses in human-readable form. Use this tool to explore the DNS. For example, find the internet addresses of some of your favorite computers, check whether the CNAME chain leading from *www.gustavus.edu.* still has the same structure as I reported, and use the zone transfer function to display the contents of your local zone. What aspects of the DNS do you find that were not mentioned in my description?

9.2 Use the freely available network packet capture and analysis tool named `wireshark` to study DNS. Capture packets while you are accessing a web site you have not previously accessed. Look at just the DNS packets and for each one expand out the DNS portion of the display. In what regards do you see confirmation of my description of DNS? What aspects of the DNS do you find that were not mentioned in my description?

9.3 Use the freely available network packet capture and analysis tool named `wireshark` to study either CIFS or NFS. Capture packets while you are accessing a file provided by a server. Look at just the CIFS or NFS packets and for each one expand out the CIFS or NFS portion of the display. In what regards do you see confirmation of my description of this system? What aspects of the system do you find that were not mentioned in my description?

9.4 I mentioned that NFS does not have operations for opening or closing files. With regard to closing, this is inarguably true, whereas with regard to opening, my claim might be only a half-truth. The NFS protocol includes a lookup operation, which is similar to a file open. Find information about the NFS lookup operation and explain how looking up a file is similar to and different from opening a file.

9.5 Compile and run the message storage server from Figure 9.7 on page 411. Using `telnet`, show that a client that is slow providing its line of text prevents other clients from being served.

9.6 Find your own IP address and network mask. On a Microsoft Windows system, you can do this with the `ipconfig` command. On most UNIX or Linux systems, you can do this using the `ifconfig` command; a typical example of its use would be `ifconfig eth0`.

From the network mask, how many bits long is your network prefix? Using this information together with the IP address, what is your actual network prefix?

9.7 Explore some of the resources and sample policies on *www.sans.org*. Write a summary of something interesting you find.

9.8 Read the paper by Pang et al. on "Characteristics of Internet Background Radiation," which you can find on the web. Write a summary no longer than one page.

9.9 In comparing CIFS and NFS, I remark that "a stateless design [such as NFS] cannot readily support file locking or keeping client-side file caches up to date." Find out what is hiding behind the word "readily." Does NFS provide any support for these services? If so, how?

9.10 The section on DNS (Section 9.2.2) mentions that type A resource records are used for internet addresses. The section on IP versions (Section 9.4.1) indicates that IPv6 addresses are different from IPv4 addresses. Which version was assumed in the section on DNS? Are the addresses for the other version also held in type A resource records? If not, what type is used?

9.11 In late 2008, a major security vulnerability became known that involved the use of MD5 digital signatures by VeriSign's RapidSSL brand. Research this topic and write a paper that explains the vulnerability and its resolution, making appropriate connections to material within this chapter. Be sure to seek out sources of information that are authoritative and that include technical details.

Notes

Most of the topics in this chapter are covered in more detail in standard networking textbooks, such as the one by Tanenbaum [144] and the one by Kurose and Ross [91]. The ultimate source of information on the various protocols are the corresponding standards documents, which can be found at such sites as *www.rfc-editor.org*. A good compromise, providing almost as much technical detail as the standards and almost as much tutorial clarity as the textbooks, is Stevens's book [138]. If you want to delve into the kernel implementation details, you could look at books on the FreeBSD [102] or Linux [74] implementations. Regarding the frequency of firewall misconfiguration, see the study by Wool [153]. One area of research I mentioned is the use of erasure coding for forward error correction; see, for example, the paper by Byers, Luby, and Mizenmacher [27]. The paper by Pang et al. that serves as the basis for an exploration project is reference [110].