Chapter 6 The Link Layer and LANs

In the previous two chapters we learned that the network layer provides a communication service between *any* two network hosts. Between the two hosts, datagrams travel over a series of communication links, some wired and some wireless, starting at the source host, passing through a series of packet switches (switches and routers) and ending at the destination host. As we continue down the protocol stack, from the network layer to the link layer, we naturally wonder how packets are sent across the *individual links* that make up the end-to-end communication path. How are the network-layer datagrams encapsulated in the link-layer frames for transmission over a single link? Are different link-layer protocols used in the different links along the communication path? How are transmission conflicts in broadcast links resolved? Is there addressing at the link layer and, if so, how does the link-layer addressing operate with the network-layer addressing we learned about in **Chapter 4**? And what exactly is the difference between a switch and a router? We'll answer these and other important questions in this chapter.

In discussing the link layer, we'll see that there are two fundamentally different types of link-layer channels. The first type are broadcast channels, which connect multiple hosts in wireless LANs, satellite networks, and hybrid fiber-coaxial cable (HFC) access networks. Since many hosts are connected to the same broadcast communication channel, a so-called medium access protocol is needed to coordinate frame transmission. In some cases, a central controller may be used to coordinate transmissions; in other cases, the hosts themselves coordinate transmissions. The second type of link-layer channel is the point-to-point communication link, such as that often found between two routers connected by a long-distance link, or between a user's office computer and the nearby Ethernet switch to which it is connected. Coordinating access to a point-to-point link is simpler; the reference material on this book's Web site has a detailed discussion of the Point-to-Point Protocol (PPP), which is used in settings ranging from dial-up service over a telephone line to high-speed point-to-point frame transport over fiber-optic links.

We'll explore several important link-layer concepts and technologies in this chapter. We'll dive deeper into error detection and correction, a topic we touched on briefly in **Chapter 3**. We'll consider multiple access networks and switched LANs, including Ethernet—by far the most prevalent wired LAN technology. We'll also look at virtual LANs, and data center networks. Although WiFi, and more generally wireless LANs, are link-layer topics, we'll postpone our study of these important topics until

Chapter 7.

6.1 Introduction to the Link Layer

Let's begin with some important terminology. We'll find it convenient in this chapter to refer to any device that runs a link-layer (i.e., layer 2) protocol as a **node**. Nodes include hosts, routers, switches, and WiFi access points (discussed in **Chapter 7**). We will also refer to the communication channels that connect adjacent nodes along the communication path as **links**. In order for a datagram to be transferred from source host to destination host, it must be moved over each of the *individual links* in the end-to-end path. As an example, in the company network shown at the bottom of **Figure 6.1**, consider sending a datagram from one of the wireless hosts to one of the servers. This datagram will actually pass through six links: a WiFi link between sending host and WiFi access point, an Ethernet link between the access point and a link-layer switch; a link between the link-layer switch and the router, a link between the two routers; an Ethernet link between the router and a link-layer switch; and finally an Ethernet link between the switch and the server. Over a given link, a transmitting node encapsulates the datagram in a **link-layer frame** and transmits the frame into the link.

In order to gain further insight into the link layer and how it relates to the network layer, let's consider a transportation analogy. Consider a travel agent who is planning a trip for a tourist traveling from Princeton, New Jersey, to Lausanne, Switzerland. The travel agent decides that it is most convenient for the tourist to take a limousine from Princeton to JFK airport, then a plane from JFK airport to Geneva's airport, and finally a train from Geneva's airport to Lausanne's train station. Once the travel agent makes the three reservations, it is the responsibility of the Princeton limousine company to get the tourist from Princeton to JFK; it is the responsibility of the airline company to get the tourist from JFK to Geneva; and it is the responsibility

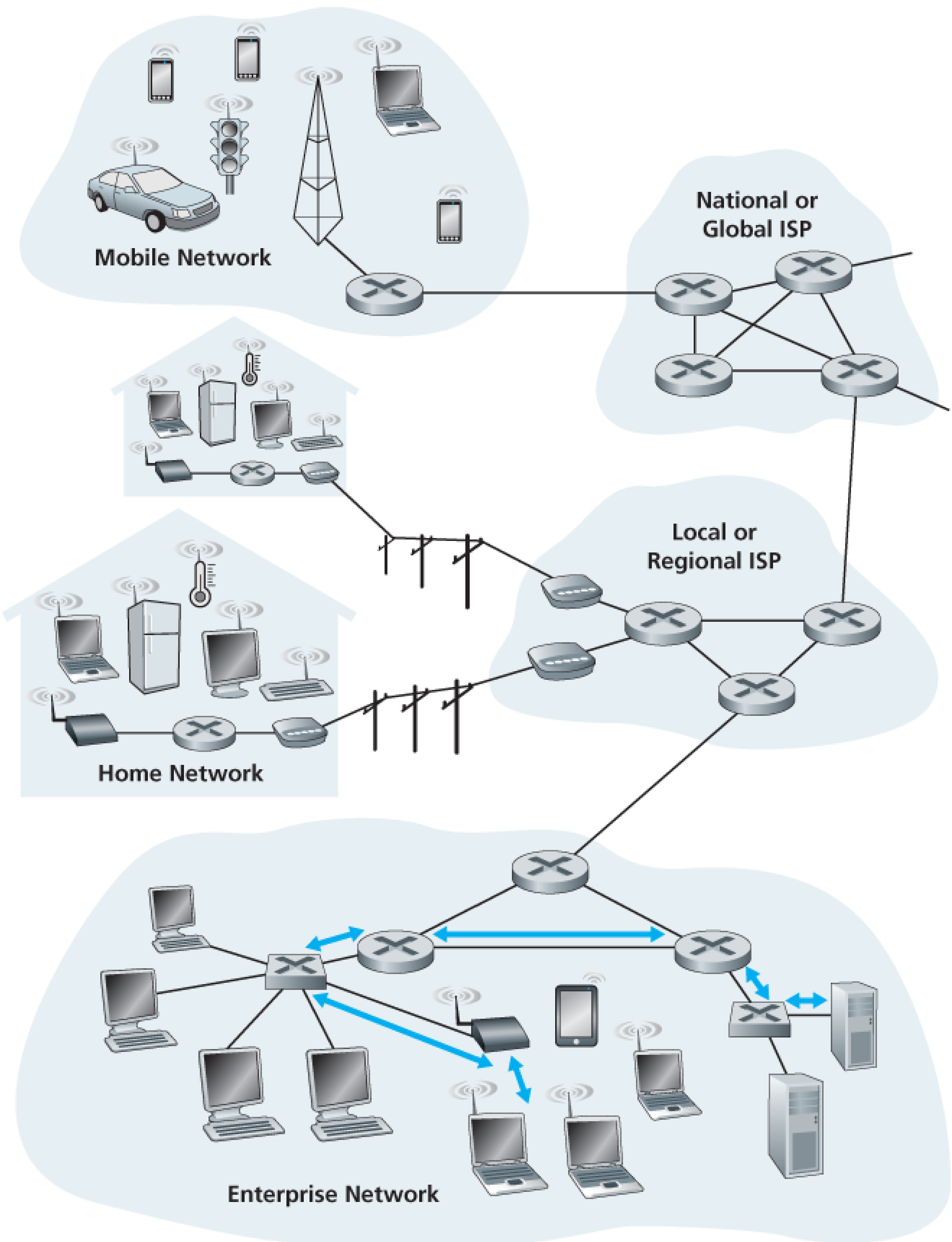


Figure 6.1 Six link-layer hops between wireless host and server

of the Swiss train service to get the tourist from Geneva to Lausanne. Each of the three segments of the trip is “direct” between two “adjacent” locations. Note that the three transportation segments are managed by different companies and use entirely different transportation modes (limousine, plane, and train). Although the transportation modes are different, they each provide the basic service of moving passengers from one location to an adjacent location. In this transportation analogy, the tourist is a datagram, each transportation segment is a link, the transportation mode is a link-layer protocol, and the travel agent is a routing protocol.

6.1.1 The Services Provided by the Link Layer

Although the basic service of any link layer is to move a datagram from one node to an adjacent node over a single communication link, the details of the provided service can vary from one link-layer protocol to the next. Possible services that can be offered by a link-layer protocol include:

- **Framing.** Almost all link-layer protocols encapsulate each network-layer datagram within a link-layer frame before transmission over the link. A frame consists of a data field, in which the network-layer datagram is inserted, and a number of header fields. The structure of the frame is specified by the link-layer protocol. We'll see several different frame formats when we examine specific link-layer protocols in the second half of this chapter.
- **Link access.** A medium access control (MAC) protocol specifies the rules by which a frame is transmitted onto the link. For point-to-point links that have a single sender at one end of the link and a single receiver at the other end of the link, the MAC protocol is simple (or nonexistent)—the sender can send a frame whenever the link is idle. The more interesting case is when multiple nodes share a single broadcast link—the so-called multiple access problem. Here, the MAC protocol serves to coordinate the frame transmissions of the many nodes.
- **Reliable delivery.** When a link-layer protocol provides reliable delivery service, it guarantees to move each network-layer datagram across the link without error. Recall that certain transport-layer protocols (such as TCP) also provide a reliable delivery service. Similar to a transport-layer reliable delivery service, a link-layer reliable delivery service can be achieved with acknowledgments and retransmissions (see **Section 3.4**). A link-layer reliable delivery service is often used for links that are prone to high error rates, such as a wireless link, with the goal of correcting an error locally—on the link where the error occurs—rather than forcing an end-to-end retransmission of the data by a transport- or application-layer protocol. However, link-layer reliable delivery can be considered an unnecessary overhead for low bit-error links, including fiber, coax, and many twisted-pair copper links. For this reason, many wired link-layer protocols do not provide a reliable delivery service.
- **Error detection and correction.** The link-layer hardware in a receiving node can incorrectly decide that a bit in a frame is zero when it was transmitted as a one, and vice versa. Such bit errors are introduced by signal attenuation and electromagnetic noise. Because there is no need to forward a datagram that has an error, many link-layer protocols provide a mechanism to detect such bit errors. This is done by having the transmitting node include error-detection bits in the frame, and having the receiving node perform an error check. Recall from **Chapters 3** and **4** that the Internet's transport layer and network layer also provide a limited form of error detection—the Internet checksum. Error detection in the link layer is usually more sophisticated and is implemented in hardware. Error correction is similar to error detection, except that a receiver not only detects when bit errors have occurred in the frame but also determines exactly where in the frame the errors have occurred (and

then corrects these errors).

6.1.2 Where Is the Link Layer Implemented?

Before diving into our detailed study of the link layer, let's conclude this introduction by considering the question of where the link layer is implemented. We'll focus here on an end system, since we learned in **Chapter 4** that the link layer is implemented in a router's line card. Is a host's link layer implemented in hardware or software? Is it implemented on a separate card or chip, and how does it interface with the rest of a host's hardware and operating system components?

Figure 6.2 shows a typical host architecture. For the most part, the link layer is implemented in a **network adapter**, also sometimes known as a **network interface card (NIC)**. At the heart of the network adapter is the link-layer controller, usually a single, special-purpose chip that implements many of the link-layer services (framing, link access, error detection, and so on). Thus, much of a link-layer controller's functionality is implemented in hardware. For example, Intel's 710 adapter **[Intel 2016]** implements the Ethernet protocols we'll study in **Section 6.5**; the Atheros AR5006 **[Atheros 2016]** controller implements the 802.11 WiFi protocols we'll study in **Chapter 7**. Until the late 1990s, most network adapters were physically separate cards (such as a PCMCIA card or a plug-in card fitting into a PC's PCI card slot) but increasingly, network adapters are being integrated onto the host's motherboard —a so-called LAN-on-motherboard configuration.

On the sending side, the controller takes a datagram that has been created and stored in host memory by the higher layers of the protocol stack, encapsulates the datagram in a link-layer frame (filling in the frame's various fields), and then transmits the frame into the communication link, following the link-access protocol. On the receiving side, a controller receives the entire frame, and extracts the network-layer datagram. If the link layer performs error detection, then it is the sending controller that sets the error-detection bits in the frame header and it is the receiving controller that performs error detection.

Figure 6.2 shows a network adapter attaching to a host's bus (e.g., a PCI or PCI-X bus), where it looks much like any other I/O device to the other host

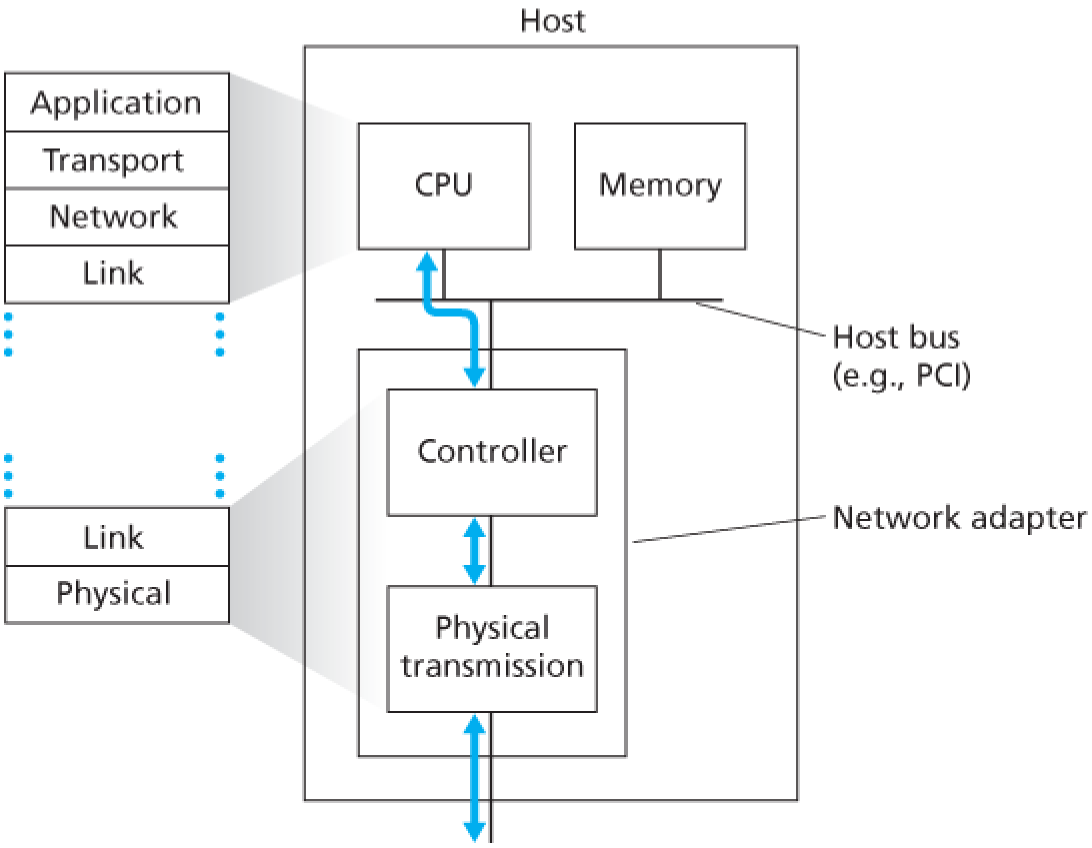


Figure 6.2 Network adapter: Its relationship to other host components and to protocol stack functionality

components. Figure 6.2 also shows that while most of the link layer is implemented in hardware, part of the link layer is implemented in software that runs on the host's CPU. The software components of the link layer implement higher-level link-layer functionality such as assembling link-layer addressing information and activating the controller hardware. On the receiving side, link-layer software responds to controller interrupts (e.g., due to the receipt of one or more frames), handling error conditions and passing a datagram up to the network layer. Thus, the link layer is a combination of hardware and software—the place in the protocol stack where software meets hardware. [Intel 2016] provides a readable overview (as well as a detailed description) of the XL710 controller from a software-programming point of view.

6.2 Error-Detection and -Correction Techniques

In the previous section, we noted that **bit-level error detection and correction**—detecting and correcting the corruption of bits in a link-layer frame sent from one node to another physically connected neighboring node—are two services often provided by the link layer. We saw in **Chapter 3** that error-detection and -correction services are also often offered at the transport layer as well. In this section, we'll examine a few of the simplest techniques that can be used to detect and, in some cases, correct such bit errors. A full treatment of the theory and implementation of this topic is itself the topic of many textbooks (for example, **[Schwartz 1980]** or **[Bertsekas 1991]**), and our treatment here is necessarily brief. Our goal here is to develop an intuitive feel for the capabilities that error-detection and -correction techniques provide and to see how a few simple techniques work and are used in practice in the link layer.

Figure 6.3 illustrates the setting for our study. At the sending node, data, *D*, to be protected against bit errors is augmented with error-detection and -correction bits (*EDC*). Typically, the data to be protected includes not only the datagram passed down from the network layer for transmission across the link, but also link-level addressing information, sequence numbers, and other fields in the link frame header. Both *D* and *EDC* are sent to the receiving node in a link-level frame. At the receiving node, a sequence of bits, *D′* and *EDC′* is received. Note that *D′* and *EDC′* may differ from the original *D* and *EDC* as a result of in-transit bit flips.

The receiver's challenge is to determine whether or not *D′* is the same as the original *D*, given that it has only received *D′* and *EDC′*. The exact wording of the receiver's decision in **Figure 6.3** (we ask whether an error is detected, not whether an error has occurred!) is important. Error-detection and -correction techniques allow the receiver to sometimes, *but not always*, detect that bit errors have occurred. Even with the use of error-detection bits there still may be **undetected bit errors**; that is, the receiver may be unaware that the received information contains bit errors. As a

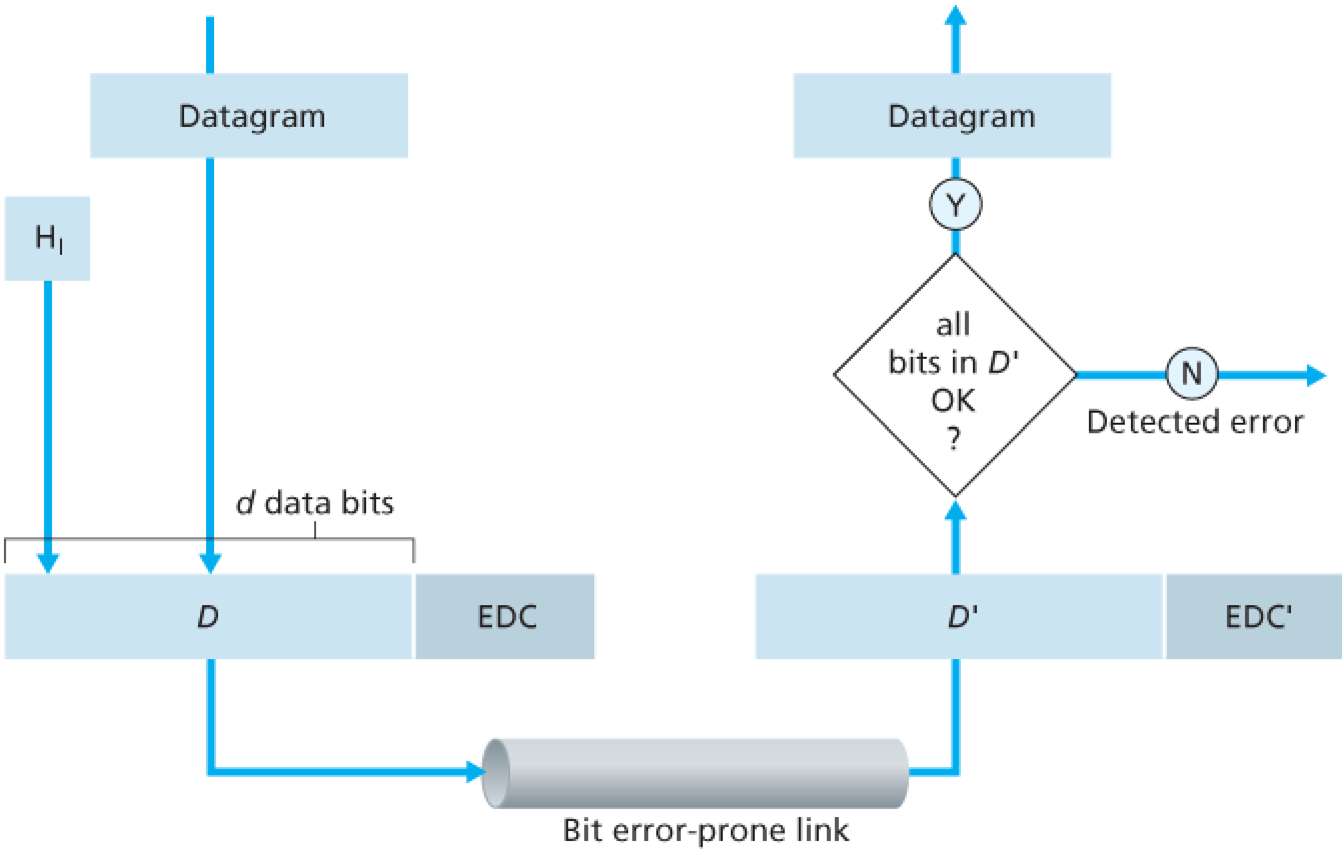


Figure 6.3 Error-detection and -correction scenario

consequence, the receiver might deliver a corrupted datagram to the network layer, or be unaware that the contents of a field in the frame's header has been corrupted. We thus want to choose an error-detection scheme that keeps the probability of such occurrences small. Generally, more sophisticated error-detection and-correction techniques (that is, those that have a smaller probability of allowing undetected bit errors) incur a larger overhead—more computation is needed to compute and transmit a larger number of error-detection and -correction bits.

Let's now examine three techniques for detecting errors in the transmitted data—parity checks (to illustrate the basic ideas behind error detection and correction), checksumming methods (which are more typically used in the transport layer), and cyclic redundancy checks (which are more typically used in the link layer in an adapter).

6.2.1 Parity Checks

Perhaps the simplest form of error detection is the use of a single **parity bit**. Suppose that the information to be sent, *D* in **Figure 6.4**, has *d* bits. In an even parity scheme, the sender simply includes one additional bit and chooses its value such that the total number of 1s in the d+1 bits (the original information plus a parity bit) is even. For odd parity schemes, the parity bit value is chosen such that there is an odd number of 1s. **Figure 6.4** illustrates an even parity scheme, with the single parity bit being stored in a separate field.

Receiver operation is also simple with a single parity bit. The receiver need only count the number of 1s in the received d+1 bits. If an odd number of 1-valued bits are found with an even parity scheme, the receiver knows that at least one bit error has occurred. More precisely, it knows that some *odd* number of bit errors have occurred.

But what happens if an even number of bit errors occur? You should convince yourself that this would result in an undetected error. If the probability of bit errors is small and errors can be assumed to occur independently from one bit to the next, the probability of multiple bit errors in a packet would be extremely small. In this case, a single parity bit might suffice. However, measurements have shown that, rather than occurring independently, errors are often clustered together in “bursts.” Under burst error conditions, the probability of undetected errors in a frame protected by single-bit parity can approach 50 percent **[Spragins 1991]**. Clearly, a more robust error-detection scheme is needed (and, fortunately, is used in practice!). But before examining error-detection schemes that are used in practice, let’s consider a simple

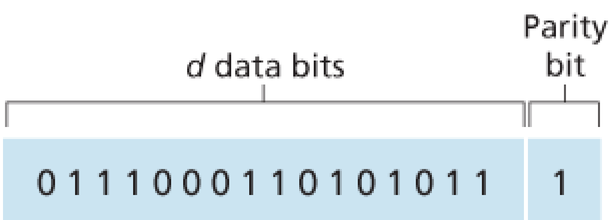


Figure 6.4 One-bit even parity

generalization of one-bit parity that will provide us with insight into error-correction techniques.

Figure 6.5 shows a two-dimensional generalization of the single-bit parity scheme. Here, the *d* bits in *D* are divided into *i* rows and *j* columns. A parity value is computed for each row and for each column. The resulting i+j+1 parity bits comprise the link-layer frame’s error-detection bits.

Suppose now that a single bit error occurs in the original *d* bits of information. With this **two-dimensional parity** scheme, the parity of both the column and the row containing the flipped bit will be in error. The receiver can thus not only *detect* the fact that a single bit error has occurred, but can use the column and row indices of the column and row with parity errors to actually identify the bit that was corrupted and *correct* that error! **Figure 6.5** shows an example in which the 1-valued bit in position (2,2) is corrupted and switched to a 0—an error that is both detectable and correctable at the receiver. Although our discussion has focused on the original *d* bits of information, a single error in the parity bits themselves is also detectable and correctable. Two-dimensional parity can also detect (but not correct!) any combination of two errors in a packet. Other properties of the two-dimensional parity scheme are explored in the problems at the end of the chapter.

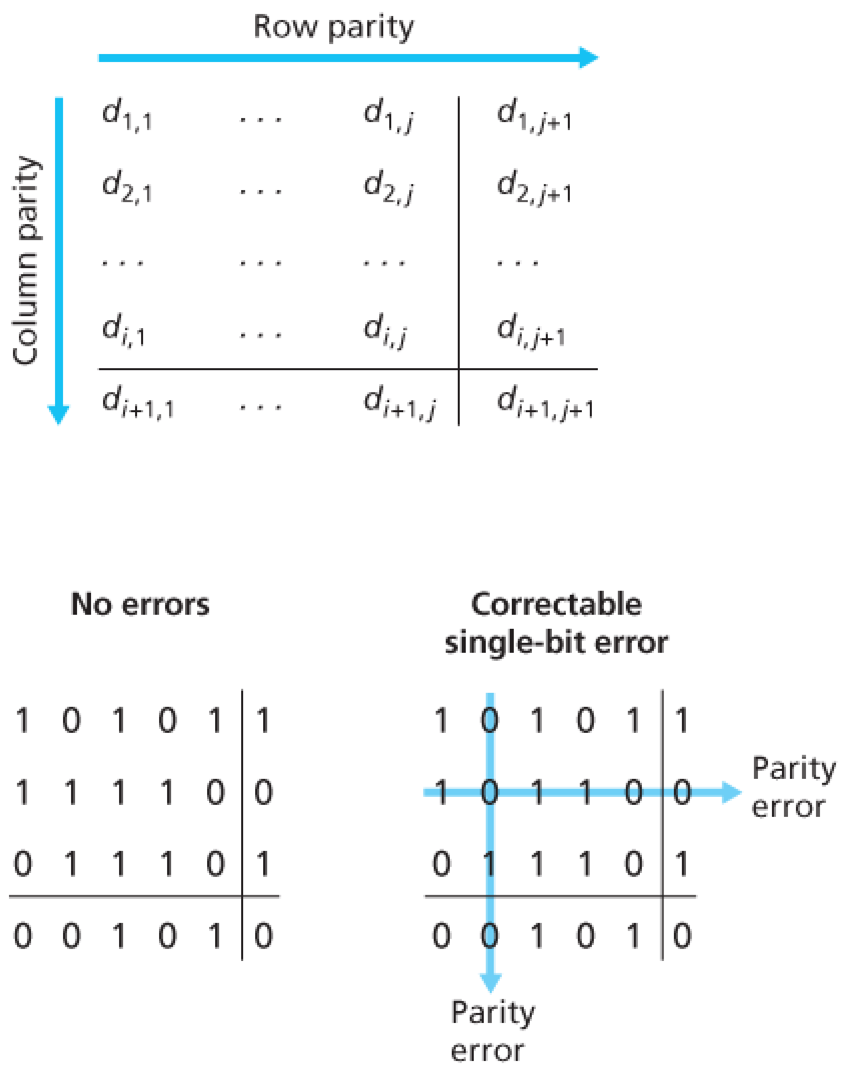


Figure 6.5 Two-dimensional even parity

The ability of the receiver to both detect and correct errors is known as **forward error correction (FEC)**. These techniques are commonly used in audio storage and playback devices such as audio CDs. In a network setting, FEC techniques can be used by themselves, or in conjunction with link-layer ARQ techniques similar to those we examined in **Chapter 3**. FEC techniques are valuable because they can decrease the number of sender retransmissions required. Perhaps more important, they allow for immediate correction of errors at the receiver. This avoids having to wait for the round-trip propagation delay needed for the sender to receive a NAK packet and for the retransmitted packet to propagate back to the receiver—a potentially important advantage for real-time network applications **[Rubenstein 1998]** or links (such as deep-space links) with long propagation delays. Research examining the use of FEC in error-control protocols includes **[Biersack 1992**; **Nonnenmacher 1998**; **Byers 1998**; **Shacham 1990]**.

6.2.2 Checksumming Methods

In checksumming techniques, the *d* bits of data in **Figure 6.4** are treated as a sequence of *k*-bit integers. One simple checksumming method is to simply sum these *k*-bit integers and use the resulting sum as the error-detection bits. The **Internet checksum** is based on this approach—bytes of data are treated as 16-bit integers and summed. The 1s complement of this sum then forms the Internet checksum that is carried in the segment header. As discussed in **Section 3.3**, the receiver checks the checksum by taking the 1s complement of the sum of the received data (including the checksum) and checking whether the result is all 1 bits. If any of the bits are 0, an error is indicated. RFC 1071 discusses the Internet checksum algorithm and its implementation in detail. In the TCP and UDP protocols, the Internet checksum is computed over all fields (header and data fields included). In IP the checksum is computed over the IP header (since the UDP or TCP segment has its own checksum). In other protocols, for example, XTP **[Strayer 1992]**, one checksum is computed over the header and another checksum is computed over the entire packet.

Checksumming methods require relatively little packet overhead. For example, the checksums in TCP and UDP use only 16 bits. However, they provide relatively weak protection against errors as compared with cyclic redundancy check, which is discussed below and which is often used in the link layer. A natural question at this point is, Why is checksumming used at the transport layer and cyclic redundancy check used at the link layer? Recall that the transport layer is typically implemented in software in a host as part of the host's operating system. Because transport-layer error detection is implemented in software, it is important to have a simple and fast error-detection scheme such as checksumming. On the other hand, error detection at the link layer is implemented in dedicated hardware in adapters, which can rapidly perform the more complex CRC operations. Feldmeier **[Feldmeier 1995]** presents fast software implementation techniques for not only weighted checksum codes, but CRC (see below) and other codes as well.

6.2.3 Cyclic Redundancy Check (CRC)

An error-detection technique used widely in today's computer networks is based on **cyclic redundancy check (CRC) codes**. CRC codes are also known as **polynomial codes**, since it is possible to view the bit string to be sent as a polynomial whose coefficients are the 0 and 1 values in the bit string, with operations on the bit string interpreted as polynomial arithmetic.

CRC codes operate as follows. Consider the *d*-bit piece of data, *D*, that the sending node wants to send to the receiving node. The sender and receiver must first agree on an r+1 bit pattern, known as a **generator**, which we will denote as *G*. We will require that the most significant (leftmost) bit of *G* be a 1. The key idea behind CRC codes is shown in **Figure 6.6**. For a given piece of data, *D*, the sender will choose *r* additional bits, *R*, and append them to *D* such that the resulting $d+r$ bit pattern (interpreted as a binary number) is exactly divisible by *G* (i.e., has no remainder) using modulo-2 arithmetic. The process of error checking with CRCs is thus simple: The receiver divides the $d+r$ received bits by *G*. If the remainder is nonzero, the receiver knows that an error has occurred; otherwise the data is accepted as being correct.

All CRC calculations are done in modulo-2 arithmetic without carries in addition or borrows in subtraction. This means that addition and subtraction are identical, and both are equivalent to the bitwise exclusive-or (XOR) of the operands. Thus, for example,

```
1011 XOR 0101 = 1110
1001 XOR 1101 = 0100
```

Also, we similarly have

```
1011 - 0101 = 1110
1001 - 1101 = 0100
```

Multiplication and division are the same as in base-2 arithmetic, except that any required addition or subtraction is done without carries or borrows. As in regular

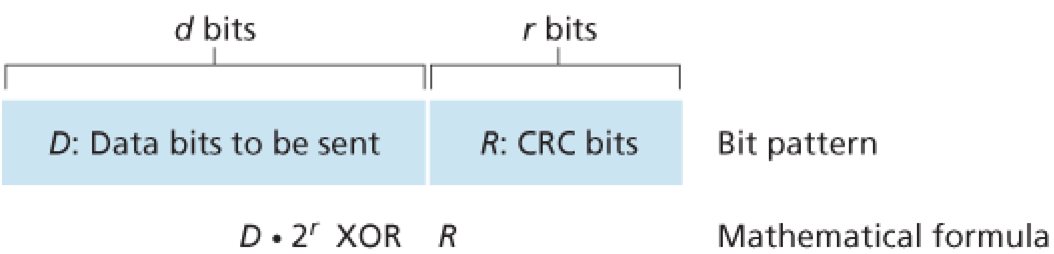


Figure 6.6 CRC

binary arithmetic, multiplication by 2^k left shifts a bit pattern by *k* places. Thus, given *D* and *R*, the quantity $D\cdot 2^r \text{XOR } R$ yields the $d+r$ bit pattern shown in **Figure 6.6**. We'll use this algebraic characterization of the $d+r$ bit pattern from **Figure 6.6** in our discussion below.

Let us now turn to the crucial question of how the sender computes *R*. Recall that we want to find *R* such that there is an *n* such that

$$D\cdot 2^r \text{XOR } R = nG$$

That is, we want to choose *R* such that *G* divides into $D\cdot 2^r \text{XOR } R$ without remainder. If we XOR (that is, add modulo-2, without carry) *R* to both sides of the above equation, we get

$D\cdot 2r=nG \text{ XOR } R$

This equation tells us that if we divide $D\cdot 2r$ by G, the value of the remainder is precisely *R*. In other words, we can calculate *R* as

$R=\text{remainder}D\cdot 2rG$

Figure 6.7 illustrates this calculation for the case of $D=101110$, $d=6$, $G=1001$, and $r=3$. The 9 bits transmitted in this case are 101 110 011. You should check these calculations for yourself and also check that indeed $D\cdot 2r=101011\cdot G \text{ XOR } R$.

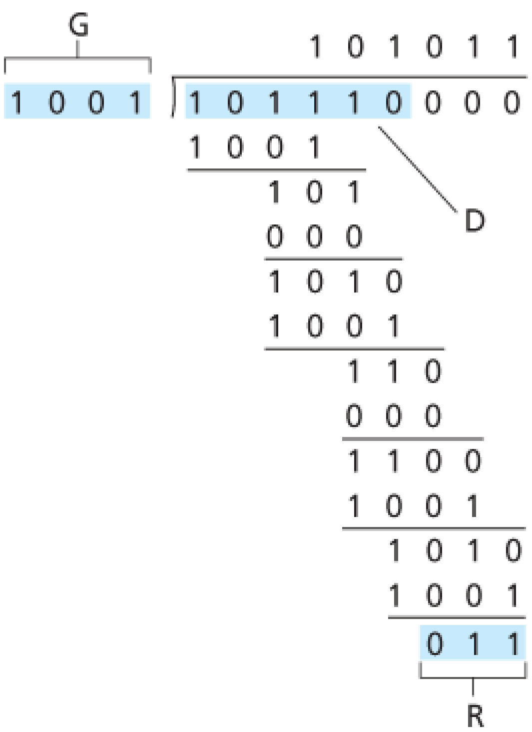


Figure 6.7 A sample CRC calculation

International standards have been defined for 8-, 12-, 16-, and 32-bit generators, *G*. The CRC-32 32-bit standard, which has been adopted in a number of link-level IEEE protocols, uses a generator of

$G_{CRC-32}=100000100110000010001110110110111$

Each of the CRC standards can detect burst errors of fewer than r+1 bits. (This means that all consecutive bit errors of *r* bits or fewer will be detected.) Furthermore, under appropriate assumptions, a burst of length greater than r+1 bits is detected with probability 1−0.5r. Also, each of the CRC standards can detect any odd number of bit errors. See [Williams 1993] for a discussion of implementing CRC checks. The theory behind CRC codes and even more powerful codes is beyond the scope of this text. The text **[Schwartz 1980]** provides an excellent introduction to this topic.

6.3 Multiple Access Links and Protocols

In the introduction to this chapter, we noted that there are two types of network links: point-to-point links and broadcast links. A **point-to-point link** consists of a single sender at one end of the link and a single receiver at the other end of the link. Many link-layer protocols have been designed for point-to-point links; the point-to-point protocol (PPP) and high-level data link control (HDLC) are two such protocols. The second type of link, a **broadcast link**, can have multiple sending and receiving nodes all connected to the same, single, shared broadcast channel. The term *broadcast* is used here because when any one node transmits a frame, the channel broadcasts the frame and each of the other nodes receives a copy. Ethernet and wireless LANs are examples of broadcast link-layer technologies. In this section we'll take a step back from specific link-layer protocols and first examine a problem of central importance to the link layer: how to coordinate the access of multiple sending and receiving nodes to a shared broadcast channel—the **multiple access problem**. Broadcast channels are often used in LANs, networks that are geographically concentrated in a single building (or on a corporate or university campus). Thus, we'll look at how multiple access channels are used in LANs at the end of this section.

We are all familiar with the notion of broadcasting—television has been using it since its invention. But traditional television is a one-way broadcast (that is, one fixed node transmitting to many receiving nodes), while nodes on a computer network broadcast channel can both send and receive. Perhaps a more apt human analogy for a broadcast channel is a cocktail party, where many people gather in a large room (the air providing the broadcast medium) to talk and listen. A second good analogy is something many readers will be familiar with—a classroom—where teacher(s) and student(s) similarly share the same, single, broadcast medium. A central problem in both scenarios is that of determining who gets to talk (that is, transmit into the channel) and when. As humans, we've evolved an elaborate set of protocols for sharing the broadcast channel:

"Give everyone a chance to speak."

"Don't speak until you are spoken to."

"Don't monopolize the conversation."

"Raise your hand if you have a question."

"Don't interrupt when someone is speaking."

"Don't fall asleep when someone is talking."

Computer networks similarly have protocols—so-called **multiple access protocols**—by which nodes regulate their transmission into the shared broadcast channel. As shown in **Figure 6.8**, multiple access protocols are needed in a wide variety of network settings, including both wired and wireless access networks, and satellite networks. Although technically each node accesses the broadcast channel through its adapter, in this section we will refer to the *node* as the sending and

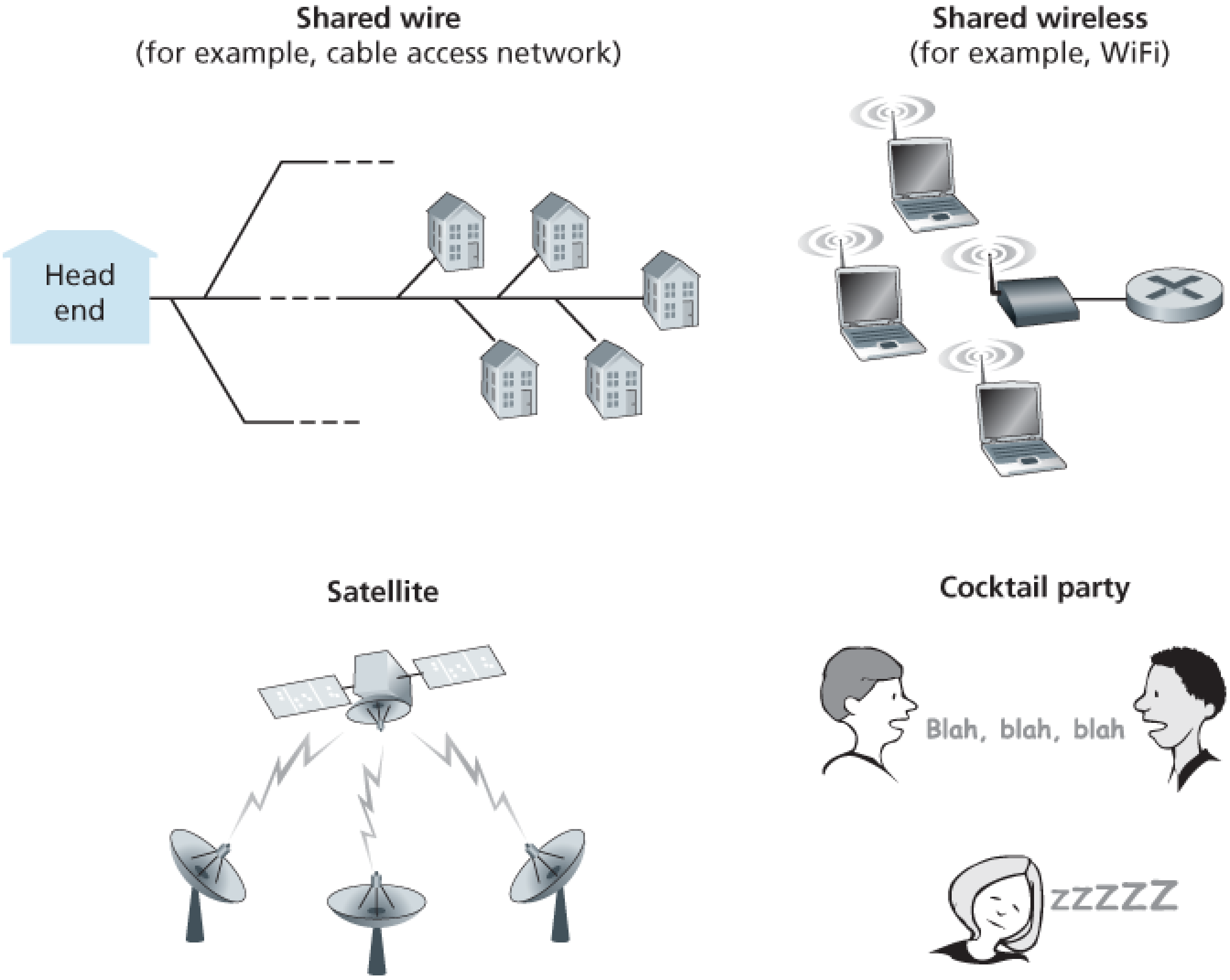


Figure 6.8 Various multiple access channels

receiving device. In practice, hundreds or even thousands of nodes can directly communicate over a broadcast channel.

Because all nodes are capable of transmitting frames, more than two nodes can transmit frames at the same time. When this happens, all of the nodes receive multiple frames at the same time; that is, the transmitted frames **collide** at all of the receivers. Typically, when there is a collision, none of the receiving nodes can make any sense of any of the frames that were transmitted; in a sense, the signals of the colliding frames become inextricably tangled together. Thus, all the frames involved in the collision are lost, and the broadcast channel is wasted during the collision interval. Clearly, if many nodes want to transmit frames frequently, many transmissions will result in collisions, and much of the bandwidth of the broadcast channel will be wasted.

In order to ensure that the broadcast channel performs useful work when multiple nodes are active, it is necessary to somehow coordinate the transmissions of the active nodes. This coordination job is the responsibility of the multiple access protocol. Over the past 40 years, thousands of papers and hundreds of PhD dissertations have been written on multiple access protocols; a comprehensive survey of the first 20 years of this body of work is **[Rom 1990]**. Furthermore, active research in multiple access protocols continues due to the continued emergence of new types of links, particularly new wireless links.

Over the years, dozens of multiple access protocols have been implemented in a variety of link-layer technologies. Nevertheless, we can classify just about any multiple access protocol as belonging to one of three categories: **channel partitioning protocols**, **random access protocols**, and **taking-turns protocols**. We'll cover these categories of multiple access protocols in the following three subsections.

Let's conclude this overview by noting that, ideally, a multiple access protocol for a broadcast channel of rate R bits per second should have the following desirable characteristics:

1. When only one node has data to send, that node has a throughput of R bps.
2. When M nodes have data to send, each of these nodes has a throughput of R/M bps. This need not necessarily imply that each of the M nodes always has an instantaneous rate of R/M, but rather that each node should have an average transmission rate of R/M over some suitably defined interval of time.
3. The protocol is decentralized; that is, there is no master node that represents a single point of failure for the network.
4. The protocol is simple, so that it is inexpensive to implement.

6.3.1 Channel Partitioning Protocols

Recall from our early discussion back in **Section 1.3** that time-division multiplexing (TDM) and frequency-division multiplexing (FDM) are two techniques that can

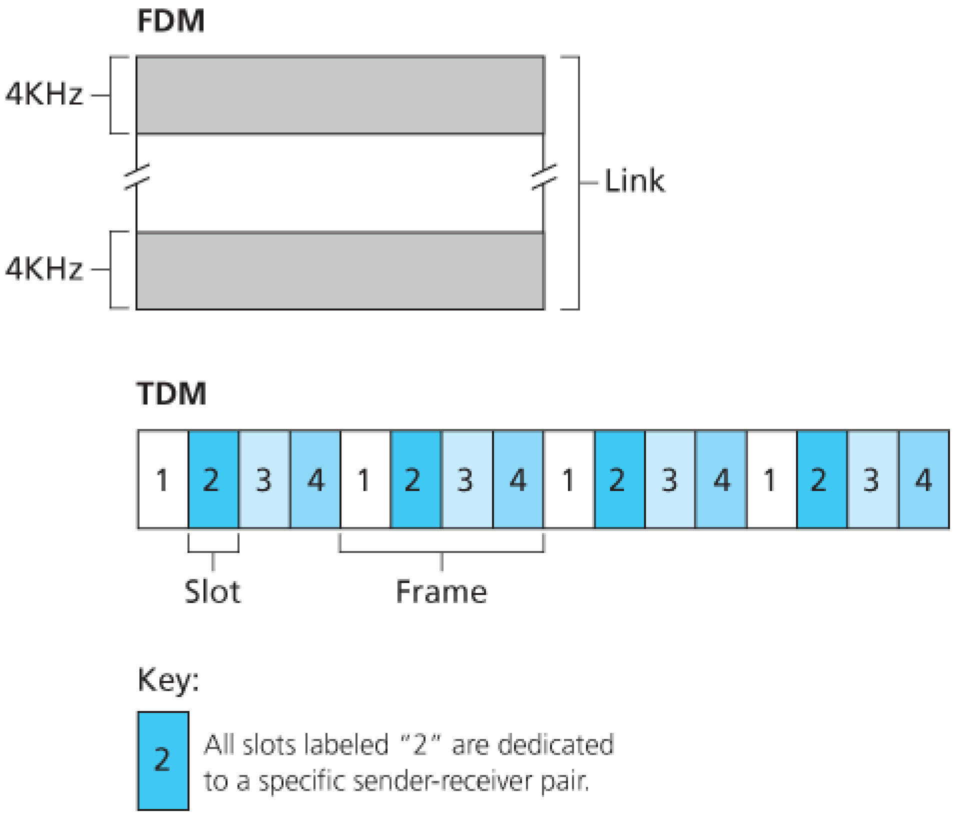


Figure 6.9 A four-node TDM and FDM example

be used to partition a broadcast channel's bandwidth among all nodes sharing that channel. As an example, suppose the channel supports *N* nodes and that the transmission rate of the channel is *R* bps. TDM divides time into **time frames** and further divides each time frame into *N* **time slots**. (The TDM time frame should not be confused with the link-layer unit of data exchanged between sending and receiving adapters, which is also called a frame. In order to reduce confusion, in this subsection we'll refer to the link-layer unit of data exchanged as a packet.) Each time slot is then assigned to one of the *N* nodes. Whenever a node has a packet to send, it transmits the packet's bits during its assigned time slot in the revolving TDM frame. Typically, slot sizes are chosen so that a single packet can be transmitted during a slot time. **Figure 6.9** shows a simple four-node TDM example. Returning to our cocktail party analogy, a TDM-regulated cocktail party would allow one partygoer to speak for a fixed period of time, then allow another partygoer to speak for the same amount of time, and so on. Once everyone had had a chance to talk, the pattern would repeat.

TDM is appealing because it eliminates collisions and is perfectly fair: Each node gets a dedicated transmission rate of *R*/*N* bps during each frame time. However, it has two major drawbacks. First, a node is limited to an average rate of *R*/*N* bps even when it is the only node with packets to send. A second drawback is that a node must always wait for its turn in the transmission sequence—again, even when it is the only node with a frame to send. Imagine the partygoer who is the only one with anything to say (and imagine that this is the even rarer circumstance where everyone wants to hear what that one person has to say). Clearly, TDM would be a poor choice for a multiple access protocol for this particular party.

While TDM shares the broadcast channel in time, FDM divides the *R* bps channel into different frequencies (each with a bandwidth of *R*/*N*) and assigns each frequency to one of the *N* nodes. FDM thus creates *N* smaller channels of *R*/*N* bps out of the single, larger *R* bps channel. FDM shares both the advantages and drawbacks of TDM. It avoids collisions and divides the bandwidth fairly among the *N* nodes. However, FDM also shares a principal disadvantage with TDM—a node is limited to a bandwidth of *R*/*N*, even when it is the only node with packets to send.

A third channel partitioning protocol is **code division multiple access (CDMA)**. While TDM and FDM assign time slots and frequencies, respectively, to the nodes, CDMA assigns a different *code* to each node. Each node then uses its unique code to encode the data bits it sends. If the codes are chosen carefully, CDMA networks have the wonderful property that different nodes can transmit *simultaneously* and yet have their respective receivers correctly receive a sender's encoded data bits (assuming the receiver knows the sender's code) in spite of interfering transmissions by other nodes. CDMA has been used in military systems for some time (due to its anti-jamming properties) and now has widespread civilian use, particularly in cellular telephony. Because CDMA's use is so tightly tied to wireless channels, we'll save our discussion of the technical details of CDMA until **Chapter 7**. For now, it will suffice to know that CDMA codes, like time slots in TDM and frequencies in FDM, can be allocated to the multiple access channel users.

6.3.2 Random Access Protocols

The second broad class of multiple access protocols are random access protocols. In a random access protocol, a transmitting node always transmits at the full rate of the channel, namely, *R* bps. When there is a collision, each node involved in the collision repeatedly retransmits its frame (that is, packet) until its frame gets through without a collision. But when a node experiences a collision, it doesn't necessarily retransmit the frame right away. *Instead it waits a random delay before retransmitting the frame*. Each node involved in a collision chooses independent random delays. Because the random delays are independently chosen, it is possible that one of the nodes will pick a delay that is sufficiently less than the delays of the other colliding nodes and will therefore be able to sneak its frame into the channel without a collision.

There are dozens if not hundreds of random access protocols described in the literature **[Rom 1990**; **Bertsekas 1991]**. In this section we'll describe a few of the most commonly used random access protocols—the ALOHA protocols **[Abramson 1970**; **Abramson 1985**; **Abramson 2009]** and the carrier sense multiple access (CSMA) protocols **[Kleinrock 1975b]**. Ethernet **[Metcalfe 1976]** is a popular and widely deployed CSMA protocol.

Slotted ALOHA

Let's begin our study of random access protocols with one of the simplest random access protocols, the slotted ALOHA protocol. In our description of slotted ALOHA, we assume the following:

- All frames consist of exactly L bits.
- Time is divided into slots of size L/R seconds (that is, a slot equals the time to transmit one frame).
- Nodes start to transmit frames only at the beginnings of slots.
- The nodes are synchronized so that each node knows when the slots begin.
- If two or more frames collide in a slot, then all the nodes detect the collision event before the slot ends.

Let p be a probability, that is, a number between 0 and 1. The operation of slotted ALOHA in each node is simple:

- When the node has a fresh frame to send, it waits until the beginning of the next slot and transmits the entire frame in the slot.
- If there isn't a collision, the node has successfully transmitted its frame and thus need not consider retransmitting the frame. (The node can prepare a new frame for transmission, if it has one.)
- If there is a collision, the node detects the collision before the end of the slot. The node retransmits its frame in each subsequent slot with probability p until the frame is transmitted without a collision.

By retransmitting with probability p, we mean that the node effectively tosses a biased coin; the event heads corresponds to "retransmit," which occurs with probability p. The event tails corresponds to "skip the slot and toss the coin again in the next slot"; this occurs with probability $(1-p)$. All nodes involved in the collision toss their coins independently.

Slotted ALOHA would appear to have many advantages. Unlike channel partitioning, slotted ALOHA allows a node to transmit continuously at the full rate, R, when that node is the only active node. (A node is said to be active if it has frames to send.) Slotted ALOHA is also highly decentralized, because each node detects collisions and independently decides when to retransmit. (Slotted ALOHA does, however, require the slots to be synchronized in the nodes; shortly we'll discuss an unslotted version of the ALOHA protocol, as well as CSMA protocols, none of which require such synchronization.) Slotted ALOHA is also an extremely simple protocol.

Slotted ALOHA works well when there is only one active node, but how efficient is it when there are multiple active nodes? There are two possible efficiency

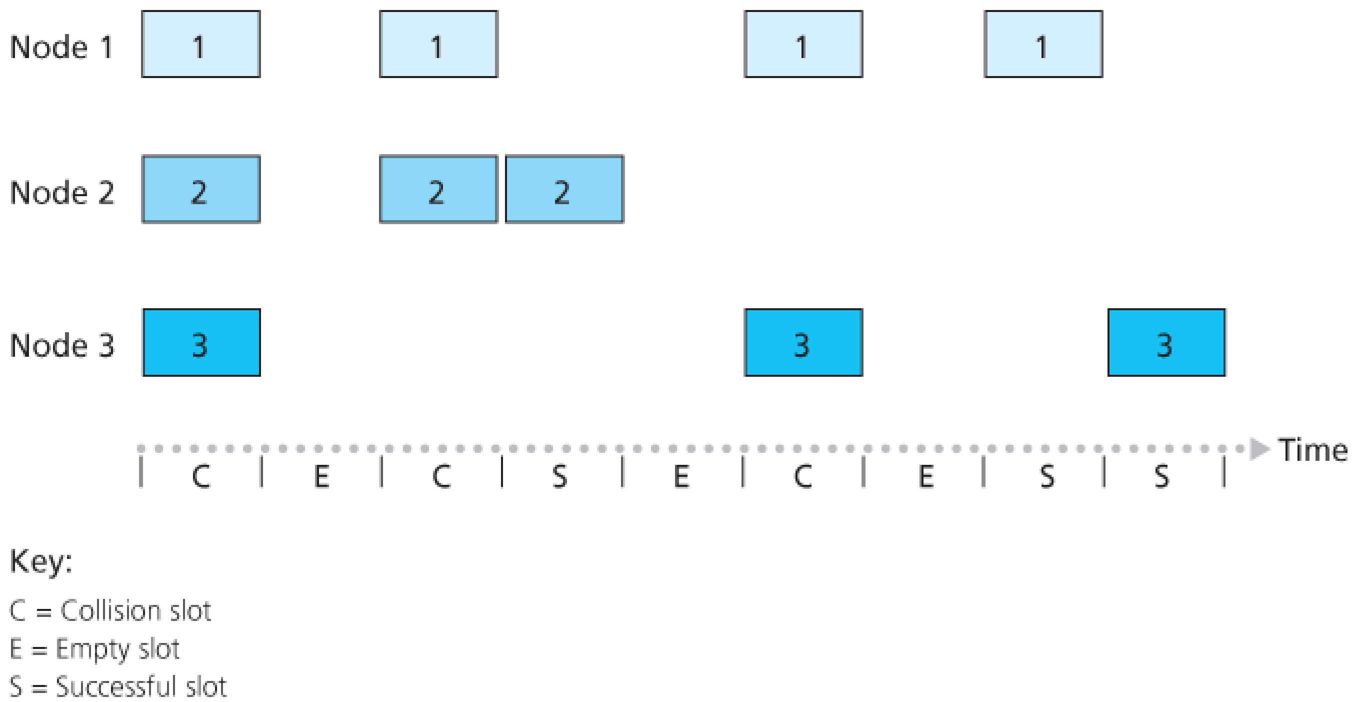


Figure 6.10 Nodes 1, 2, and 3 collide in the first slot. Node 2 finally succeeds in the fourth slot, node 1 in the eighth slot, and node 3 in the ninth slot

concerns here. First, as shown in **Figure 6.10**, when there are multiple active nodes, a certain fraction of the slots will have collisions and will therefore be "wasted." The second concern is that another fraction of the slots will be *empty* because all active nodes refrain from transmitting as a result of the probabilistic transmission policy. The only "unwasted" slots will be those in which exactly one node transmits. A slot in which exactly one node transmits is said to be a **successful slot**. The **efficiency** of a slotted multiple access protocol is defined to be the long-run fraction of successful slots in the case when there are a large number of active nodes, each always having a large number of frames to send. Note that if no form of access control were used, and each node were to immediately retransmit after each collision, the efficiency would be zero. Slotted ALOHA clearly increases the efficiency beyond zero, but by how much?

We now proceed to outline the derivation of the maximum efficiency of slotted ALOHA. To keep this derivation simple, let's modify the protocol a little and assume that each node attempts to transmit a frame in each slot with probability *p*. (That is, we assume that each node always has a frame to send and that the node transmits with probability *p* for a fresh frame as well as for a frame that has already suffered a collision.) Suppose there are *N* nodes. Then the probability that a given slot is a successful slot is the probability that one of the nodes transmits and that the remaining $N-1$ nodes do not transmit. The probability that a given node transmits is *p;* the probability that the remaining nodes do not transmit is $(1-p)^{N-1}$. Therefore the probability a given node has a success is $p(1-p)^{N-1}$. Because there are *N* nodes, the probability that any one of the *N* nodes has a success is $Np(1-p)^{N-1}$.

Thus, when there are *N* active nodes, the efficiency of slotted ALOHA is $Np(1-p)^{N-1}$. To obtain the *maximum* efficiency for *N* active nodes, we have to find the *p** that maximizes this expression. (See the homework problems for a general outline of this derivation.) And to obtain the maximum efficiency for a large number of active nodes, we take the limit of $Np^*(1-p^*)^{N-1}$ as *N* approaches infinity. (Again, see the homework problems.) After performing these calculations, we'll find that the maximum efficiency of the protocol is given by $1/e=0.37$. That is, when a large number of nodes have many frames to transmit, then (at best) only 37 percent of the slots do useful work. Thus the effective transmission rate of the channel is not *R* bps but only 0.37 *R* bps! A similar analysis also shows that 37 percent of the slots go empty and 26 percent of slots have collisions. Imagine the poor network administrator who has purchased a 100-Mbps slotted ALOHA system, expecting to be able to use the network to transmit data among a large number of users at an aggregate rate of, say, 80 Mbps! Although the channel is capable of transmitting a given frame at the full channel rate of 100 Mbps, in the long run, the successful throughput of this channel will be less than 37 Mbps.

ALOHA

The slotted ALOHA protocol required that all nodes synchronize their transmissions to start at the beginning of a slot. The first ALOHA protocol **[Abramson 1970]** was actually an unslotted, fully decentralized protocol. In pure ALOHA, when a frame first arrives (that is, a network-layer datagram is passed down from the network layer at the sending node), the node immediately transmits the frame in its entirety into the broadcast channel. If a transmitted frame experiences a collision with one or more other transmissions, the node will then immediately (after completely transmitting its collided frame) retransmit the frame with probability *p*. Otherwise, the node waits for a frame transmission time. After this wait, it then transmits the frame with probability *p*, or waits (remaining idle) for another frame time with probability 1 – *p*.

To determine the maximum efficiency of pure ALOHA, we focus on an individual node. We'll make the same assumptions as in our slotted ALOHA analysis and take the frame transmission time to be the unit of time. At any given time, the probability that a node is transmitting a frame is *p*. Suppose this frame begins transmission at time t_0. As shown in **Figure 6.11**, in order for this frame to be successfully transmitted, no other nodes can begin their transmission in the interval of time $[t_0-1, t_0]$. Such a transmission would overlap with the beginning of the transmission of node *i*'s frame. The probability that all other nodes do not begin a transmission in this interval is $(1-p)^{N-1}$. Similarly, no other node can begin a transmission while node *i* is transmitting, as such a transmission would overlap with the latter part of node *i*'s transmission. The probability that all other nodes do not begin a transmission in this interval is also $(1-p)^{N-1}$. Thus, the probability that a given node has a successful transmission is $p(1-p)^{2(N-1)}$. By taking limits as in the slotted ALOHA case, we find that the maximum efficiency of the pure ALOHA protocol is only $1/(2e)$—exactly half that of slotted ALOHA. This then is the price to be paid for a fully decentralized ALOHA protocol.

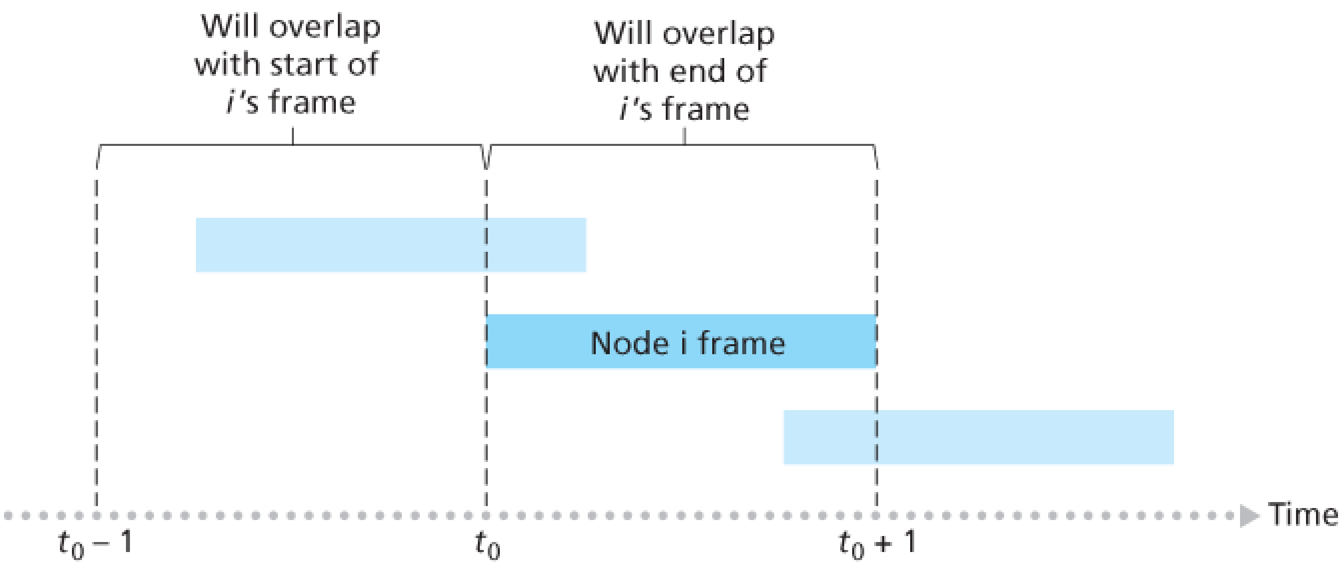


Figure 6.11 Interfering transmissions in pure ALOHA

Carrier Sense Multiple Access (CSMA)

In both slotted and pure ALOHA, a node's decision to transmit is made independently of the activity of the other nodes attached to the broadcast channel. In particular, a node neither pays attention to whether another node happens to be transmitting when it begins to transmit, nor stops transmitting if another node begins to interfere with its transmission. In our cocktail party analogy, ALOHA protocols are quite like a boorish partygoer who continues to chatter away regardless of whether other people are talking. As humans, we have human protocols that allow us not only to behave with more civility, but also to decrease the amount of time spent "colliding" with each other in conversation and, consequently, to increase the amount of data we exchange in our conversations. Specifically, there are two important rules for polite human conversation:

- **Listen before speaking.** If someone else is speaking, wait until they are finished. In the networking world, this is called **carrier sensing**—a node listens to the channel before transmitting. If a frame from another node is currently being transmitted into the channel, a node then waits until it detects no transmissions for a short amount of time and then begins transmission.
- **If someone else begins talking at the same time, stop talking.** In the networking world, this is called **collision detection**—a transmitting node listens to the channel while it is transmitting. If it detects that another node is transmitting an interfering frame, it stops transmitting and waits a random amount of time before repeating the sense-and-transmit-when-idle cycle.

These two rules are embodied in the family of **carrier sense multiple access (CSMA)** and **CSMA with collision detection (CSMA/CD)** protocols **[Kleinrock 1975b**; **Metcalfe 1976**; **Lam 1980**; **Rom 1990]**. Many variations on CSMA and

CASE HISTORY

> NORM ABRAMSON AND ALOHANET
>
> Norm Abramson, a PhD engineer, had a passion for surfing and an interest in packet switching. This combination of interests brought him to the University of Hawaii in 1969. Hawaii consists of many mountainous islands, making it difficult to install and operate land-based networks. When not surfing, Abramson thought about how to design a network that does packet switching over radio. The network he designed had one central host and several secondary nodes scattered over the Hawaiian Islands. The network had two channels, each using a different frequency band. The downlink channel broadcasted packets from the central host to the secondary hosts; and the upstream channel sent packets from the secondary hosts to the central host. In addition to sending informational packets, the central host also sent on the downstream channel an acknowledgment for each packet successfully received from the secondary hosts.
>
> Because the secondary hosts transmitted packets in a decentralized fashion, collisions on the upstream channel inevitably occurred. This observation led Abramson to devise the pure ALOHA protocol, as described in this chapter. In 1970, with continued funding from ARPA, Abramson connected his ALOHAnet to the ARPAnet. Abramson's work is important not only because it was the first example of a radio packet network, but also because it inspired Bob Metcalfe. A few years later, Metcalfe modified the ALOHA protocol to create the CSMA/CD protocol and the Ethernet LAN.

CSMA/CD have been proposed. Here, we'll consider a few of the most important, and fundamental, characteristics of CSMA and CSMA/CD.

The first question that you might ask about CSMA is why, if all nodes perform carrier sensing, do collisions occur in the first place? After all, a node will refrain from transmitting whenever it senses that another node is transmitting. The answer to the question can best be illustrated using space-time diagrams **[Molle 1987]**. **Figure 6.12** shows a space-time diagram of four nodes (A, B, C, D) attached to a linear broadcast bus. The horizontal axis shows the position of each node in space; the vertical axis represents time.

At time t_0, node B senses the channel is idle, as no other nodes are currently transmitting. Node B thus begins transmitting, with its bits propagating in both directions along the broadcast medium. The downward propagation of B's bits in **Figure 6.12** with increasing time indicates that a nonzero amount of time is needed for B's bits actually to propagate (albeit at near the speed of light) along the broadcast medium. At time t1(t1>t0), node D has a frame to send. Although node B is currently transmitting at time t_1, the bits being transmitted by B have yet to reach D, and thus D senses

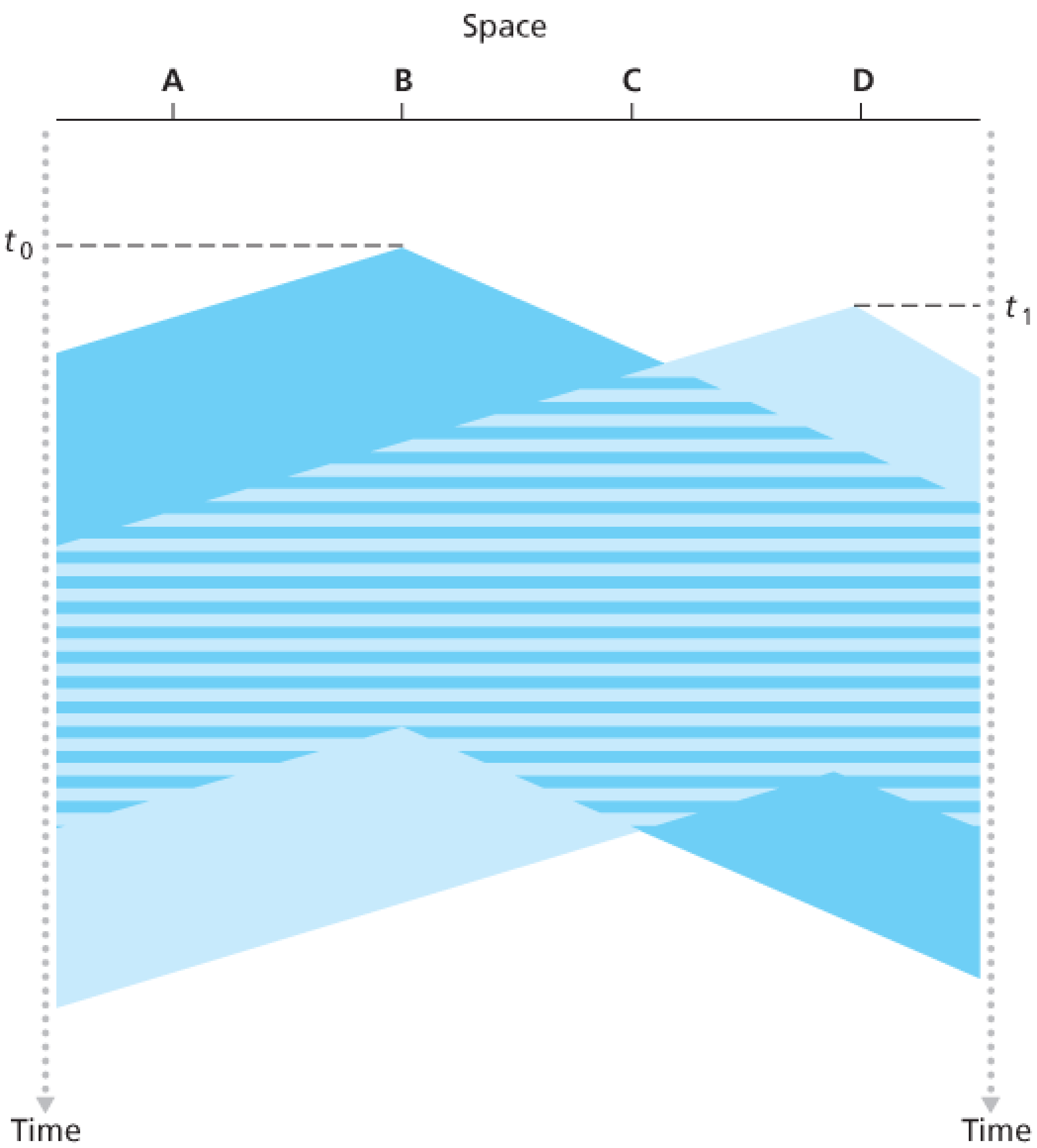


Figure 6.12 Space-time diagram of two CSMA nodes with colliding transmissions

the channel idle at t_1. In accordance with the CSMA protocol, D thus begins transmitting its frame. A short time later, B's transmission begins to interfere with D's transmission at D. From **Figure 6.12**, it is evident that the end-to-end **channel propagation delay** of a broadcast channel—the time it takes for a signal to propagate from one of the nodes to another—will play a crucial role in determining its performance. The longer this propagation delay, the larger the chance that a carrier-sensing node is not yet able to sense a transmission that has already begun at another node in the network.

Carrier Sense Multiple Access with Collision Dection (CSMA/CD)

In **Figure 6.12**, nodes do not perform collision detection; both B and D continue to transmit their frames in their entirety even though a collision has occurred. When a node performs collision detection, it ceases transmission as soon as it detects a collision. **Figure 6.13** shows the same scenario as in **Figure 6.12**, except that the two

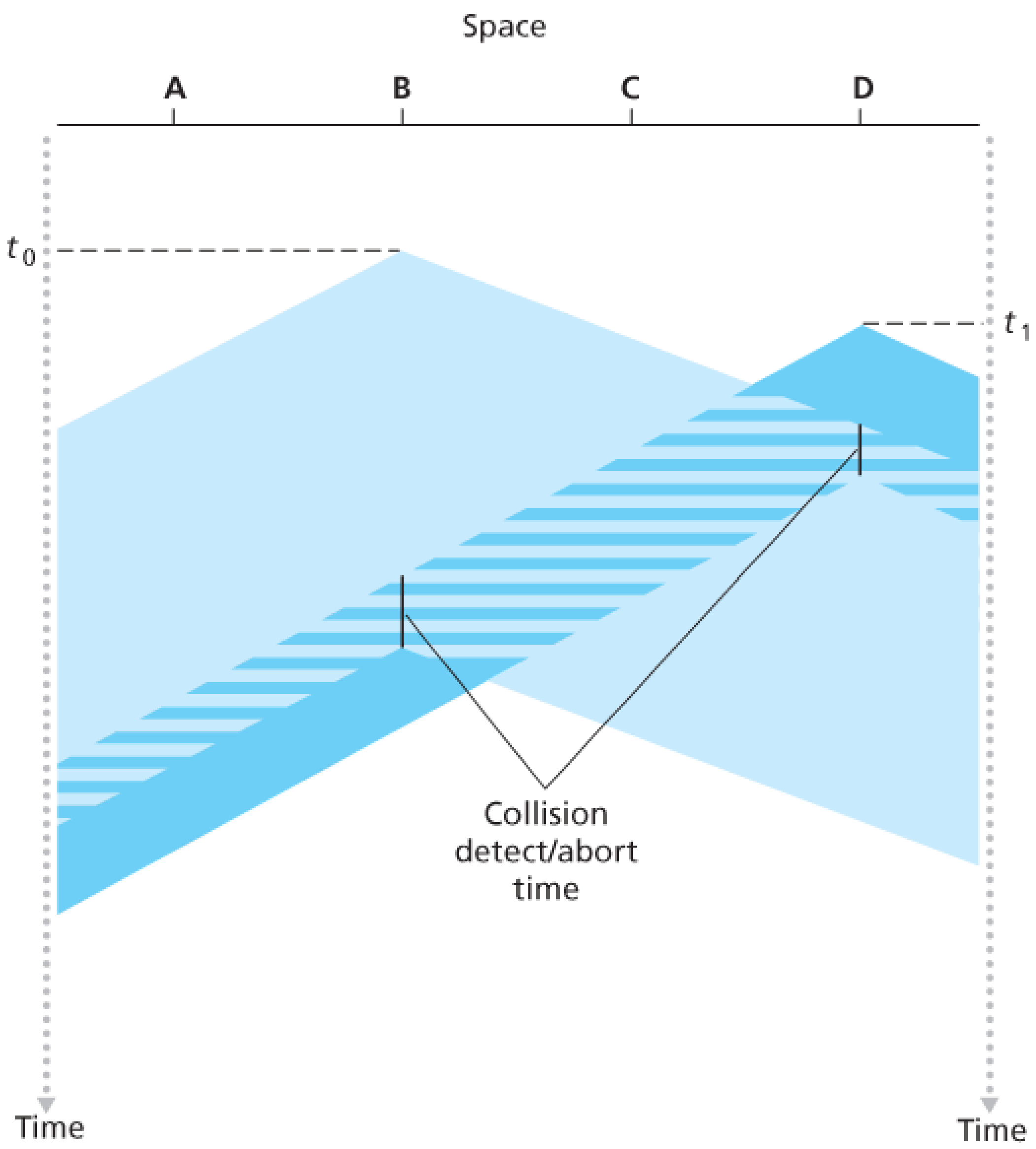


Figure 6.13 CSMA with collision detection

nodes each abort their transmission a short time after detecting a collision. Clearly, adding collision detection to a multiple access protocol will help protocol performance by not transmitting a useless, damaged (by interference with a frame from another node) frame in its entirety.

Before analyzing the CSMA/CD protocol, let us now summarize its operation from the perspective of an adapter (in a node) attached to a broadcast channel:

1. The adapter obtains a datagram from the network layer, prepares a link-layer frame, and puts the frame adapter buffer.
2. If the adapter senses that the channel is idle (that is, there is no signal energy entering the adapter from the channel), it starts to transmit the frame. If, on the other hand, the adapter senses that the channel is busy, it waits until it senses no signal energy and then starts to transmit the frame.
3. While transmitting, the adapter monitors for the presence of signal energy coming from other adapters using the broadcast channel.
4. If the adapter transmits the entire frame without detecting signal energy from other adapters, the

adapter is finished with the frame. If, on the other hand, the adapter detects signal energy from other adapters while transmitting, it aborts the transmission (that is, it stops transmitting its frame).

5. After aborting, the adapter waits a random amount of time and then returns to step 2.

The need to wait a random (rather than fixed) amount of time is hopefully clear—if two nodes transmitted frames at the same time and then both waited the same fixed amount of time, they'd continue colliding forever. But what is a good interval of time from which to choose the random backoff time? If the interval is large and the number of colliding nodes is small, nodes are likely to wait a large amount of time (with the channel remaining idle) before repeating the sense-and-transmit-when-idle step. On the other hand, if the interval is small and the number of colliding nodes is large, it's likely that the chosen random values will be nearly the same, and transmitting nodes will again collide. What we'd like is an interval that is short when the number of colliding nodes is small, and long when the number of colliding nodes is large.

The **binary exponential backoff** algorithm, used in Ethernet as well as in DOCSIS cable network multiple access protocols **[DOCSIS 2011]**, elegantly solves this problem. Specifically, when transmitting a frame that has already experienced *n* collisions, a node chooses the value of *K* at random from $\{0,1,2,\ldots 2^n-1\}$. Thus, the more collisions experienced by a frame, the larger the interval from which *K* is chosen. For Ethernet, the actual amount of time a node waits is $K \cdot 512$ bit times (i.e., *K* times the amount of time needed to send 512 bits into the Ethernet) and the maximum value that *n* can take is capped at 10.

Let's look at an example. Suppose that a node attempts to transmit a frame for the first time and while transmitting it detects a collision. The node then chooses K=0 with probability 0.5 or chooses K=1 with probability 0.5. If the node chooses K=0, then it immediately begins sensing the channel. If the node chooses K=1, it waits 512 bit times (e.g., 5.12 microseconds for a 100 Mbps Ethernet) before beginning the sense-and-transmit-when-idle cycle. After a second collision, *K* is chosen with equal probability from {0,1,2,3}. After three collisions, *K* is chosen with equal probability from {0,1,2,3,4,5,6,7}. After 10 or more collisions, *K* is chosen with equal probability from {0,1,2,…, 1023}. Thus, the size of the sets from which *K* is chosen grows exponentially with the number of collisions; for this reason this algorithm is referred to as binary exponential backoff.

We also note here that each time a node prepares a new frame for transmission, it runs the CSMA/CD algorithm, not taking into account any collisions that may have occurred in the recent past. So it is possible that a node with a new frame will immediately be able to sneak in a successful transmission while several other nodes are in the exponential backoff state.

CSMA/CD Efficiency

When only one node has a frame to send, the node can transmit at the full channel rate (e.g., for Ethernet typical rates are 10 Mbps, 100 Mbps, or 1 Gbps). However, if many nodes have frames to transmit, the effective transmission rate of the channel can be much less. We define the **efficiency of CSMA/CD** to be the long-run fraction of time during which frames are being transmitted on the channel without collisions when there is a large number of active nodes, with each node having a large number of frames to send. In order to present a closed-form approximation of the efficiency of Ethernet, let d_{prop} denote the maximum time it takes signal energy to propagate between any two adapters. Let d_{trans} be the time to transmit a maximum-size frame (approximately 1.2 msecs for a 10 Mbps Ethernet). A derivation of the efficiency of CSMA/CD is beyond the scope of this book (see **[Lam 1980]** and **[Bertsekas 1991]**). Here we simply state the following approximation:

Efficiency=11+5dprop/dtrans

We see from this formula that as d_{prop} approaches 0, the efficiency approaches 1. This matches our intuition that if the propagation delay is zero, colliding nodes will abort immediately without wasting the channel. Also, as d_{trans} becomes very large, efficiency approaches 1. This is also intuitive because when a frame grabs the channel, it will hold on to the channel for a very long time; thus, the channel will be doing productive work most of the time.

6.3.3 Taking-Turns Protocols

Recall that two desirable properties of a multiple access protocol are (1) when only one node is active, the active node has a throughput of R bps, and (2) when M nodes are active, then each active node has a throughput of nearly R/M bps. The ALOHA and CSMA protocols have this first property but not the second. This has motivated researchers to create another class of protocols—the **taking-turns protocols**. As with random access protocols, there are dozens of taking-turns protocols, and each one of these protocols has many variations. We'll discuss two of the more important protocols here. The first one is the **polling protocol**. The polling protocol requires one of the nodes to be designated as a master node. The master node **polls** each of the nodes in a round-robin fashion. In particular, the master node first sends a message to node 1, saying that it (node 1) can transmit up to some maximum number of frames. After node 1 transmits some frames, the master node tells node 2 it (node 2) can transmit up to the maximum number of frames. (The master node can determine when a node has finished sending its frames by observing the lack of a signal on the channel.) The procedure continues in this manner, with the master node polling each of the nodes in a cyclic manner.

The polling protocol eliminates the collisions and empty slots that plague random access protocols. This allows polling to achieve a much higher efficiency. But it also has a few drawbacks. The first drawback is that the protocol introduces a polling delay—the amount of time required to notify a node that it can transmit. If, for example, only one node is active, then the node will transmit at a rate less than *R* bps, as the master node must poll each of the inactive nodes in turn each time the active node has sent its maximum number of frames. The second drawback, which is potentially more serious, is that if the master node fails, the entire channel becomes inoperative. The 802.15 protocol and the Bluetooth protocol we will study in **Section 6.3** are examples of polling protocols.

The second taking-turns protocol is the **token-passing protocol**. In this protocol there is no master node. A small, special-purpose frame known as a **token** is exchanged among the nodes in some fixed order. For example, node 1 might always send the token to node 2, node 2 might always send the token to node 3, and node *N* might always send the token to node 1. When a node receives a token, it holds onto the token only if it has some frames to transmit; otherwise, it immediately forwards the token to the next node. If a node does have frames to transmit when it receives the token, it sends up to a maximum number of frames and then forwards the token to the next node. Token passing is decentralized and highly efficient. But it has its problems as well. For example, the failure of one node can crash the entire channel. Or if a node accidentally neglects to release the token, then some recovery procedure must be invoked to get the token back in circulation. Over the years many token-passing protocols have been developed, including the fiber distributed data interface (FDDI) protocol **[Jain 1994]** and the IEEE 802.5 token ring protocol **[IEEE 802.5 2012]**, and each one had to address these as well as other sticky issues.

6.3.4 DOCSIS: The Link-Layer Protocol for Cable Internet Access

In the previous three subsections, we've learned about three broad classes of multiple access protocols: channel partitioning protocols, random access protocols, and taking turns protocols. A cable access network will make for an excellent case study here, as we'll find aspects of *each* of these three classes of multiple access protocols with the cable access network!

Recall from **Section 1.2.1** that a cable access network typically connects several thousand residential cable modems to a cable modem termination system (CMTS) at the cable network headend. The Data-Over-Cable Service Interface Specifications (DOCSIS) **[DOCSIS 2011]** specifies the cable data network architecture and its protocols. DOCSIS uses FDM to divide the downstream (CMTS to modem) and upstream (modem to CMTS) network segments into multiple frequency channels. Each downstream channel is 6 MHz wide, with a maximum throughput of approximately 40 Mbps per channel (although this data rate is seldom seen at a cable modem in practice); each upstream channel has a maximum channel width of 6.4 MHz, and a maximum upstream throughput of approximately 30 Mbps. Each upstream and

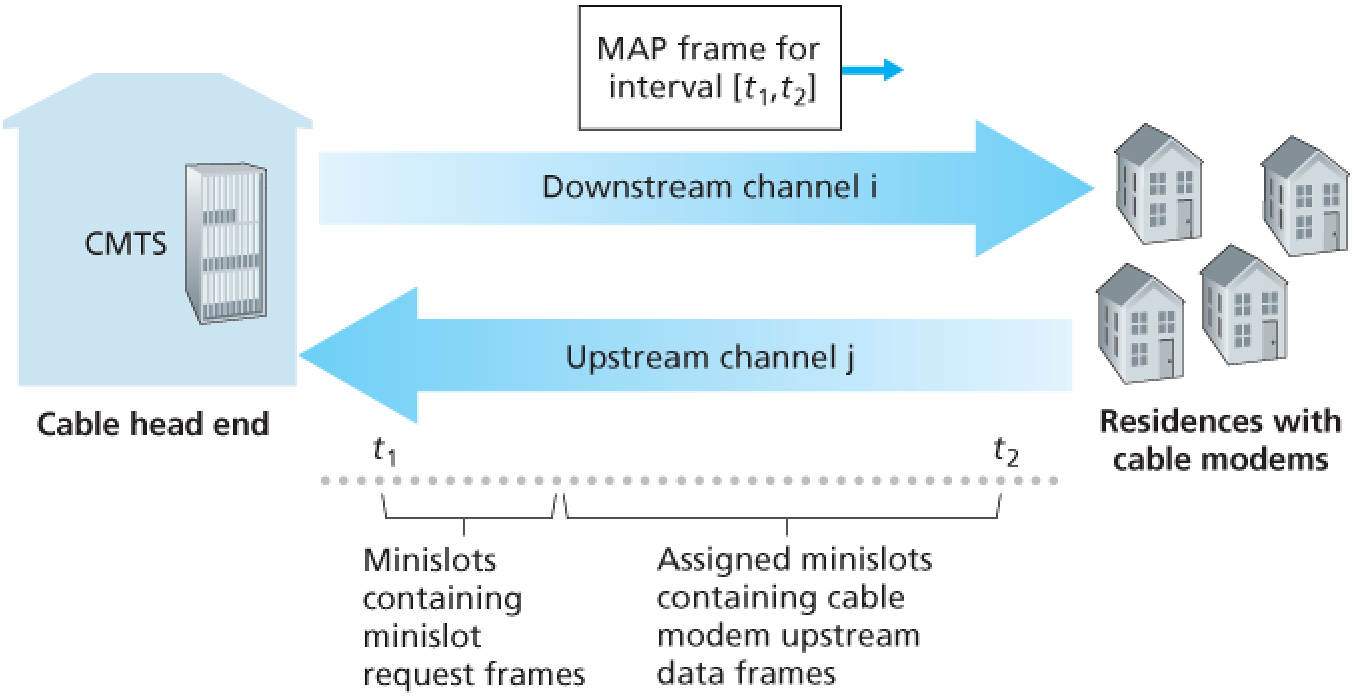


Figure 6.14 Upstream and downstream channels between CMTS and cable modems

downstream channel is a broadcast channel. Frames transmitted on the downstream channel by the CMTS are received by all cable modems receiving that channel; since there is just a single CMTS transmitting into the downstream channel, however, there is no multiple access problem. The upstream direction, however, is more interesting and technically challenging, since multiple cable modems share the same upstream channel (frequency) to the CMTS, and thus collisions can potentially occur.

As illustrated in **Figure 6.14**, each upstream channel is divided into intervals of time (TDM-like), each containing a sequence of mini-slots during which cable modems can transmit to the CMTS. The CMTS explicitly grants permission to individual cable modems to transmit during specific mini-slots. The CMTS accomplishes this by sending a control message known as a MAP message on a downstream channel to specify which cable modem (with data to send) can transmit during which mini-slot for the interval of time specified in the control message. Since mini-slots are explicitly allocated to cable modems, the CMTS can ensure there are no colliding transmissions during a mini-slot.

But how does the CMTS know which cable modems have data to send in the first place? This is accomplished by having cable modems send mini-slot-request frames to the CMTS during a special set of interval mini-slots that are dedicated for this purpose, as shown in **Figure 6.14**. These mini-slot-request frames are transmitted in a random access manner and so may collide with each other. A cable modem can neither sense whether the upstream channel is busy nor detect collisions. Instead, the cable modem infers that its mini-slot-request frame experienced a collision if it does not receive a response to the requested allocation in the next downstream control message. When a collision is inferred, a cable modem uses binary exponential backoff to defer the retransmission of its mini-slot-request frame to a future time slot. When there is little traffic on the upstream channel, a cable modem may actually transmit data frames during slots nominally assigned for mini-slot-request frames (and thus avoid having to wait for a mini-slot assignment).

A cable access network thus serves as a terrific example of multiple access protocols in action—FDM, TDM, random access, and centrally allocated time slots all within one network!

6.4 Switched Local Area Networks

Having covered broadcast networks and multiple access protocols in the previous section, let's turn our attention next to switched local networks. **Figure 6.15** shows a switched local network connecting three departments, two servers and a router with four switches. Because these switches operate at the link layer, they switch link-layer frames (rather than network-layer datagrams), don't recognize network-layer addresses, and don't use routing algorithms like RIP or OSPF to determine

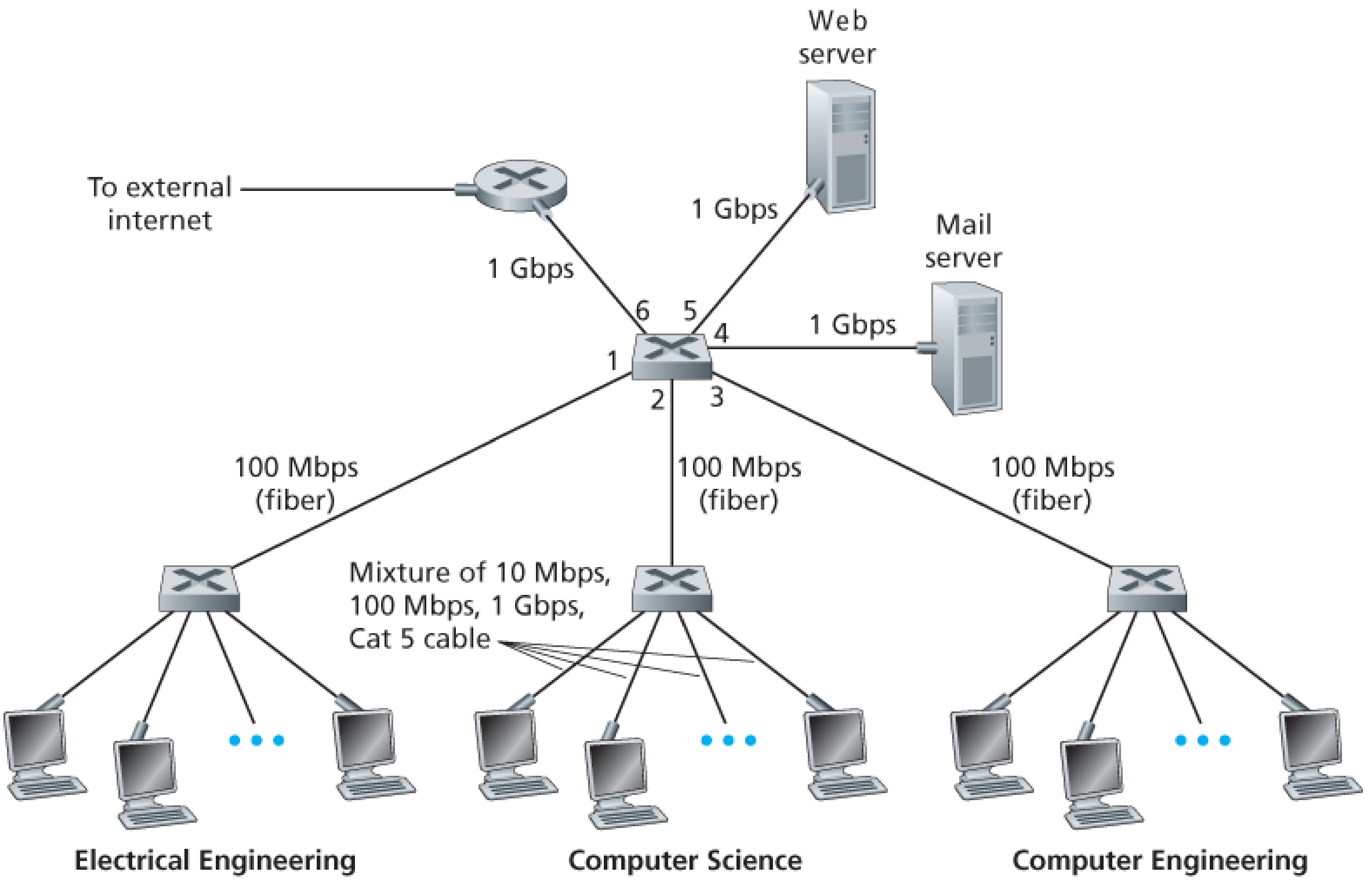


Figure 6.15 An institutional network connected together by four switches

paths through the network of layer-2 switches. Instead of using IP addresses, we will soon see that they use link-layer addresses to forward link-layer frames through the network of switches. We'll begin our study of switched LANs by first covering link-layer addressing (**Section 6.4.1**). We then examine the celebrated Ethernet protocol (**Section 6.5.2**). After examining link-layer addressing and Ethernet, we'll look at how link-layer switches operate (**Section 6.4.3**), and then see (**Section 6.4.4**) how these switches are often used to build large-scale LANs.

6.4.1 Link-Layer Addressing and ARP

Hosts and routers have link-layer addresses. Now you might find this surprising, recalling from **Chapter 4** that hosts and routers have network-layer addresses as well. You might be asking, why in the world do we need to have addresses at both the network and link layers? In addition to describing the syntax and function of the link-layer addresses, in this section we hope to shed some light on why the two layers of addresses are useful and, in fact, indispensable. We'll also cover the Address Resolution Protocol (ARP), which provides a mechanism to translate IP addresses to link-layer addresses.

MAC Addresses

In truth, it is not hosts and routers that have link-layer addresses but rather their adapters (that is, network interfaces) that have link-layer addresses. A host or router with multiple network interfaces will thus have multiple link-layer addresses associated with it, just as it would also have multiple IP addresses associated with it. It's important to note, however, that link-layer switches do not have link-layer addresses associated with their interfaces that connect to hosts and routers. This is because the job of the link-layer switch is to carry datagrams between hosts and routers; a switch does this job transparently, that is, without the host or router having to explicitly address the frame to the intervening switch. This is illustrated in **Figure 6.16**. A link-layer address is variously called a **LAN address**, a **physical address**, or a **MAC address**. Because MAC address seems to be the most popular term, we'll henceforth refer to link-layer addresses as MAC addresses. For most LANs (including Ethernet and 802.11 wireless LANs), the MAC address is 6 bytes long, giving 2^{48} possible MAC addresses. As shown in **Figure 6.16**, these 6-byte addresses are typically expressed in hexadecimal notation, with each byte of the address expressed as a pair of hexadecimal numbers. Although MAC addresses were designed to be permanent, it is now possible to change an adapter's MAC address via software. For the rest of this section, however, we'll assume that an adapter's MAC address is fixed.

One interesting property of MAC addresses is that no two adapters have the same address. This might seem surprising given that adapters are manufactured in many countries by many companies. How does a company manufacturing adapters in Taiwan make sure that it is using different addresses from a company manufacturing

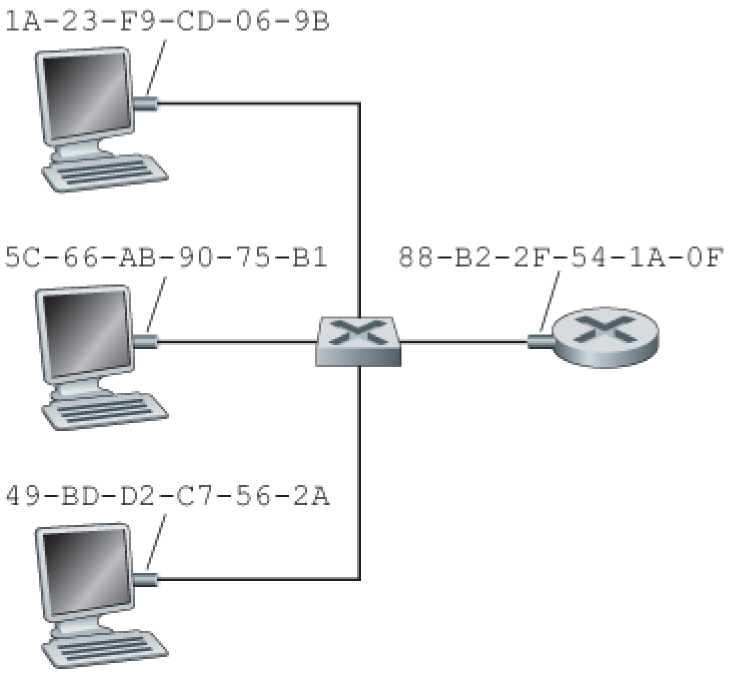


Figure 6.16 Each interface connected to a LAN has a unique MAC address

adapters in Belgium? The answer is that the IEEE manages the MAC address space. In particular, when a company wants to manufacture adapters, it purchases a chunk of the address space consisting of 2^{24} addresses for a nominal fee. IEEE allocates the chunk of 2^{24} addresses by fixing the first 24 bits of a MAC address and letting the company create unique combinations of the last 24 bits for each adapter.

An adapter's MAC address has a flat structure (as opposed to a hierarchical structure) and doesn't change no matter where the adapter goes. A laptop with an Ethernet interface always has the same MAC address, no matter where the computer goes. A smartphone with an 802.11 interface always has the same MAC address, no matter where the smartphone goes. Recall that, in contrast, IP addresses have a hierarchical structure (that is, a network part and a host part), and a host's IP addresses needs to be changed when the host moves, i.e., changes the network to which it is attached. An adapter's MAC address is analogous to a person's social security number, which also has a flat addressing structure and which doesn't change no matter where the person goes. An IP address is analogous to a person's postal address, which is hierarchical and which must be changed whenever a person moves. Just as a person may find it useful to have both a postal address and a social security number, it is useful for a host and router interfaces to have both a network-layer address and a MAC address.

When an adapter wants to send a frame to some destination adapter, the sending adapter inserts the destination adapter's MAC address into the frame and then sends the frame into the LAN. As we will soon see, a switch occasionally broadcasts an incoming frame onto all of its interfaces. We'll see in **Chapter 7** that 802.11 also broadcasts frames. Thus, an adapter may receive a frame that isn't addressed to it. Thus, when an adapter receives a frame, it will check to see whether the destination MAC address in the frame matches its own MAC address. If there is a match, the adapter extracts the enclosed datagram and passes the datagram up the protocol stack. If there isn't a match, the adapter discards the frame, without passing the network-layer datagram up. Thus, the destination only will be interrupted when the frame is received.

However, sometimes a sending adapter *does* want all the other adapters on the LAN to receive and *process* the frame it is about to send. In this case, the sending adapter inserts a special MAC **broadcast address** into the destination address field of the frame. For LANs that use 6-byte addresses (such as Ethernet and 802.11), the broadcast address is a string of 48 consecutive 1s (that is, FF-FF-FF-FF-FF-FF in hexadecimal notation).

Address Resolution Protocol (ARP)

Because there are both network-layer addresses (for example, Internet IP addresses) and link-layer addresses (that is, MAC addresses), there is a need to translate between them. For the Internet, this is the job of the **Address Resolution Protocol (ARP)** **[RFC 826]**.

To understand the need for a protocol such as ARP, consider the network shown in **Figure 6.17**. In this simple example, each host and router has a single IP address and single MAC address. As usual, IP addresses are shown in dotted-decimal

> PRINCIPLES IN PRACTICE
>
> KEEPING THE LAYERS INDEPENDENT
>
> There are several reasons why hosts and router interfaces have MAC addresses in addition to network-layer addresses. First, LANs are designed for arbitrary network-layer protocols, not just for IP and the Internet. If adapters were assigned IP addresses rather than “neutral” MAC addresses, then adapters would not easily be able to support other network-layer protocols (for example, IPX or DECnet). Second, if adapters were to use network-layer addresses instead of MAC addresses, the network-layer address would have to be stored in the adapter RAM and reconfigured every time the adapter was moved (or powered up). Another option is to not use any addresses in the adapters and have each adapter pass the data (typically, an IP datagram) of each frame it receives up the protocol stack. The network layer could then check for a matching network-layer address. One problem with this option is that the host would be interrupted by every frame sent on the LAN, including by frames that were destined for other hosts on the same broadcast LAN. In summary, in order for the layers to be largely independent building blocks in a network architecture, different layers need to have their own addressing scheme. We have now seen three types of addresses: host names for the application layer, IP addresses for the network layer, and MAC addresses for the link layer.

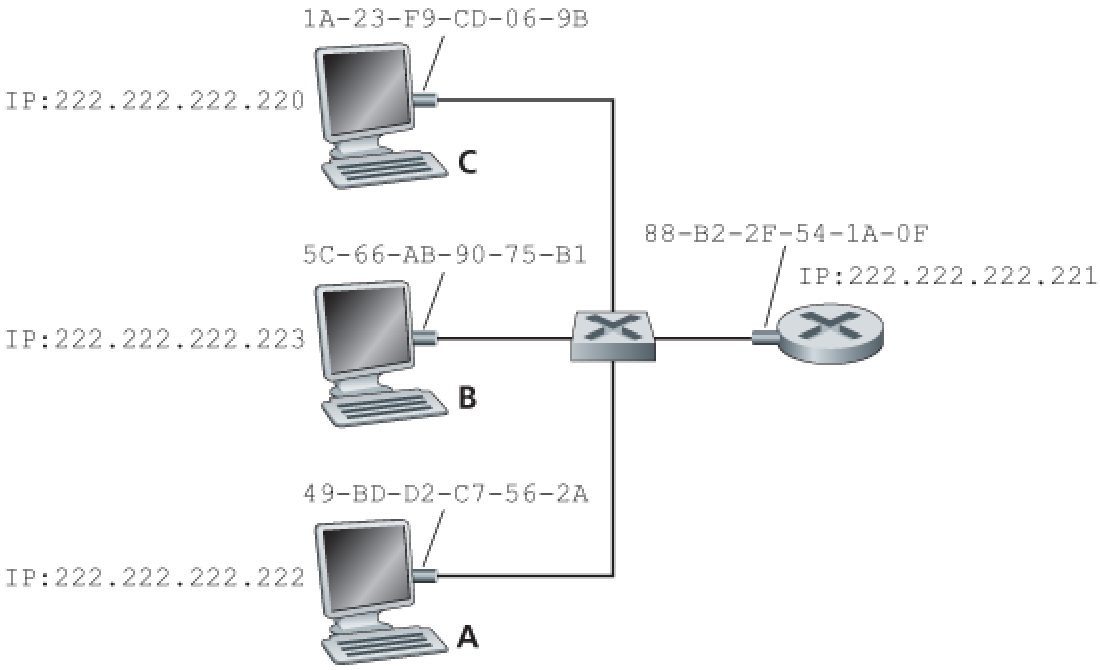


Figure 6.17 Each interface on a LAN has an IP address and a MAC address

notation and MAC addresses are shown in hexadecimal notation. For the purposes of this discussion, we will assume in this section that the switch broadcasts all frames; that is, whenever a switch receives a frame on one interface, it forwards the frame on all of its other interfaces. In the next section, we will provide a more accurate explanation of how switches operate.

Now suppose that the host with IP address 222.222.222.220 wants to send an IP datagram to host 222.222.222.222. In this example, both the source and destination are in the same subnet, in the addressing sense of **Section 4.3.3**. To send a datagram, the source must give its adapter not only the IP datagram but also the MAC address for destination 222.222.222.222. The sending adapter will then construct a link-layer frame containing the destination's MAC address and send the frame into the LAN.

The important question addressed in this section is, How does the sending host determine the MAC address for the destination host with IP address 222.222.222.222? As you might have guessed, it uses ARP. An ARP module in the sending host takes any IP address on the same LAN as input, and returns the corresponding MAC address. In the example at hand, sending host 222.222.222.220 provides its ARP module the IP address 222.222.222.222, and the ARP module returns the corresponding MAC address 49-BD-D2-C7-56-2A.

So we see that ARP resolves an IP address to a MAC address. In many ways it is analogous to DNS (studied in **Section 2.5**), which resolves host names to IP addresses. However, one important difference between the two resolvers is that DNS resolves host names for hosts anywhere in the Internet, whereas ARP resolves IP addresses only for hosts and router interfaces on the same subnet. If a node in California were to try to use ARP to resolve the IP address for a node in Mississippi, ARP would return with an error.

IP Address	MAC Address	TTL
222.222.222.221	88-B2-2F-54-1A-0F	13:45:00
222.222.222.223	5C-66-AB-90-75-B1	13:52:00

Figure 6.18 A possible ARP table in 222.222.222.220

Now that we have explained what ARP does, let's look at how it works. Each host and router has an **ARP table** in its memory, which contains mappings of IP addresses to MAC addresses. **Figure 6.18** shows what an ARP table in host 222.222.222.220 might look like. The ARP table also contains a time-to-live (TTL) value, which indicates when each mapping will be deleted from the table. Note that a table does not necessarily contain an entry for every host and router on the subnet; some may have never been entered into the table, and others may have expired. A typical expiration time for an entry is 20 minutes from when an entry is placed in an ARP table.

Now suppose that host 222.222.222.220 wants to send a datagram that is IP-addressed to another host or router on that subnet. The sending host needs to obtain the MAC address of the destination given the IP address. This task is easy if the sender's ARP table has an entry for the destination node. But what if the ARP table doesn't currently have an entry for the destination? In particular, suppose 222.222.222.220 wants to send a datagram to 222.222.222.222. In this case, the sender uses the ARP protocol to resolve the address. First, the sender constructs a special packet called an **ARP packet**. An ARP packet has several fields, including the sending and receiving IP and MAC addresses. Both ARP query and response packets have the same format. The purpose of the ARP query packet is to query all the other hosts and routers on the subnet to determine the MAC address corresponding to the IP address that is being resolved.

Returning to our example, 222.222.222.220 passes an ARP query packet to the adapter along with an indication that the adapter should send the packet to the MAC broadcast address, namely, FF-FF-FF-FF-FF-FF. The adapter encapsulates the ARP packet in a link-layer frame, uses the broadcast address for the frame's destination address, and transmits the frame into the subnet. Recalling our social security number/postal address analogy, an ARP query is equivalent to a person shouting out in a crowded room of cubicles in some company (say, AnyCorp): "What is the social security number of the person whose postal address is Cubicle 13, Room 112, AnyCorp, Palo Alto, California?" The frame containing the ARP query is received by all the other adapters on the subnet, and (because of the broadcast address) each adapter passes the ARP packet within the frame up to its ARP module. Each of these ARP modules checks to see if its IP address matches the destination IP address in the ARP packet. The one with a match sends back to the querying host a response ARP packet with the desired mapping. The querying host 222.222.222.220 can then update its ARP table and send its IP datagram, encapsulated in a link-layer frame whose destination MAC is that of the host or router responding to the earlier ARP query.

There are a couple of interesting things to note about the ARP protocol. First, the query ARP message is sent within a broadcast frame, whereas the response ARP message is sent within a standard frame. Before reading on you should think about why this is so. Second, ARP is plug-and-play; that is, an ARP table gets built automatically—it doesn't have to be configured by a system administrator. And if a host becomes disconnected from the subnet, its entry is eventually deleted from the other ARP tables in the subnet.

Students often wonder if ARP is a link-layer protocol or a network-layer protocol. As we've seen, an ARP packet is encapsulated within a link-layer frame and thus lies architecturally above the link layer. However, an ARP packet has fields containing link-layer addresses and thus is arguably a link-layer protocol, but it also contains network-layer addresses and thus is also arguably a network-layer protocol. In the end, ARP is probably best considered a protocol that straddles the boundary between the link and network layers—not fitting neatly into the simple layered protocol stack we studied in **Chapter 1**. Such are the complexities of real-world protocols!

Sending a Datagram off the Subnet

It should now be clear how ARP operates when a host wants to send a datagram to another host *on the same subnet.* But now let's look at the more complicated situation when a host on a subnet wants to send a network-layer datagram to a host *off the subnet* (that is, across a router onto another subnet). Let's discuss this issue in the context of **Figure 6.19**, which shows a simple network consisting of two subnets interconnected by a router.

There are several interesting things to note about **Figure 6.19**. Each host has exactly one IP address and one adapter. But, as discussed in **Chapter 4**, a router has an IP address for *each* of its interfaces. For each router interface there is also an ARP module (in the router) and an adapter. Because the router in **Figure 6.19** has two interfaces, it has two IP addresses, two ARP modules, and two adapters. Of course, each adapter in the network has its own MAC address.

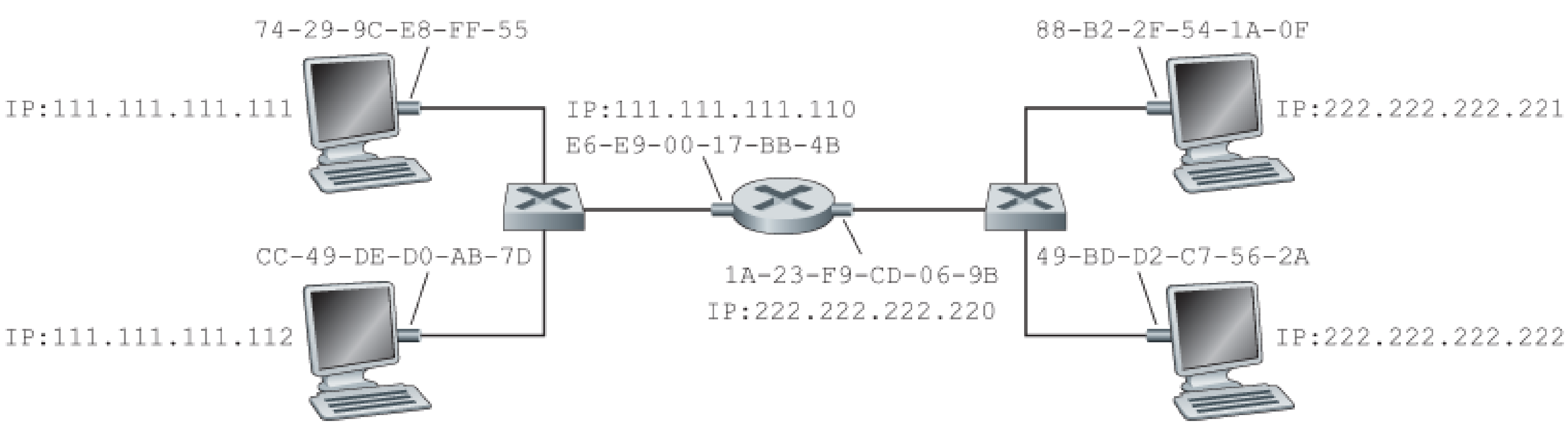


Figure 6.19 Two subnets interconnected by a router

Also note that Subnet 1 has the network address 111.111.111/24 and that Subnet 2 has the network address 222.222.222/24. Thus all of the interfaces connected to Subnet 1 have addresses of the form 111.111.111.xxx and all of the interfaces connected to Subnet 2 have addresses of the form 222.222.222.xxx.

Now let's examine how a host on Subnet 1 would send a datagram to a host on Subnet 2. Specifically, suppose that host 111.111.111.111 wants to send an IP datagram to a host 222.222.222.222. The sending host passes the datagram to its adapter, as usual. But the sending host must also indicate to its adapter an appropriate destination MAC address. What MAC address should the adapter use? One might be tempted to guess that the appropriate MAC address is that of the adapter for host 222.222.222.222, namely, 49-BD-D2-C7-56-2A. This guess, however, would be wrong! If the sending adapter were to use that MAC address, then none of the adapters on Subnet 1 would bother to pass the IP datagram up to its network layer, since the frame's destination address would not match the MAC address of any adapter on Subnet 1. The datagram would just die and go to datagram heaven.

If we look carefully at **Figure 6.19**, we see that in order for a datagram to go from 111.111.111.111 to a host on Subnet 2, the datagram must first be sent to the router interface 111.111.111.110, which is the IP address of the first-hop router on the path to the final destination. Thus, the appropriate MAC address for the frame is the address of the adapter for router interface 111.111.111.110, namely, E6-E9-00-17-BB-4B. How does the sending host acquire the MAC address for 111.111.111.110? By using ARP, of course! Once the sending adapter has this MAC address, it creates a frame (containing the datagram addressed to 222.222.222.222) and sends the frame into Subnet 1. The router adapter on Subnet 1 sees that the link-layer frame is addressed to it, and therefore passes the frame to the network layer of the router. Hooray—the IP datagram has successfully been moved from source host to the router! But we are not finished. We still have to move the datagram from the router to the destination. The router now has to determine the correct interface on which the datagram is to be forwarded. As discussed in **Chapter 4**, this is done by consulting a forwarding table in the router. The forwarding table tells the router that the datagram is to be forwarded via router interface 222.222.222.220. This interface then passes the datagram to its adapter, which encapsulates the datagram in a new frame and sends the frame into Subnet 2. This time, the destination MAC address of the frame is indeed the MAC address of the ultimate destination. And how does the router obtain this destination MAC address? From ARP, of course!

ARP for Ethernet is defined in RFC 826. A nice introduction to ARP is given in the TCP/IP tutorial, RFC 1180. We'll explore ARP in more detail in the homework problems.

6.4.2 Ethernet

Ethernet has pretty much taken over the wired LAN market. In the 1980s and the early 1990s, Ethernet faced many challenges from other LAN technologies, including token ring, FDDI, and ATM. Some of these other technologies succeeded in capturing a part of the LAN market for a few years. But since its invention in the mid-1970s, Ethernet has continued to evolve and grow and has held on to its dominant position. Today, Ethernet is by far the most prevalent wired LAN technology, and it is likely to remain so for the foreseeable future. One might say that Ethernet has been to local area networking what the Internet has been to global networking.

There are many reasons for Ethernet's success. First, Ethernet was the first widely deployed high-speed LAN. Because it was deployed early, network administrators became intimately familiar with Ethernet—its wonders and its quirks—and were reluctant to switch over to other LAN technologies when they came on the scene. Second, token ring, FDDI, and ATM were more complex and expensive than Ethernet, which further discouraged network administrators from switching over. Third, the most compelling reason to switch to another LAN technology (such as FDDI or ATM) was usually the higher data rate of the new technology; however, Ethernet always fought back, producing versions that operated at equal data rates or higher. Switched Ethernet was also introduced in the early 1990s, which further increased its effective data rates. Finally, because Ethernet has been so popular, Ethernet hardware (in particular, adapters and switches) has become a commodity and is remarkably cheap.

The original Ethernet LAN was invented in the mid-1970s by Bob Metcalfe and David Boggs. The original Ethernet LAN used a coaxial bus to interconnect the nodes. Bus topologies for Ethernet actually persisted throughout the 1980s and into the mid-1990s. Ethernet with a bus topology is a broadcast LAN—all transmitted frames travel to and are processed by *all* adapters connected to the bus. Recall that we covered Ethernet's CSMA/CD multiple access protocol with binary exponential backoff in **Section 6.3.2**.

By the late 1990s, most companies and universities had replaced their LANs with Ethernet installations using a hub-based star topology. In such an installation the hosts (and routers) are directly connected to a hub with twisted-pair copper wire. A **hub** is a physical-layer device that acts on individual bits rather than frames. When a bit, representing a zero or a one, arrives from one interface, the hub simply re-creates the bit, boosts its energy strength, and transmits the bit onto all the other interfaces. Thus, Ethernet with a hub-based star topology is also a broadcast LAN—whenever a hub receives a bit from one of its interfaces, it sends a copy out on all of its other interfaces. In particular, if a hub receives frames from two different interfaces at the same time, a collision occurs and the nodes that created the frames must retransmit.

In the early 2000s Ethernet experienced yet another major evolutionary change. Ethernet installations continued to use a star topology, but the hub at the center was replaced with a **switch**. We'll be examining switched Ethernet in depth later in this chapter. For now, we only mention that a switch is not only "collision-less" but is also a bona-fide store-and-forward packet switch; but unlike routers, which operate up through layer 3, a switch operates only up through layer 2.

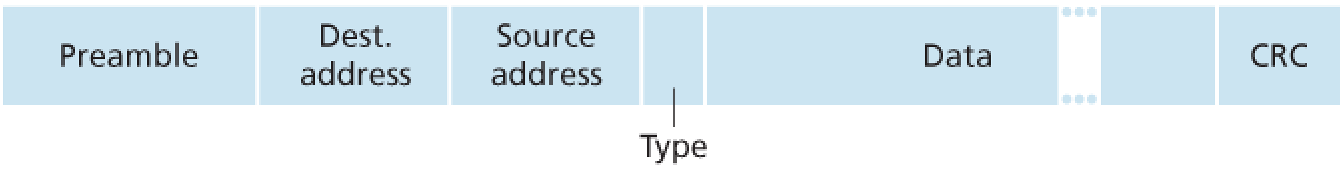


Figure 6.20 Ethernet frame structure

Ethernet Frame Structure

We can learn a lot about Ethernet by examining the Ethernet frame, which is shown in **Figure 6.20**. To give this discussion about Ethernet frames a tangible context, let's consider sending an IP datagram from one host to another host, with both hosts on the same Ethernet LAN (for example, the Ethernet LAN in **Figure 6.17**.) (Although the payload of our Ethernet frame is an IP datagram, we note that an Ethernet frame can carry other network-layer packets as well.) Let the sending adapter, adapter A, have the MAC address AA-AA-AA-AA-AA-AA and the receiving adapter, adapter B, have the MAC address BB-BB-BB-BB-BB-BB. The sending adapter encapsulates the IP datagram within an Ethernet frame and passes the frame to the physical layer. The receiving adapter receives the frame from the physical layer, extracts the IP datagram, and passes the IP datagram to the network layer. In this context, let's now examine the six fields of the Ethernet frame, as shown in **Figure 6.20**.

- **Data field (46 to 1,500 bytes).** This field carries the IP datagram. The maximum transmission unit (MTU) of Ethernet is 1,500 bytes. This means that if the IP datagram exceeds 1,500 bytes, then the host has to fragment the datagram, as discussed in **Section 4.3.2**. The minimum size of the data field is 46 bytes. This means that if the IP datagram is less than 46 bytes, the data field has to be "stuffed" to fill it out to 46 bytes. When stuffing is used, the data passed to the network layer contains the stuffing as well as an IP datagram. The network layer uses the length field in the IP datagram header to remove the stuffing.
- **Destination address (6 bytes).** This field contains the MAC address of the destination adapter, BB-BB-BB-BB-BB-BB. When adapter B receives an Ethernet frame whose destination address is either BB-BB-BB-BB-BB-BB or the MAC broadcast address, it passes the contents of the frame's data field to the network layer; if it receives a frame with any other MAC address, it discards the frame.
- **Source address (6 bytes).** This field contains the MAC address of the adapter that transmits the frame onto the LAN, in this example, AA-AA-AA-AA-AA-AA.
- **Type field (2 bytes).** The type field permits Ethernet to multiplex network-layer protocols. To understand this, we need to keep in mind that hosts can use other network-layer protocols besides IP. In fact, a given host may support multiple network-layer protocols using different protocols for different applications. For this reason, when the Ethernet frame arrives at adapter B, adapter B needs to know to which network-layer protocol it should pass (that is, demultiplex) the contents of the data field. IP and other network-layer protocols (for example, Novell IPX or AppleTalk) each have their own, standardized type number. Furthermore, the ARP protocol (discussed in the previous

section) has its own type number, and if the arriving frame contains an ARP packet (i.e., has a type field of 0806 hexadecimal), the ARP packet will be demultiplexed up to the ARP protocol. Note that the type field is analogous to the protocol field in the network-layer datagram and the port-number fields in the transport-layer segment; all of these fields serve to glue a protocol at one layer to a protocol at the layer above.

- **Cyclic redundancy check (CRC) (4 bytes).** As discussed in **Section 6.2.3**, the purpose of the CRC field is to allow the receiving adapter, adapter B, to detect bit errors in the frame.
- **Preamble (8 bytes).** The Ethernet frame begins with an 8-byte preamble field. Each of the first 7 bytes of the preamble has a value of 10101010; the last byte is 10101011. The first 7 bytes of the preamble serve to "wake up" the receiving adapters and to synchronize their clocks to that of the sender's clock. Why should the clocks be out of synchronization? Keep in mind that adapter A aims to transmit the frame at 10 Mbps, 100 Mbps, or 1 Gbps, depending on the type of Ethernet LAN. However, because nothing is absolutely perfect, adapter A will not transmit the frame at exactly the target rate; there will always be some *drift* from the target rate, a drift which is not known *a priori* by the other adapters on the LAN. A receiving adapter can lock onto adapter A's clock simply by locking onto the bits in the first 7 bytes of the preamble. The last 2 bits of the eighth byte of the preamble (the first two consecutive 1s) alert adapter B that the "important stuff" is about to come.

All of the Ethernet technologies provide connectionless service to the network layer. That is, when adapter A wants to send a datagram to adapter B, adapter A encapsulates the datagram in an Ethernet frame and sends the frame into the LAN, without first handshaking with adapter B. This layer-2 connectionless service is analogous to IP's layer-3 datagram service and UDP's layer-4 connectionless service.

Ethernet technologies provide an unreliable service to the network layer. Specifically, when adapter B receives a frame from adapter A, it runs the frame through a CRC check, but neither sends an acknowledgment when a frame passes the CRC check nor sends a negative acknowledgment when a frame fails the CRC check. When a frame fails the CRC check, adapter B simply discards the frame. Thus, adapter A has no idea whether its transmitted frame reached adapter B and passed the CRC check. This lack of reliable transport (at the link layer) helps to make Ethernet simple and cheap. But it also means that the stream of datagrams passed to the network layer can have gaps.

> CASE HISTORY
>
> BOB METCALFE AND ETHERNET
>
> As a PhD student at Harvard University in the early 1970s, Bob Metcalfe worked on the ARPAnet at MIT. During his studies, he also became exposed to Abramson's work on ALOHA and random access protocols. After completing his PhD and just before beginning a job at Xerox Palo Alto Research Center (Xerox PARC), he visited Abramson and his University of Hawaii colleagues for three months, getting a firsthand look at ALOHAnet. At Xerox PARC, Metcalfe

> became exposed to Alto computers, which in many ways were the forerunners of the personal computers of the 1980s. Metcalfe saw the need to network these computers in an inexpensive manner. So armed with his knowledge about ARPAnet, ALOHAnet, and random access protocols, Metcalfe—along with colleague David Boggs—invented Ethernet.
>
> Metcalfe and Boggs's original Ethernet ran at 2.94 Mbps and linked up to 256 hosts separated by up to one mile. Metcalfe and Boggs succeeded at getting most of the researchers at Xerox PARC to communicate through their Alto computers. Metcalfe then forged an alliance between Xerox, Digital, and Intel to establish Ethernet as a 10 Mbps Ethernet standard, ratified by the IEEE. Xerox did not show much interest in commercializing Ethernet. In 1979, Metcalfe formed his own company, 3Com, which developed and commercialized networking technology, including Ethernet technology. In particular, 3Com developed and marketed Ethernet cards in the early 1980s for the immensely popular IBM PCs.

If there are gaps due to discarded Ethernet frames, does the application at Host B see gaps as well? As we learned in **Chapter 3**, this depends on whether the application is using UDP or TCP. If the application is using UDP, then the application in Host B will indeed see gaps in the data. On the other hand, if the application is using TCP, then TCP in Host B will not acknowledge the data contained in discarded frames, causing TCP in Host A to retransmit. Note that when TCP retransmits data, the data will eventually return to the Ethernet adapter at which it was discarded. Thus, in this sense, Ethernet does retransmit data, although Ethernet is unaware of whether it is transmitting a brand-new datagram with brand-new data, or a datagram that contains data that has already been transmitted at least once.

Ethernet Technologies

In our discussion above, we've referred to Ethernet as if it were a single protocol standard. But in fact, Ethernet comes in *many* different flavors, with somewhat bewildering acronyms such as 10BASE-T, 10BASE-2, 100BASE-T, 1000BASE-LX, 10GBASE-T and 40GBASE-T. These and many other Ethernet technologies have been standardized over the years by the IEEE 802.3 CSMA/CD (Ethernet) working group **[IEEE 802.3 2012]**. While these acronyms may appear bewildering, there is actually considerable order here. The first part of the acronym refers to the speed of the standard: 10, 100, 1000, or 10G, for 10 Megabit (per second), 100 Megabit, Gigabit, 10 Gigabit and 40 Gigibit Ethernet, respectively. "BASE" refers to baseband Ethernet, meaning that the physical media only carries Ethernet traffic; almost all of the 802.3 standards are for baseband Ethernet. The final part of the acronym refers to the physical media itself; Ethernet is both a link-layer *and* a physical-layer specification and is carried over a variety of physical media including coaxial cable, copper wire, and fiber. Generally, a "T" refers to twisted-pair copper wires.

Historically, an Ethernet was initially conceived of as a segment of coaxial cable. The early 10BASE-2 and 10BASE-5 standards specify 10 Mbps Ethernet over two types of coaxial cable, each limited in length to 500 meters. Longer runs could be obtained by using a **repeater**—a physical-layer device that receives a signal on the input side, and regenerates the signal on the output side. A coaxial cable corresponds nicely to our view of Ethernet as a broadcast medium—all frames transmitted by one interface are received at other interfaces, and Ethernet's CDMA/CD protocol nicely solves the multiple access problem. Nodes simply attach to the cable, and *voila*, we have a local area network!

Ethernet has passed through a series of evolutionary steps over the years, and today's Ethernet is very different from the original bus-topology designs using coaxial cable. In most installations today, nodes are connected to a switch via point-to-point segments made of twisted-pair copper wires or fiber-optic cables, as shown in **Figures 6.15**–**6.17**.

In the mid-1990s, Ethernet was standardized at 100 Mbps, 10 times faster than 10 Mbps Ethernet. The original Ethernet MAC protocol and frame format were preserved, but higher-speed physical layers were defined for copper wire (100BASE-T) and fiber (100BASE-FX, 100BASE-SX, 100BASE-BX). **Figure 6.21** shows these different standards and the common Ethernet MAC protocol and frame format. 100 Mbps Ethernet is limited to a 100-meter distance over twisted pair, and to

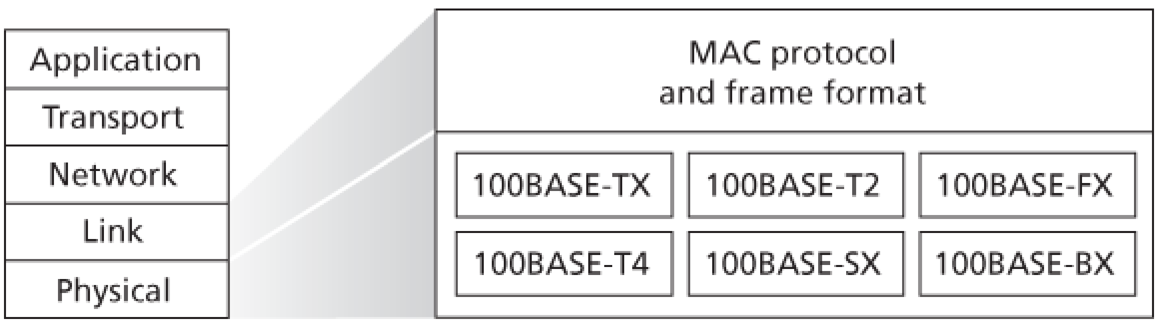


Figure 6.21 100 Mbps Ethernet standards: A common link layer, different physical layers

several kilometers over fiber, allowing Ethernet switches in different buildings to be connected.

Gigabit Ethernet is an extension to the highly successful 10 Mbps and 100 Mbps Ethernet standards. Offering a raw data rate of 40,000 Mbps, 40 Gigabit Ethernet maintains full compatibility with the huge installed base of Ethernet equipment. The standard for Gigabit Ethernet, referred to as IEEE 802.3z, does the following:

- Uses the standard Ethernet frame format (**Figure 6.20**) and is backward compatible with 10BASE-T and 100BASE-T technologies. This allows for easy integration of Gigabit Ethernet with the existing installed base of Ethernet equipment.
- Allows for point-to-point links as well as shared broadcast channels. Point-to-point links use switches while broadcast channels use hubs, as described earlier. In Gigabit Ethernet jargon, hubs are called *buffered distributors*.
- Uses CSMA/CD for shared broadcast channels. In order to have acceptable efficiency, the

maximum distance between nodes must be severely restricted.

- Allows for full-duplex operation at 40 Gbps in both directions for point-to-point channels.

Initially operating over optical fiber, Gigabit Ethernet is now able to run over category 5 UTP cabling.

Let's conclude our discussion of Ethernet technology by posing a question that may have begun troubling you. In the days of bus topologies and hub-based star topologies, Ethernet was clearly a broadcast link (as defined in **Section 6.3**) in which frame collisions occurred when nodes transmitted at the same time. To deal with these collisions, the Ethernet standard included the CSMA/CD protocol, which is particularly effective for a wired broadcast LAN spanning a small geographical region. But if the prevalent use of Ethernet today is a switch-based star topology, using store-and-forward packet switching, is there really a need anymore for an Ethernet MAC protocol? As we'll see shortly, a switch coordinates its transmissions and never forwards more than one frame onto the same interface at any time. Furthermore, modern switches are full-duplex, so that a switch and a node can each send frames to each other at the same time without interference. In other words, in a switch-based Ethernet LAN there are no collisions and, therefore, there is no need for a MAC protocol!

As we've seen, today's Ethernets are *very* different from the original Ethernet conceived by Metcalfe and Boggs more than 30 years ago—speeds have increased by three orders of magnitude, Ethernet frames are carried over a variety of media, switched-Ethernets have become dominant, and now even the MAC protocol is often unnecessary! Is all of this *really* still Ethernet? The answer, of course, is "yes, by definition." It is interesting to note, however, that through all of these changes, there has indeed been one enduring constant that has remained unchanged over 30 years—Ethernet's frame format. Perhaps this then is the one true and timeless centerpiece of the Ethernet standard.

6.4.3 Link-Layer Switches

Up until this point, we have been purposefully vague about what a switch actually does and how it works. The role of the switch is to receive incoming link-layer frames and forward them onto outgoing links; we'll study this forwarding function in detail in this subsection. We'll see that the switch itself is **transparent** to the hosts and routers in the subnet; that is, a host/router addresses a frame to another host/router (rather than addressing the frame to the switch) and happily sends the frame into the LAN, unaware that a switch will be receiving the frame and forwarding it. The rate at which frames arrive to any one of the switch's output interfaces may temporarily exceed the link capacity of that interface. To accommodate this problem, switch output interfaces have buffers, in much the same way that router output interfaces have buffers for datagrams. Let's now take a closer look at how switches operate.

Forwarding and Filtering

Filtering is the switch function that determines whether a frame should be forwarded to some interface or should just be dropped. **Forwarding** is the switch function that determines the interfaces to which a frame should be directed, and then moves the frame to those interfaces. Switch filtering and forwarding are done with a **switch table**. The switch table contains entries for some, but not necessarily all, of the hosts and routers on a LAN. An entry in the switch table contains (1) a MAC address, (2) the switch interface that leads toward that MAC address, and (3) the time at which the entry was placed in the table. An example switch table for the uppermost switch in **Figure 6.15** is shown in **Figure 6.22**. This description of frame forwarding may sound similar to our discussion of datagram forwarding

Address	Interface	Time
62-FE-F7-11-89-A3	1	9:32
7C-BA-B2-B4-91-10	3	9:36
....	....	....

Figure 6.22 Portion of a switch table for the uppermost switch in Figure 6.15

in **Chapter 4**. Indeed, in our discussion of generalized forwarding in **Section 4.4**, we learned that many modern packet switches can be configured to forward on the basis of layer-2 destination MAC addresses (i.e., function as a layer-2 switch) or layer-3 IP destination addresses (i.e., function as a layer-3 router). Nonetheless, we'll make the important distinction that switches forward packets based on MAC addresses rather than on IP addresses. We will also see that a traditional (i.e., in a non-SDN context) switch table is constructed in a very different manner from a router's forwarding table.

To understand how switch filtering and forwarding work, suppose a frame with destination address DD-DD-DD-DD-DD-DD arrives at the switch on interface *x*. The switch indexes its table with the MAC address DD-DD-DD-DD-DD-DD. There are three possible cases:

- There is no entry in the table for DD-DD-DD-DD-DD-DD. In this case, the switch forwards copies of the frame to the output buffers preceding *all* interfaces except for interface *x*. In other words, if there is no entry for the destination address, the switch broadcasts the frame.
- There is an entry in the table, associating DD-DD-DD-DD-DD-DD with interface *x*. In this case, the frame is coming from a LAN segment that contains adapter DD-DD-DD-DD-DD-DD. There being no need to forward the frame to any of the other interfaces, the switch performs the filtering function by discarding the frame.
- There is an entry in the table, associating DD-DD-DD-DD-DD-DD with interface $y \neq x$. In this case, the frame needs to be forwarded to the LAN segment attached to interface *y*. The switch performs its forwarding function by putting the frame in an output buffer that precedes interface *y*.

Let's walk through these rules for the uppermost switch in **Figure 6.15** and its switch table in **Figure 6.22**. Suppose that a frame with destination address 62-FE-F7-11-89-A3 arrives at the switch from interface 1. The switch examines its table and sees that the destination is on the LAN segment connected to interface 1 (that is, Electrical Engineering). This means that the frame has already been broadcast on the LAN segment that contains the destination. The switch therefore filters (that is, discards) the frame. Now suppose a frame with the same destination address arrives from interface 2. The switch again examines its table and sees that the destination is in the direction of interface 1; it therefore forwards the frame to the output buffer preceding interface 1. It should be clear from this example that as long as the switch table is complete and accurate, the switch forwards frames toward destinations without any broadcasting.

In this sense, a switch is "smarter" than a hub. But how does this switch table get configured in the first place? Are there link-layer equivalents to network-layer routing protocols? Or must an overworked manager manually configure the switch table?

Self-Learning

A switch has the wonderful property (particularly for the already-overworked network administrator) that its table is built automatically, dynamically, and autonomously—without any intervention from a network administrator or from a configuration protocol. In other words, switches are **self-learning**. This capability is accomplished as follows:

1. The switch table is initially empty.
2. For each incoming frame received on an interface, the switch stores in its table (1) the MAC address in the frame's *source address field*, (2) the interface from which the frame arrived, and (3) the current time. In this manner the switch records in its table the LAN segment on which the sender resides. If every host in the LAN eventually sends a frame, then every host will eventually get recorded in the table.
3. The switch deletes an address in the table if no frames are received with that address as the source address after some period of time (the **aging time**). In this manner, if a PC is replaced by another PC (with a different adapter), the MAC address of the original PC will eventually be purged from the switch table.

Let's walk through the self-learning property for the uppermost switch in **Figure 6.15** and its corresponding switch table in **Figure 6.22**. Suppose at time 9:39 a frame with source address 01-12-23-34-45-56 arrives from interface 2. Suppose that this address is not in the switch table. Then the switch adds a new entry to the table, as shown in **Figure 6.23**.

Continuing with this same example, suppose that the aging time for this switch is 60 minutes, and no frames with source address 62-FE-F7-11-89-A3 arrive to the switch between 9:32 and 10:32. Then at time 10:32, the switch removes this address from its table.

Address	Interface	Time
01-12-23-34-45-56	2	9:39
62-FE-F7-11-89-A3	1	9:32
7C-BA-B2-B4-91-10	3	9:36
....	....	....

Figure 6.23 Switch learns about the location of an adapter with address 01-12-23-34-45-56

Switches are **plug-and-play devices** because they require no intervention from a network administrator or user. A network administrator wanting to install a switch need do nothing more than connect the LAN segments to the switch interfaces. The administrator need not configure the switch tables at the time of installation or when a host is removed from one of the LAN segments. Switches are also full-duplex, meaning any switch interface can send and receive at the same time.

Properties of Link-Layer Switching

Having described the basic operation of a link-layer switch, let's now consider their features and properties. We can identify several advantages of using switches, rather than broadcast links such as buses or hub-based star topologies:

- **Elimination of collisions.** In a LAN built from switches (and without hubs), there is no wasted bandwidth due to collisions! The switches buffer frames and never transmit more than one frame on a segment at any one time. As with a router, the maximum aggregate throughput of a switch is the sum of all the switch interface rates. Thus, switches provide a significant performance improvement over LANs with broadcast links.
- **Heterogeneous links.** Because a switch isolates one link from another, the different links in the LAN can operate at different speeds and can run over different media. For example, the uppermost switch in **Figure 6.15** might have three1 Gbps 1000BASE-T copper links, two 100 Mbps 100BASE-FX fiber links, and one 100BASE-T copper link. Thus, a switch is ideal for mixing legacy equipment with new equipment.
- **Management.** In addition to providing enhanced security (see sidebar on Focus on Security), a switch also eases network management. For example, if an adapter malfunctions and continually sends Ethernet frames (called a jabbering adapter), a switch can detect the problem and internally disconnect the malfunctioning adapter. With this feature, the network administrator need not get out of bed and drive back to work in order to correct the problem. Similarly, a cable cut disconnects only that host that was using the cut cable to connect to the switch. In the days of coaxial cable, many a

network manager spent hours "walking the line" (or more accurately, "crawling the floor") to find the cable break that brought down the entire network. Switches also gather statistics on bandwidth usage, collision rates, and traffic types, and make this information available to the network manager. This information can be used to debug and correct problems, and to plan how the LAN should evolve in the future. Researchers are exploring adding yet more management functionality into Ethernet LANs in prototype deployments **[Casado 2007**; **Koponen 2011]**.

FOCUS ON SECURITY

SNIFFING A SWITCHED LAN: SWITCH POISONING

When a host is connected to a switch, it typically only receives frames that are intended for it. For example, consider a switched LAN in **Figure 6.17**. When host A sends a frame to host B, and there is an entry for host B in the switch table, then the switch will forward the frame *only* to host B. If host C happens to be running a sniffer, host C will not be able to sniff this A-to-B frame. Thus, in a switched-LAN environment (in contrast to a broadcast link environment such as 802.11 LANs or hub–based Ethernet LANs), it is more difficult for an attacker to sniff frames. *However*, because the switch broadcasts frames that have destination addresses that are not in the switch table, the sniffer at C can still sniff some frames that are not intended for C. Furthermore, a sniffer will be able sniff all Ethernet broadcast frames with broadcast destination address FF–FF–FF–FF–FF–FF. A well-known attack against a switch, called **switch poisoning**, is to send tons of packets to the switch with many different bogus source MAC addresses, thereby filling the switch table with bogus entries and leaving no room for the MAC addresses of the legitimate hosts. This causes the switch to broadcast most frames, which can then be picked up by the sniffer **[Skoudis 2006]**. As this attack is rather involved even for a sophisticated attacker, switches are significantly less vulnerable to sniffing than are hubs and wireless LANs.

Switches Versus Routers

As we learned in **Chapter 4**, routers are store-and-forward packet switches that forward packets using network-layer addresses. Although a switch is also a store-and-forward packet switch, it is fundamentally different from a router in that it forwards packets using MAC addresses. Whereas a router is a layer-3 packet switch, a switch is a layer-2 packet switch. Recall, however, that we learned in **Section 4.4** that modern switches using the "match plus action" operation can be used to forward a layer-2 frame based on the frame's destination MAC address, as well as a layer-3 datagram using the datagram's destination IP address. Indeed, we saw that switches using the OpenFlow standard can perform generalized packet forwarding based on any of eleven different frame, datagram, and transport-layer header fields.

Even though switches and routers are fundamentally different, network administrators must often choose between them when installing an interconnection device. For example, for the network in **Figure 6.15**, the network administrator could just as easily have used a router instead of a switch to connect the department LANs, servers, and internet gateway router. Indeed, a router would permit interdepartmental communication without creating collisions. Given that both switches and routers are candidates for interconnection devices, what are the pros and cons of the two approaches?

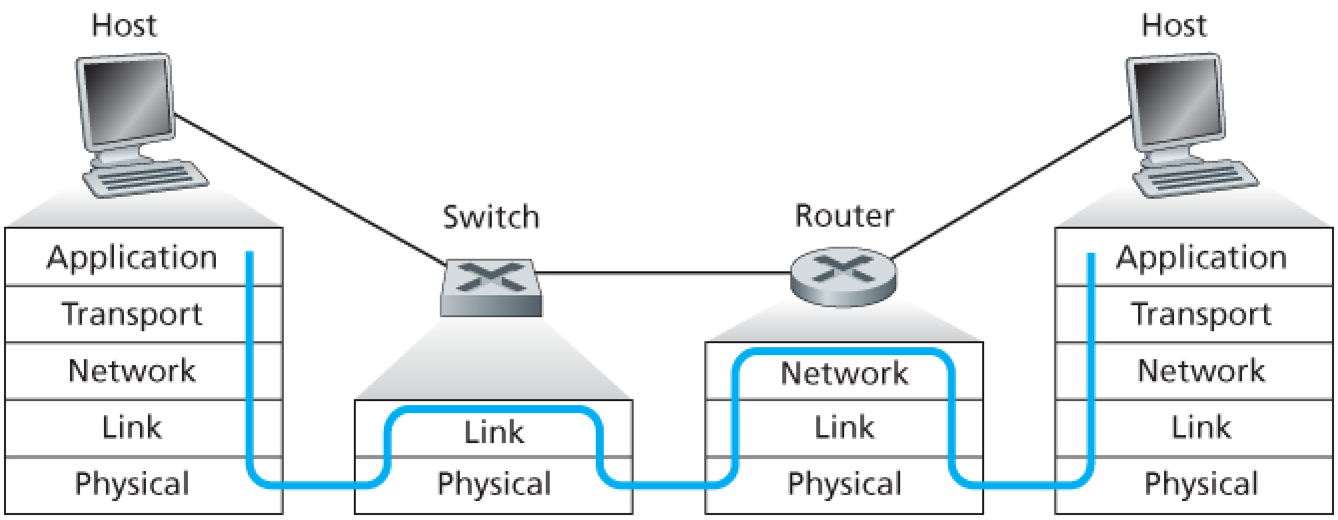


Figure 6.24 Packet processing in switches, routers, and hosts

First consider the pros and cons of switches. As mentioned above, switches are plug-and-play, a property that is cherished by all the overworked network administrators of the world. Switches can also have relatively high filtering and forwarding rates—as shown in **Figure 6.24**, switches have to process frames only up through layer 2, whereas routers have to process datagrams up through layer 3. On the other hand, to prevent the cycling of broadcast frames, the active topology of a switched network is restricted to a spanning tree. Also, a large switched network would require large ARP tables in the hosts and routers and would generate substantial ARP traffic and processing. Furthermore, switches are susceptible to broadcast storms—if one host goes haywire and transmits an endless stream of Ethernet broadcast frames, the switches will forward all of these frames, causing the entire network to collapse.

Now consider the pros and cons of routers. Because network addressing is often hierarchical (and not flat, as is MAC addressing), packets do not normally cycle through routers even when the network has redundant paths. (However, packets can cycle when router tables are misconfigured; but as we learned in **Chapter 4**, IP uses a special datagram header field to limit the cycling.) Thus, packets are not restricted to a spanning tree and can use the best path between source and destination. Because routers do not have the spanning tree restriction, they have allowed the Internet to be built with a rich topology that includes, for example, multiple active links between Europe and North America. Another feature of routers is that they provide firewall protection against layer-2 broadcast storms. Perhaps the most significant drawback of routers, though, is that they are not plug-and-play—they and the hosts that connect to them need their IP addresses to be configured. Also, routers often have a larger per-packet processing time than switches, because they have to process up through the layer-3 fields. Finally, there are two different ways to pronounce the word *router*, either as "rootor" or as "rowter," and people waste a lot of time arguing over the proper pronunciation **[Perlman 1999]**.

Given that both switches and routers have their pros and cons (as summarized in **Table 6.1**), when should an institutional network (for example, a university campus

Table 6.1 Comparison of the typical features of popular interconnection devices

	Hubs	Routers	Switches
Traffic isolation	No	Yes	Yes
Plug and play	Yes	No	Yes
Optimal routing	No	Yes	No

network or a corporate campus network) use switches, and when should it use routers? Typically, small networks consisting of a few hundred hosts have a few LAN segments. Switches suffice for these small networks, as they localize traffic and increase aggregate throughput without requiring any configuration of IP addresses. But larger networks consisting of thousands of hosts typically include routers within the network (in addition to switches). The routers provide a more robust isolation of traffic, control broadcast storms, and use more "intelligent" routes among the hosts in the network.

For more discussion of the pros and cons of switched versus routed networks, as well as a discussion of how switched LAN technology can be extended to accommodate two orders of magnitude more hosts than today's Ethernets, see **[Meyers 2004**; **Kim 2008]**.

6.4.4 Virtual Local Area Networks (VLANs)

In our earlier discussion of **Figure 6.15**, we noted that modern institutional LANs are often configured hierarchically, with each workgroup (department) having its own switched LAN connected to the switched LANs of other groups via a switch hierarchy. While such a configuration works well in an ideal world, the real world is often far from ideal. Three drawbacks can be identified in the configuration in **Figure 6.15**:

- **Lack of traffic isolation.** Although the hierarchy localizes group traffic to within a single switch, broadcast traffic (e.g., frames carrying ARP and DHCP messages or frames whose destination has not yet been learned by a self-learning switch) must still traverse the entire institutional network.

Limiting the scope of such broadcast traffic would improve LAN performance. Perhaps more importantly, it also may be desirable to limit LAN broadcast traffic for security/privacy reasons. For example, if one group contains the company's executive management team and another group contains disgruntled employees running Wireshark packet sniffers, the network manager may well prefer that the executives' traffic never even reaches employee hosts. This type of isolation could be provided by replacing the center switch in **Figure 6.15** with a router. We'll see shortly that this isolation also can be achieved via a switched (layer 2) solution.

- **Inefficient use of switches.** If instead of three groups, the institution had 10 groups, then 10 first-level switches would be required. If each group were small, say less than 10 people, then a single 96-port switch would likely be large enough to accommodate everyone, but this single switch would not provide traffic isolation.
- **Managing users.** If an employee moves between groups, the physical cabling must be changed to connect the employee to a different switch in **Figure 6.15**. Employees belonging to two groups make the problem even harder.

Fortunately, each of these difficulties can be handled by a switch that supports **virtual local area networks (VLANs)**. As the name suggests, a switch that supports VLANs allows multiple *virtual* local area networks to be defined over a single *physical* local area network infrastructure. Hosts within a VLAN communicate with each other as if they (and no other hosts) were connected to the switch. In a port-based VLAN, the switch's ports (interfaces) are divided into groups by the network manager. Each group constitutes a VLAN, with the ports in each VLAN forming a broadcast domain (i.e., broadcast traffic from one port can only reach other ports in the group). **Figure 6.25** shows a single switch with 16 ports. Ports 2 to 8 belong to the EE VLAN, while ports 9 to 15 belong to the CS VLAN (ports 1 and 16 are unassigned). This VLAN solves all of the difficulties noted above—EE and CS VLAN frames are isolated from each other, the two switches in **Figure 6.15** have been replaced by a single switch, and if the user at switch port 8 joins the CS Department, the network operator simply reconfigures the VLAN software so that port 8 is now associated with the CS VLAN. One can easily imagine how the VLAN switch is configured and operates—the network manager declares a port to belong

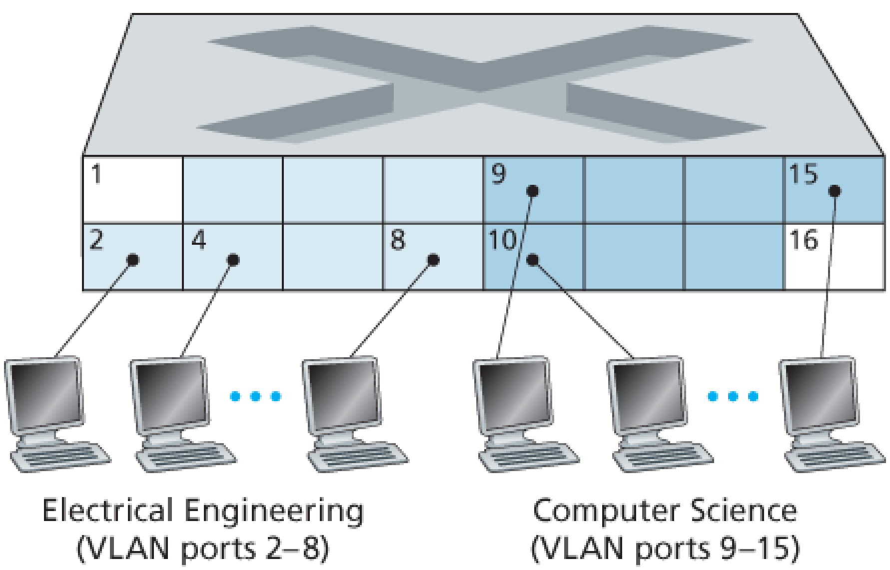


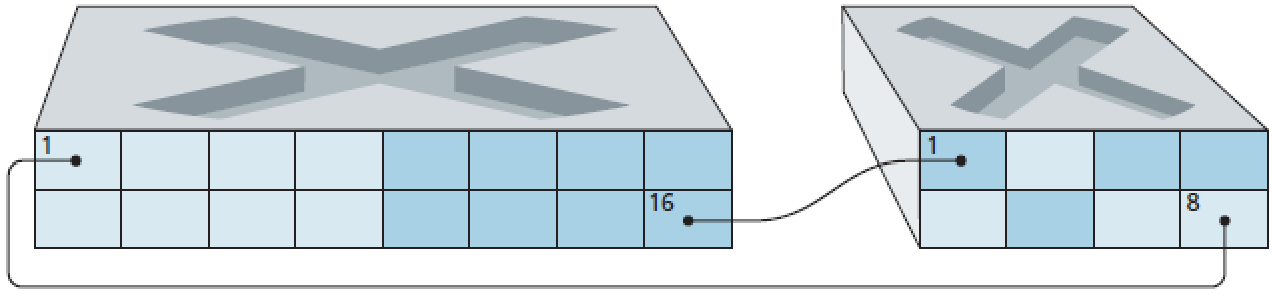
Figure 6.25 A single switch with two configured VLANs

to a given VLAN (with undeclared ports belonging to a default VLAN) using switch management software, a table of port-to-VLAN mappings is maintained within the switch; and switch hardware only delivers frames between ports belonging to the same VLAN.

But by completely isolating the two VLANs, we have introduced a new difficulty! How can traffic from the EE Department be sent to the CS Department? One way to handle this would be to connect a VLAN switch port (e.g., port 1 in **Figure 6.25**) to an external router and configure that port to belong both the EE and CS VLANs. In this case, even though the EE and CS departments share the same physical switch, the logical configuration would look as if the EE and CS departments had separate switches connected via a router. An IP datagram going from the EE to the CS department would first cross the EE VLAN to reach the router and then be forwarded by the router back over the CS VLAN to the CS host. Fortunately, switch vendors make such configurations easy for the network manager by building a single device that contains both a VLAN switch *and* a router, so a separate external router is not needed. A homework problem at the end of the chapter explores this scenario in more detail.

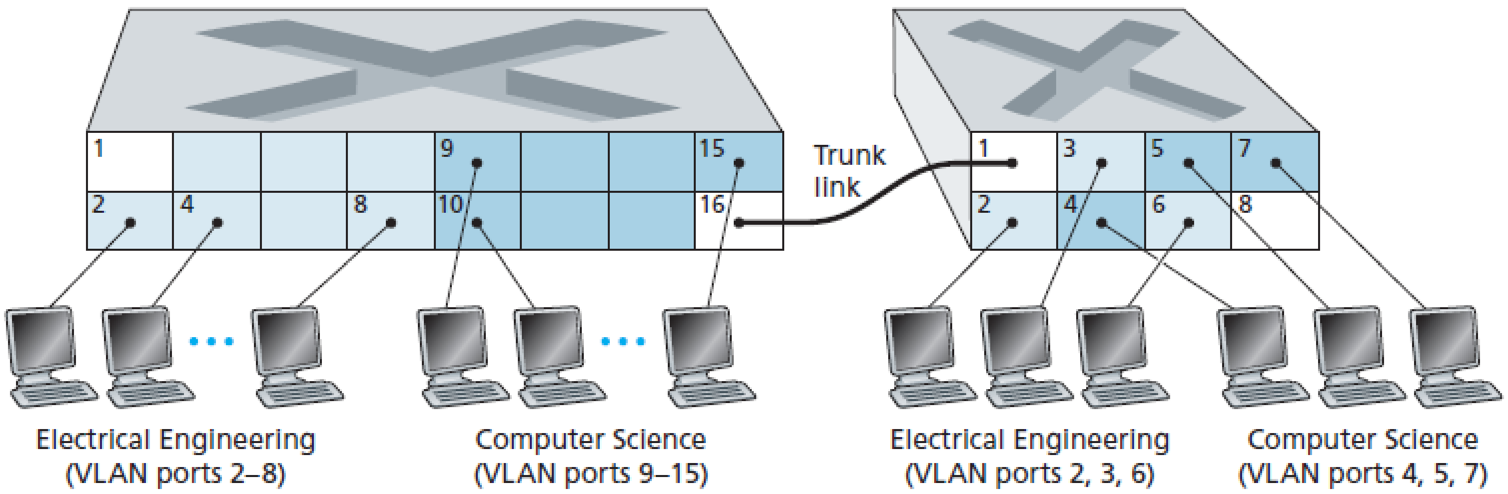
Returning again to **Figure 6.15**, let's now suppose that rather than having a separate Computer Engineering department, some EE and CS faculty are housed in a separate building, where (of course!) they need network access, and (of course!) they'd like to be part of their department's VLAN. **Figure 6.26** shows a second 8-port switch, where the switch ports have been defined as belonging to the EE or the CS VLAN, as needed. But how should these two switches be interconnected? One easy solution would be to define a port belonging to the CS VLAN on each switch (similarly for the EE VLAN) and to connect these ports to each other, as shown in **Figure 6.26(a)**. This solution doesn't scale, however, since *N* VLANS would require *N* ports on each switch simply to interconnect the two switches.

A more scalable approach to interconnecting VLAN switches is known as **VLAN trunking**. In the VLAN trunking approach shown in **Figure 6.26(b)**, a special port on each switch (port 16 on the left switch and port 1 on the right switch) is configured as a trunk port to interconnect the two VLAN switches. The trunk port belongs to all VLANs, and frames sent to any VLAN are forwarded over the trunk link to the other switch. But this raises yet another question: How does a switch know that a frame arriving on a trunk port belongs to a particular VLAN? The IEEE has defined an extended Ethernet frame format, 802.1Q, for frames crossing a VLAN trunk. As shown in **Figure 6.27**, the 802.1Q frame consists of the standard Ethernet frame with a four-byte **VLAN tag** added into the header that carries the identity of the VLAN to which the frame belongs. The VLAN tag is added into a frame by the switch at the sending side of a VLAN trunk, parsed, and removed by the switch at the receiving side of the trunk. The VLAN tag itself consists of a 2-byte Tag Protocol Identifier (TPID) field (with a fixed hexadecimal value of 81-00), a 2-byte Tag Control Information field that contains a 12-bit VLAN identifier field, and a 3-bit priority field that is similar in intent to the IP datagram TOS field.



a.

Figure 6.26 Connecting two VLAN switches with two VLANs: (a) two cables (b) trunked



b.

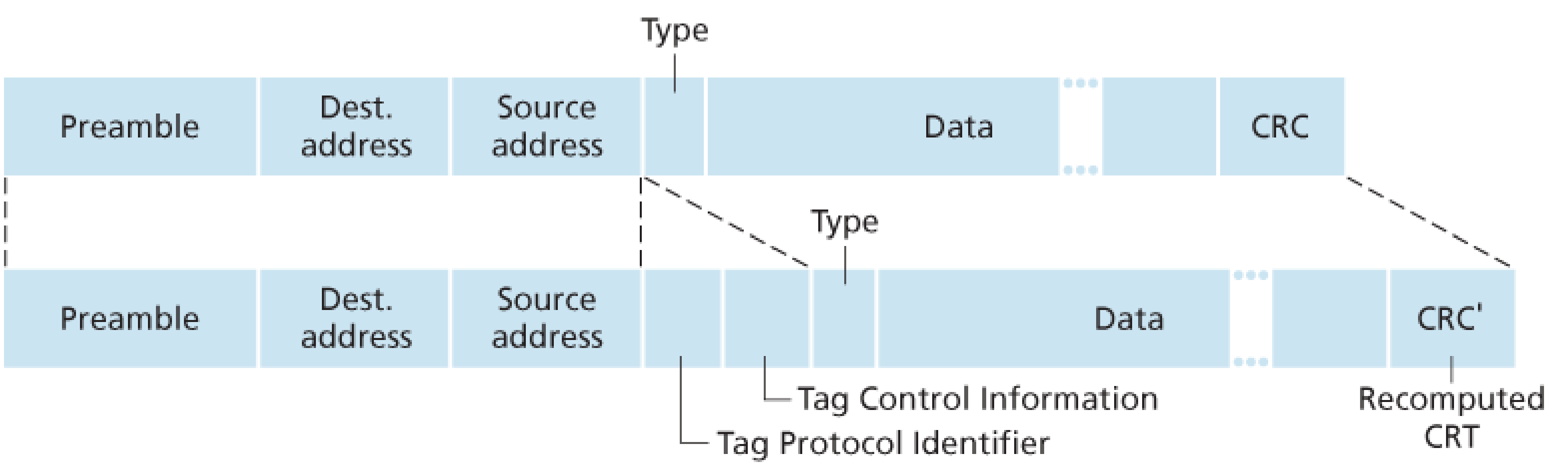


Figure 6.27 Original Ethernet frame (top), 802.1Q-tagged Ethernet VLAN frame (below)

In this discussion, we've only briefly touched on VLANs and have focused on port-based VLANs. We should also mention that VLANs can be defined in several other ways. In MAC-based VLANs, the network manager specifies the set of MAC addresses that belong to each VLAN; whenever a device attaches to a port, the port is connected into the appropriate VLAN based on the MAC address of the device. VLANs can also be defined based on network-layer protocols (e.g., IPv4, IPv6, or Appletalk) and other criteria. It is also possible for VLANs to be extended across IP routers, allowing islands of LANs to be connected together to form a single VLAN that could span the globe **[Yu 2011]**. See the 802.1Q standard **[IEEE 802.1q 2005]** for more details.

6.5 Link Virtualization: A Network as a Link Layer

Because this chapter concerns link-layer protocols, and given that we're now nearing the chapter's end, let's reflect on how our understanding of the term *link* has evolved. We began this chapter by viewing the link as a physical wire connecting two communicating hosts. In studying multiple access protocols, we saw that multiple hosts could be connected by a shared wire and that the "wire" connecting the hosts could be radio spectra or other media. This led us to consider the link a bit more abstractly as a channel, rather than as a wire. In our study of Ethernet LANs (**Figure 6.15**) we saw that the interconnecting media could actually be a rather complex switched infrastructure. Throughout this evolution, however, the hosts themselves maintained the view that the interconnecting medium was simply a link-layer channel connecting two or more hosts. We saw, for example, that an Ethernet host can be blissfully unaware of whether it is connected to other LAN hosts by a single short LAN segment (**Figure 6.17**) or by a geographically dispersed switched LAN (**Figure 6.15**) or by a VLAN (**Figure 6.26**).

In the case of a dialup modem connection between two hosts, the link connecting the two hosts is actually the telephone network—a logically separate, global telecommunications network with its own switches, links, and protocol stacks for data transfer and signaling. From the Internet link-layer point of view, however, the dial-up connection through the telephone network is viewed as a simple "wire." In this sense, the Internet virtualizes the telephone network, viewing the telephone network as a link-layer technology providing link-layer connectivity between two Internet hosts. You may recall from our discussion of overlay networks in **Chapter 2** that an overlay network similarly views the Internet as a means for providing connectivity between overlay nodes, seeking to overlay the Internet in the same way that the Internet overlays the telephone network.

In this section, we'll consider Multiprotocol Label Switching (MPLS) networks. Unlike the circuit-switched telephone network, MPLS is a packet-switched, virtual-circuit network in its own right. It has its own packet formats and forwarding behaviors. Thus, from a pedagogical viewpoint, a discussion of MPLS fits well into a study of either the network layer or the link layer. From an Internet viewpoint, however, we can consider MPLS, like the telephone network and switched-Ethernets, as a link-layer technology that serves to interconnect IP devices. Thus, we'll consider MPLS in our discussion of the link layer. Frame-relay and ATM networks can also be used to interconnect IP devices, though they represent a slightly older (but still deployed) technology and will not be covered here; see the very readable book **[Goralski 1999]** for details. Our treatment of MPLS will be necessarily brief, as entire books could be (and have been) written on these networks. We recommend **[Davie 2000]** for details on MPLS. We'll focus here primarily on how MPLS servers interconnect to IP devices, although we'll dive a bit deeper into the underlying technologies as well.

6.5.1 Multiprotocol Label Switching (MPLS)

Multiprotocol Label Switching (MPLS) evolved from a number of industry efforts in the mid-to-late 1990s to improve the forwarding speed of IP routers by adopting a key concept from the world of virtual-circuit networks: a fixed-length label. The goal was not to abandon the destination-based IP datagram-forwarding infrastructure for one based on fixed-length labels and virtual circuits, but to augment it by selectively labeling datagrams and allowing routers to forward datagrams based on fixed-length labels (rather than destination IP addresses) when possible. Importantly, these techniques work hand-in-hand with IP, using IP addressing and routing. The IETF unified these efforts in the MPLS protocol **[RFC 3031**, **RFC 3032]**, effectively blending VC techniques into a routed datagram network.

Let's begin our study of MPLS by considering the format of a link-layer frame that is handled by an MPLS-capable router. **Figure 6.28** shows that a link-layer frame transmitted between MPLS-capable devices has a small MPLS header added between the layer-2 (e.g., Ethernet) header and layer-3 (i.e., IP) header. RFC 3032 defines the format of the MPLS header for such links; headers are defined for ATM and frame-relayed networks as well in other RFCs. Among the fields in the MPLS

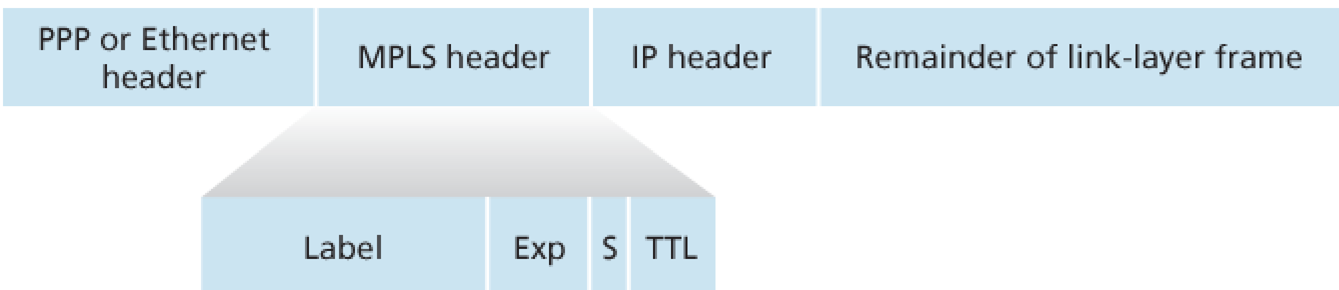


Figure 6.28 MPLS header: Located between link- and network-layer headers

header are the label, 3 bits reserved for experimental use, a single S bit, which is used to indicate the end of a series of "stacked" MPLS headers (an advanced topic that we'll not cover here), and a time-to-live field.

It's immediately evident from **Figure 6.28** that an MPLS-enhanced frame can only be sent between routers that are both MPLS capable (since a non-MPLS-capable router would be quite confused when it found an MPLS header where it had expected to find the IP header!). An MPLS-capable router is often referred to as a **label-switched router**, since it forwards an MPLS frame by looking up the MPLS label in its forwarding table and then immediately passing the datagram to the appropriate output interface. Thus, the MPLS-capable router need *not* extract the destination IP address and perform a lookup of the longest prefix match in the forwarding table. But how does a router know if its neighbor is indeed MPLS capable, and how does a router know what label to associate with the given IP destination? To answer these questions, we'll need to take a look at the interaction among a group of MPLS-capable routers.

In the example in **Figure 6.29**, routers R1 through R4 are MPLS capable. R5 and R6 are standard IP routers. R1 has advertised to R2 and R3 that it (R1) can route to destination A, and that a received frame with MPLS label 6 will be forwarded to destination A. Router R3 has advertised to router R4 that it can route to destinations A and D, and that incoming frames with MPLS labels 10 and 12, respectively, will be switched toward those destinations. Router R2 has also advertised to router R4 that it (R2) can reach destination A, and that a received frame with MPLS label 8 will be switched toward A. Note that router R4 is now in the interesting position of having

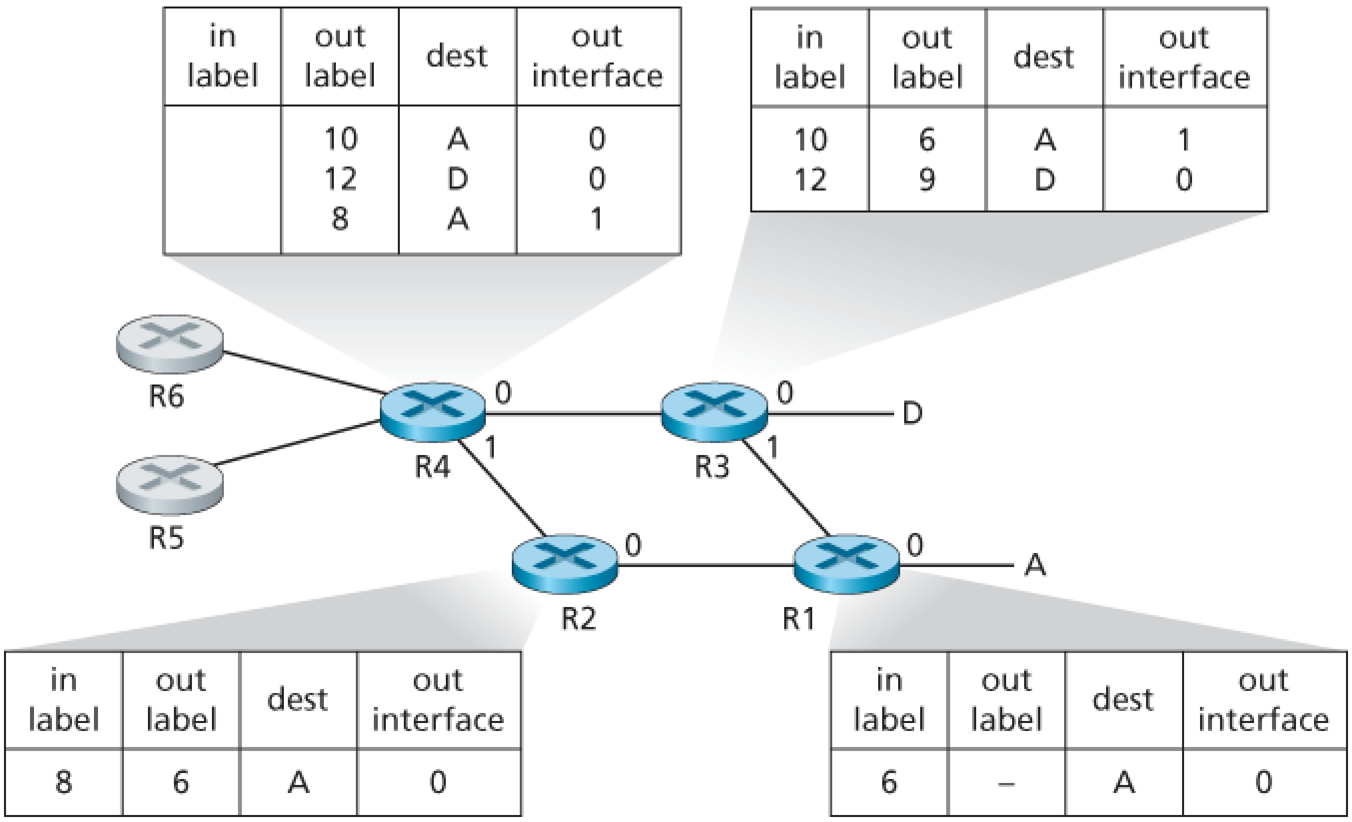


Figure 6.29 MPLS-enhanced forwarding

two MPLS paths to reach A: via interface 0 with outbound MPLS label 10, and via interface 1 with an MPLS label of 8. The broad picture painted in **Figure 6.29** is that IP devices R5, R6, A, and D are connected together via an MPLS infrastructure (MPLS-capable routers R1, R2, R3, and R4) in much the same way that a switched LAN or an ATM network can connect together IP devices. And like a switched LAN or ATM network, the MPLS-capable routers R1 through R4 do so *without ever touching the IP header of a packet*.

In our discussion above, we've not specified the specific protocol used to distribute labels among the MPLS-capable routers, as the details of this signaling are well beyond the scope of this book. We note, however, that the IETF working group on MPLS has specified in **[RFC 3468]** that an extension of the RSVP protocol, known as RSVP-TE **[RFC 3209]**, will be the focus of its efforts for MPLS signaling. We've also not discussed how MPLS actually computes the paths for packets among MPLS capable routers, nor how it gathers link-state information (e.g., amount of link bandwidth unreserved by MPLS) to use in these path computations. Existing link-state routing algorithms (e.g., OSPF) have been extended to flood this information to MPLS-capable routers. Interestingly, the actual path computation algorithms are not standardized, and are currently vendor-specific.

Thus far, the emphasis of our discussion of MPLS has been on the fact that MPLS performs switching based on labels, without needing to consider the IP address of a packet. The true advantages of MPLS and the reason for current interest in MPLS, however, lie not in the potential increases in switching speeds, but rather in the new traffic management capabilities that MPLS enables. As noted above, R4 has *two* MPLS paths to A. If forwarding were performed up at the IP layer on the basis of IP address, the IP routing protocols we studied in **Chapter 5** would specify only a single, least-cost path to A. Thus, MPLS provides the ability to forward packets along routes that would not be possible using standard IP routing protocols. This is one simple form of **traffic engineering** using MPLS **[RFC 3346**; **RFC 3272**; **RFC 2702**; **Xiao 2000]**, in which a network operator can override normal IP routing and force some of the traffic headed toward a given destination along one path, and other traffic destined toward the same destination along another path (whether for policy, performance, or some other reason).

It is also possible to use MPLS for many other purposes as well. It can be used to perform fast restoration of MPLS forwarding paths, e.g., to reroute traffic over a precomputed failover path in response to link failure **[Kar 2000**; **Huang 2002**; **RFC 3469]**. Finally, we note that MPLS can, and has, been used to implement so-called **virtual private networks** (VPNs). In implementing a VPN for a customer, an ISP uses its MPLS-enabled network to connect together the customer’s various networks. MPLS can be used to isolate both the resources and addressing used by the customer’s VPN from that of other users crossing the ISP’s network; see **[DeClercq 2002]** for details.

Our discussion of MPLS has been brief, and we encourage you to consult the references we’ve mentioned. We note that with so many possible uses for MPLS, it appears that it is rapidly becoming the Swiss Army knife of Internet traffic engineering!

6.6 Data Center Networking

In recent years, Internet companies such as Google, Microsoft, Facebook, and Amazon (as well as their counterparts in Asia and Europe) have built massive data centers, each housing tens to hundreds of thousands of hosts, and concurrently supporting many distinct cloud applications (e.g., search, e-mail, social networking, and e-commerce). Each data center has its own **data center network** that interconnects its hosts with each other and interconnects the data center with the Internet. In this section, we provide a brief introduction to data center networking for cloud applications.

The cost of a large data center is huge, exceeding $12 million per month for a 100,000 host data center **[Greenberg 2009a]**. Of these costs, about 45 percent can be attributed to the hosts themselves (which need to be replaced every 3–4 years); 25 percent to infrastructure, including transformers, uninterruptable power supplies (UPS) systems, generators for long-term outages, and cooling systems; 15 percent for electric utility costs for the power draw; and 15 percent for networking, including network gear (switches, routers and load balancers), external links, and transit traffic costs. (In these percentages, costs for equipment are amortized so that a common cost metric is applied for one-time purchases and ongoing expenses such as power.) While networking is not the largest cost, networking innovation is the key to reducing overall cost and maximizing performance **[Greenberg 2009a]**.

The worker bees in a data center are the hosts: They serve content (e.g., Web pages and videos), store e-mails and documents, and collectively perform massively distributed computations (e.g., distributed index computations for search engines). The hosts in data centers, called **blades** and resembling pizza boxes, are generally commodity hosts that include CPU, memory, and disk storage. The hosts are stacked in racks, with each rack typically having 20 to 40 blades. At the top of each rack there is a switch, aptly named the **Top of Rack (TOR) switch**, that interconnects the hosts in the rack with each other and with other switches in the data center. Specifically, each host in the rack has a network interface card that connects to its TOR switch, and each TOR switch has additional ports that can be connected to other switches. Today hosts typically have 40 Gbps Ethernet connections to their TOR switches **[Greenberg 2015]**. Each host is also assigned its own data-center-internal IP address.

The data center network supports two types of traffic: traffic flowing between external clients and internal hosts and traffic flowing between internal hosts. To handle flows between external clients and internal hosts, the data center network includes one or more **border routers**, connecting the data center network to the public Internet. The data center network therefore interconnects the racks with each other and connects the racks to the border routers. **Figure 6.30** shows an example of a data center network. **Data center network design**, the art of designing the interconnection network and protocols that connect the racks with each other and with the border routers, has become an important branch of computer networking research in recent years **[Al-Fares 2008**; **Greenberg 2009a**; **Greenberg 2009b**; **Mysore 2009**; **Guo 2009**; **Wang 2010]**.

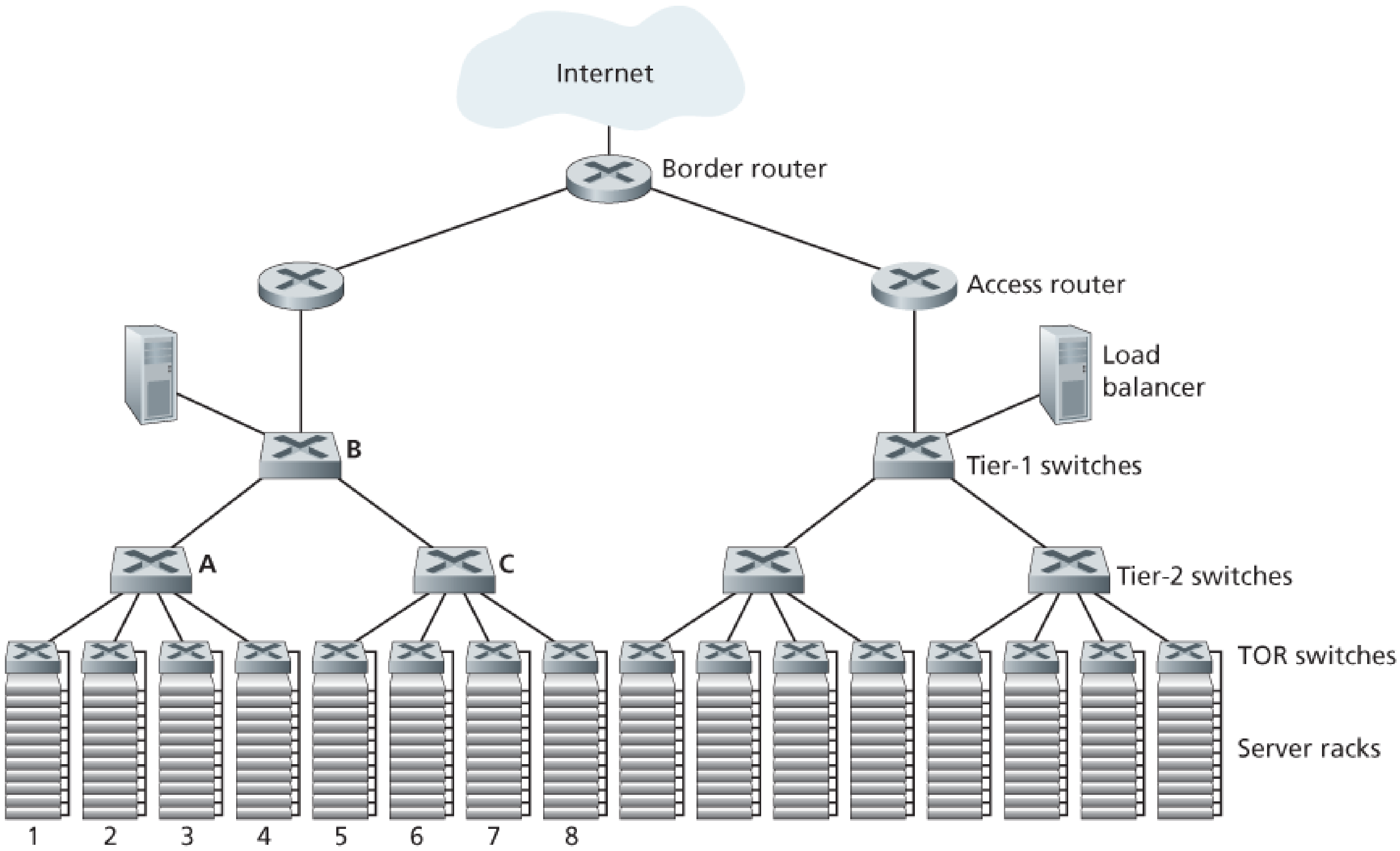


Figure 6.30 A data center network with a hierarchical topology

Load Balancing

A cloud data center, such as a Google or Microsoft data center, provides many applications concurrently, such as search, e-mail, and video applications. To support requests from external clients, each application is associated with a publicly visible IP address to which clients send their requests and from which they receive responses. Inside the data center, the external requests are first directed to a **load balancer** whose job it is to distribute requests to the hosts, balancing the load across the hosts as a function of their current load. A large data center will often have several load balancers, each one devoted to a set of specific cloud applications. Such a load balancer is sometimes referred to as a "layer-4 switch" since it makes decisions based on the destination port number (layer 4) as well as destination IP address in the packet. Upon receiving a request for a particular application, the load balancer forwards it to one of the hosts that handles the application. (A host may then invoke the services of other hosts to help process the request.) When the host finishes processing the request, it sends its response back to the load balancer, which in turn relays the response back to the external client. The load balancer not only balances the work load across hosts, but also provides a NAT-like function, translating the public external IP address to the internal IP address of the appropriate host, and then translating back for packets traveling in the reverse direction back to the clients. This prevents clients from contacting hosts directly, which has the security benefit of hiding the internal network structure and preventing clients from directly interacting with the hosts.

Hierarchical Architecture

For a small data center housing only a few thousand hosts, a simple network consisting of a border router, a load balancer, and a few tens of racks all interconnected by a single Ethernet switch could possibly suffice. But to scale to tens to hundreds of thousands of hosts, a data center often employs a **hierarchy of routers and switches**, such as the topology shown in **Figure 6.30**. At the top of the hierarchy, the border router connects to access routers (only two are shown in **Figure 6.30**, but there can be many more). Below each access router there are three tiers of switches. Each access router connects to a top-tier switch, and each top-tier switch connects to multiple second-tier switches and a load balancer. Each second-tier switch in turn connects to multiple racks via the racks' TOR switches (third-tier switches). All links typically use Ethernet for their link-layer and physical-layer protocols, with a mix of copper and fiber cabling. With such a hierarchical design, it is possible to scale a data center to hundreds of thousands of hosts.

Because it is critical for a cloud application provider to continually provide applications with high availability, data centers also include redundant network equipment and redundant links in their designs (not shown in **Figure 6.30**). For example, each TOR switch can connect to two tier-2 switches, and each access router, tier-1 switch, and tier-2 switch can be duplicated and integrated into the design **[Cisco 2012**; **Greenberg 2009b]**. In the hierarchical design in **Figure 6.30**, observe that the hosts below each access router form a single subnet. In order to localize ARP broadcast traffic, each of these subnets is further partitioned into smaller VLAN subnets, each comprising a few hundred hosts **[Greenberg 2009a]**.

Although the conventional hierarchical architecture just described solves the problem of scale, it suffers from *limited host-to-host capacity* **[Greenberg 2009b]**. To understand this limitation, consider again **Figure 6.30**, and suppose each host connects to its TOR switch with a 1 Gbps link, whereas the links between switches are 10 Gbps Ethernet links. Two hosts in the same rack can always communicate at a full 1 Gbps, limited only by the rate of the hosts' network interface cards. However, if there are many simultaneous flows in the data center network, the maximum rate between two hosts in *different* racks can be much less. To gain insight into this issue, consider a traffic pattern consisting of 40 simultaneous flows between 40 pairs of hosts in different racks. Specifically, suppose each of 10 hosts in rack 1 in **Figure 6.30** sends a flow to a corresponding host in rack 5. Similarly, there are ten simultaneous flows between pairs of hosts in racks 2 and 6, ten simultaneous flows between racks 3 and 7, and ten simultaneous flows between racks 4 and 8. If each flow evenly shares a link's capacity with other flows traversing that link, then the 40 flows crossing the 10 Gbps A-to-B link (as well as the 10 Gbps B-to-C link) will each only receive 10 Gbps/40=250 Mbps, which is significantly less than the 1 Gbps network interface card rate. The problem becomes even more acute for flows between hosts that need to travel higher up the hierarchy. One possible solution to this limitation is to deploy higher-rate switches and routers. But this would significantly increase the cost of the data center, because switches and routers with high port speeds are very expensive.

Supporting high-bandwidth host-to-host communication is important because a key requirement in data centers is flexibility in placement of computation and services [Greenberg 2009b; Farrington 2010]. For example, a large-scale Internet search engine may run on thousands of hosts spread across multiple racks with significant bandwidth requirements between all pairs of hosts. Similarly, a cloud computing service such as EC2 may wish to place the multiple virtual machines comprising a customer's service on the physical hosts with the most capacity irrespective of their location in the data center. If these physical hosts are spread across multiple racks, network bottlenecks as described above may result in poor performance.

Trends in Data Center Networking

In order to reduce the cost of data centers, and at the same time improve their delay and throughput performance, Internet cloud giants such as Google, Facebook, Amazon, and Microsoft are continually deploying new data center network designs. Although these designs are proprietary, many important trends can nevertheless be identified.

One such trend is to deploy new interconnection architectures and network protocols that overcome the drawbacks of the traditional hierarchical designs. One such approach is to replace the hierarchy of switches and routers with a **fully connected topology** [Facebook 2014; Al-Fares 2008; Greenberg 2009b; Guo 2009], such as the topology shown in Figure 6.31. In this design, each tier-1 switch connects to all of the tier-2 switches so that (1) host-to-host traffic never has to rise above the switch tiers, and (2) with n tier-1 switches, between any two tier-2 switches there are n disjoint paths. Such a design can significantly improve the host-to-host capacity. To see this, consider again our example of 40 flows. The topology in Figure 6.31 can handle such a flow pattern since there are four distinct paths between the first tier-2 switch and the second tier-2 switch, together providing an aggregate capacity of 40 Gbps between the first two tier-2 switches. Such a design not only alleviates the host-to-host capacity limitation, but also creates a more flexible computation and service environment in which communication between any two racks not connected to the same switch is logically equivalent, irrespective of their locations in the data center.

Another major trend is to employ shipping container–based modular data centers (MDCs) [YouTube 2009; Waldrop 2007]. In an MDC, a factory builds, within a

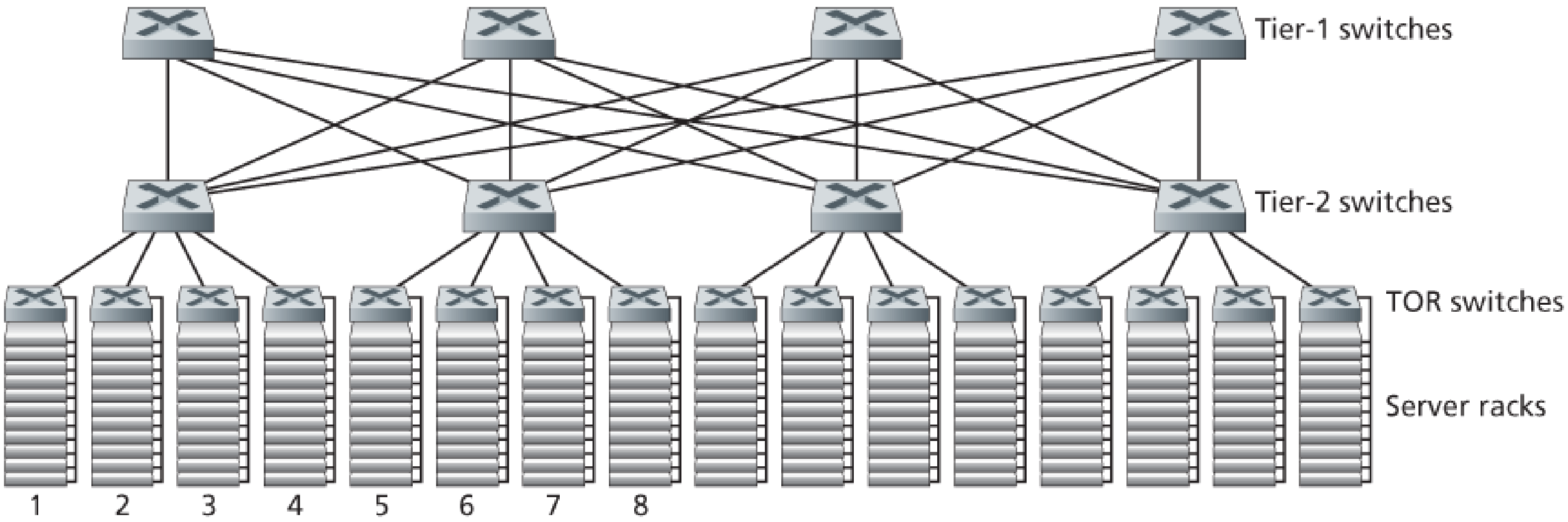


Figure 6.31 Highly interconnected data network topology

standard 12-meter shipping container, a “mini data center” and ships the container to the data center location. Each container has up to a few thousand hosts, stacked in tens of racks, which are packed closely together. At the data center location, multiple containers are interconnected with each other and also with the Internet. Once a prefabricated container is deployed at a data center, it is often difficult to service. Thus, each container is designed for graceful performance degradation: as components (servers and switches) fail over time, the container continues to operate but with degraded performance. When many components have failed and performance has dropped below a threshold, the entire container is removed and replaced with a fresh one.

Building a data center out of containers creates new networking challenges. With an MDC, there are two types of networks: the container-internal networks within each of the containers and the core network connecting each container **[Guo 2009**; **Farrington 2010]**. Within each container, at the scale of up to a few thousand hosts, it is possible to build a fully connected network (as described above) using inexpensive commodity Gigabit Ethernet switches. However, the design of the core network, interconnecting hundreds to thousands of containers while providing high host-to-host bandwidth across containers for typical workloads, remains a challenging problem. A hybrid electrical/optical switch architecture for interconnecting the containers is proposed in **[Farrington 2010]**.

When using highly interconnected topologies, one of the major issues is designing routing algorithms among the switches. One possibility **[Greenberg 2009b]** is to use a form of random routing. Another possibility **[Guo 2009]** is to deploy multiple network interface cards in each host, connect each host to multiple low-cost commodity switches, and allow the hosts themselves to intelligently route traffic among the switches. Variations and extensions of these approaches are currently being deployed in contemporary data centers.

Another important trend is that large cloud providers are increasingly building or customizing just about everything that is in their data centers, including network adapters, switches routers, TORs, software, and networking protocols [Greenberg 2015, Singh 2015]. Another trend, pioneered by Amazon, is to improve reliability with "availability zones," which essentially replicate distinct data centers in different nearby buildings. By having the buildings nearby (a few kilometers apart), transactional data can be synchronized across the data centers in the same availability zone while providing fault tolerance [Amazon 2014]. Many more innovations in data center design are likely to continue to come; interested readers are encouraged to see the recent papers and videos on data center network design.

6.7 Retrospective: A Day in the Life of a Web Page Request

Now that we've covered the link layer in this chapter, and the network, transport and application layers in earlier chapters, our journey down the protocol stack is complete! In the very beginning of this book (**Section 1.1**), we wrote "much of this book is concerned with computer network protocols," and in the first five chapters, we've certainly seen that this is indeed the case! Before heading into the topical chapters in second part of this book, we'd like to wrap up our journey down the protocol stack by taking an integrated, holistic view of the protocols we've learned about so far. One way then to take this "big picture" view is to identify the many (many!) protocols that are involved in satisfying even the simplest request: downloading a Web page. **Figure 6.32** illustrates our setting: a student, Bob, connects a laptop to his school's Ethernet switch and downloads a Web page (say the home page of **www.google.com**). As we now know, there's a *lot* going on "under the hood" to satisfy this seemingly simple request. A Wireshark lab at the end of this chapter examines trace files containing a number of the packets involved in similar scenarios in more detail.

6.7.1 Getting Started: DHCP, UDP, IP, and Ethernet

Let's suppose that Bob boots up his laptop and then connects it to an Ethernet cable connected to the school's Ethernet switch, which in turn is connected to the school's router, as shown in **Figure 6.32**. The school's router is connected to an ISP, in this example, comcast.net. In this example, comcast.net is providing the DNS service for the school; thus, the DNS server resides in the Comcast network rather than the school network. We'll assume that the DHCP server is running within the router, as is often the case.

When Bob first connects his laptop to the network, he can't do anything (e.g., download a Web page) without an IP address. Thus, the first network-related

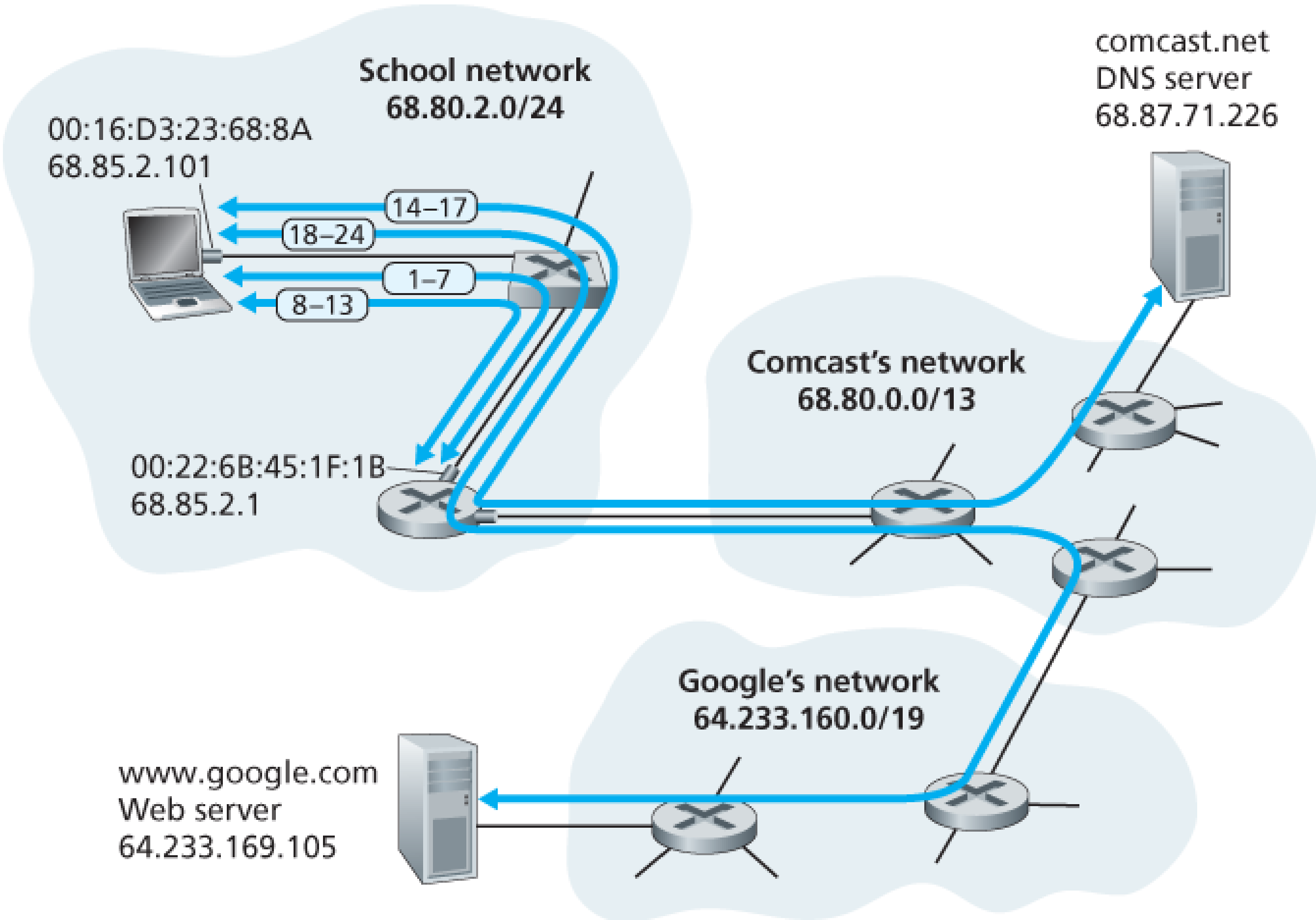


Figure 6.32 A day in the life of a Web page request: Network setting and actions

action taken by Bob's laptop is to run the DHCP protocol to obtain an IP address, as well as other information, from the local DHCP server:

1. The operating system on Bob's laptop creates a **DHCP request message** (**Section 4.3.3**) and puts this message within a **UDP segment** (**Section 3.3**) with destination port 67 (DHCP server) and source port 68 (DHCP client). The UDP segment is then placed within an **IP datagram** (**Section 4.3.1**) with a broadcast IP destination address (255.255.255.255) and a source IP address of 0.0.0.0, since Bob's laptop doesn't yet have an IP address.
2. The IP datagram containing the DHCP request message is then placed within an **Ethernet frame** (**Section 6.4.2**). The Ethernet frame has a destination MAC addresses of FF:FF:FF:FF:FF:FF so that the frame will be broadcast to all devices connected to the switch (hopefully including a DHCP server); the frame's source MAC address is that of Bob's laptop, 00:16:D3:23:68:8A.
3. The broadcast Ethernet frame containing the DHCP request is the first frame sent by Bob's laptop to the Ethernet switch. The switch broadcasts the incoming frame on all outgoing ports, including the port connected to the router.
4. The router receives the broadcast Ethernet frame containing the DHCP request on its interface with MAC address 00:22:6B:45:1F:1B and the IP datagram is extracted from the Ethernet frame. The datagram's broadcast IP destination address indicates that this IP datagram should be processed by upper layer protocols at this node, so the datagram's payload (a UDP segment) is

thus **demultiplexed** (**Section 3.2**) up to UDP, and the DHCP request message is extracted from the UDP segment. The DHCP server now has the DHCP request message.

5. Let's suppose that the DHCP server running within the router can allocate IP addresses in the **CIDR** (**Section 4.3.3**) block 68.85.2.0/24. In this example, all IP addresses used within the school are thus within Comcast's address block. Let's suppose the DHCP server allocates address 68.85.2.101 to Bob's laptop. The DHCP server creates a **DHCP ACK message** (**Section 4.3.3**) containing this IP address, as well as the IP address of the DNS server (68.87.71.226), the IP address for the default gateway router (68.85.2.1), and the subnet block (68.85.2.0/24) (equivalently, the "network mask"). The DHCP message is put inside a UDP segment, which is put inside an IP datagram, which is put inside an Ethernet frame. The Ethernet frame has a source MAC address of the router's interface to the home network (00:22:6B:45:1F:1B) and a destination MAC address of Bob's laptop (00:16:D3:23:68:8A).
6. The Ethernet frame containing the DHCP ACK is sent (unicast) by the router to the switch. Because the switch is **self-learning** (**Section 6.4.3**) and previously received an Ethernet frame (containing the DHCP request) from Bob's laptop, the switch knows to forward a frame addressed to 00:16:D3:23:68:8A only to the output port leading to Bob's laptop.
7. Bob's laptop receives the Ethernet frame containing the DHCP ACK, extracts the IP datagram from the Ethernet frame, extracts the UDP segment from the IP datagram, and extracts the DHCP ACK message from the UDP segment. Bob's DHCP client then records its IP address and the IP address of its DNS server. It also installs the address of the default gateway into its **IP forwarding table** (**Section 4.1**). Bob's laptop will send all datagrams with destination address outside of its subnet 68.85.2.0/24 to the default gateway. At this point, Bob's laptop has initialized its networking components and is ready to begin processing the Web page fetch. (Note that only the last two DHCP steps of the four presented in **Chapter 4** are actually necessary.)

6.7.2 Still Getting Started: DNS and ARP

When Bob types the URL for **www.google.com** into his Web browser, he begins the long chain of events that will eventually result in Google's home page being displayed by his Web browser. Bob's Web browser begins the process by creating a **TCP socket** (**Section 2.7**) that will be used to send the **HTTP request** (**Section 2.2**) to **www.google.com**. In order to create the socket, Bob's laptop will need to know the IP address of **www.google.com**. We learned in **Section 2.5**, that the **DNS protocol** is used to provide this name-to-IP-address translation service.

8. The operating system on Bob's laptop thus creates a **DNS query message** (Section 2.5.3), putting the string "**www.google.com**" in the question section of the DNS message. This DNS message is then placed within a UDP segment with a destination port of 53 (DNS server). The UDP segment is then placed within an IP datagram with an IP destination address of

68.87.71.226 (the address of the DNS server returned in the DHCP ACK in step 5) and a source IP address of 68.85.2.101.

9. Bob's laptop then places the datagram containing the DNS query message in an Ethernet frame. This frame will be sent (addressed, at the link layer) to the gateway router in Bob's school's network. However, even though Bob's laptop knows the IP address of the school's gateway router (68.85.2.1) via the DHCP ACK message in step 5 above, it doesn't know the gateway router's MAC address. In order to obtain the MAC address of the gateway router, Bob's laptop will need to use the **ARP protocol** (**Section 6.4.1**).
10. Bob's laptop creates an **ARP query** message with a target IP address of 68.85.2.1 (the default gateway), places the ARP message within an Ethernet frame with a broadcast destination address (FF:FF:FF:FF:FF:FF) and sends the Ethernet frame to the switch, which delivers the frame to all connected devices, including the gateway router.
11. The gateway router receives the frame containing the ARP query message on the interface to the school network, and finds that the target IP address of 68.85.2.1 in the ARP message matches the IP address of its interface. The gateway router thus prepares an **ARP reply**, indicating that its MAC address of 00:22:6B:45:1F:1B corresponds to IP address 68.85.2.1. It places the ARP reply message in an Ethernet frame, with a destination address of 00:16:D3:23:68:8A (Bob's laptop) and sends the frame to the switch, which delivers the frame to Bob's laptop.
12. Bob's laptop receives the frame containing the ARP reply message and extracts the MAC address of the gateway router (00:22:6B:45:1F:1B) from the ARP reply message.
13. Bob's laptop can now (*finally!*) address the Ethernet frame containing the DNS query to the gateway router's MAC address. Note that the IP datagram in this frame has an IP destination address of 68.87.71.226 (the DNS server), while the frame has a destination address of 00:22:6B:45:1F:1B (the gateway router). Bob's laptop sends this frame to the switch, which delivers the frame to the gateway router.

6.7.3 Still Getting Started: Intra-Domain Routing to the DNS Server

14. The gateway router receives the frame and extracts the IP datagram containing the DNS query. The router looks up the destination address of this datagram (68.87.71.226) and determines from its forwarding table that the datagram should be sent to the leftmost router in the Comcast network in **Figure 6.32**. The IP datagram is placed inside a link-layer frame appropriate for the link connecting the school's router to the leftmost Comcast router and the frame is sent over this link.
15. The leftmost router in the Comcast network receives the frame, extracts the IP datagram, examines the datagram's destination address (68.87.71.226) and determines the outgoing interface on which to forward the datagram toward the DNS server from its forwarding table, which has been filled in by Comcast's intra-domain protocol (such as **RIP**, **OSPF** or **IS-IS**,

Section 5.3) as well as the **Internet's inter-domain protocol**, **BGP** (**Section 5.4**).

16. Eventually the IP datagram containing the DNS query arrives at the DNS server. The DNS server extracts the DNS query message, looks up the name **www.google.com** in its DNS database (**Section 2.5**), and finds the **DNS resource record** that contains the IP address (64.233.169.105) for **www.google.com**. (assuming that it is currently cached in the DNS server). Recall that this cached data originated in the **authoritative DNS server** (Section 2.5.2) for googlecom. The DNS server forms a **DNS reply message** containing this hostname-to-IP-address mapping, and places the DNS reply message in a UDP segment, and the segment within an IP datagram addressed to Bob's laptop (68.85.2.101). This datagram will be forwarded back through the Comcast network to the school's router and from there, via the Ethernet switch to Bob's laptop.
17. Bob's laptop extracts the IP address of the server **www.google.com** from the DNS message. *Finally*, after a *lot* of work, Bob's laptop is now ready to contact the **www.google.com** server!

6.7.4 Web Client-Server Interaction: TCP and HTTP

18. Now that Bob's laptop has the IP address of **www.google.com**, it can create the **TCP socket** (**Section 2.7**) that will be used to send the **HTTP GET** message (**Section 2.2.3**) to **www.google.com**. When Bob creates the TCP socket, the TCP in Bob's laptop must first perform a **three-way handshake** (**Section 3.5.6**) with the TCP in **www.google.com**. Bob's laptop thus first creates a **TCP SYN** segment with destination port 80 (for HTTP), places the TCP segment inside an IP datagram with a destination IP address of 64.233.169.105 (**www.google.com**), places the datagram inside a frame with a destination MAC address of 00:22:6B:45:1F:1B (the gateway router) and sends the frame to the switch.
19. The routers in the school network, Comcast's network, and Google's network forward the datagram containing the TCP SYN toward **www.google.com**, using the forwarding table in each router, as in steps 14–16 above. Recall that the router forwarding table entries governing forwarding of packets over the inter-domain link between the Comcast and Google networks are determined by the **BGP** protocol (**Chapter 5**).
20. Eventually, the datagram containing the TCP SYN arrives at **www.google.com**. The TCP SYN message is extracted from the datagram and demultiplexed to the welcome socket associated with port 80. A connection socket (**Section 2.7**) is created for the TCP connection between the Google HTTP server and Bob's laptop. A TCP SYNACK (**Section 3.5.6**) segment is generated, placed inside a datagram addressed to Bob's laptop, and finally placed inside a link-layer frame appropriate for the link connecting **www.google.com** to its first-hop router.
21. The datagram containing the TCP SYNACK segment is forwarded through the Google, Comcast, and school networks, eventually arriving at the Ethernet card in Bob's laptop. The datagram is demultiplexed within the operating system to the TCP socket created in step 18, which enters the connected state.

22. With the socket on Bob's laptop now (*finally!*) ready to send bytes to **www.google.com**, Bob's browser creates the HTTP GET message (**Section 2.2.3**) containing the URL to be fetched. The HTTP GET message is then written into the socket, with the GET message becoming the payload of a TCP segment. The TCP segment is placed in a datagram and sent and delivered to **www.google.com** as in steps 18–20 above.
23. The HTTP server at **www.google.com** reads the HTTP GET message from the TCP socket, creates an **HTTP response** message (**Section 2.2**), places the requested Web page content in the body of the HTTP response message, and sends the message into the TCP socket.
24. The datagram containing the HTTP reply message is forwarded through the Google, Comcast, and school networks, and arrives at Bob's laptop. Bob's Web browser program reads the HTTP response from the socket, extracts the html for the Web page from the body of the HTTP response, and finally (*finally!*) displays the Web page!

Our scenario above has covered a lot of networking ground! If you've understood most or all of the above example, then you've also covered a lot of ground since you first read **Section 1.1**, where we wrote "much of this book is concerned with computer network protocols" and you may have wondered what a protocol actually was! As detailed as the above example might seem, we've omitted a number of possible additional protocols (e.g., NAT running in the school's gateway router, wireless access to the school's network, security protocols for accessing the school network or encrypting segments or datagrams, network management protocols), and considerations (Web caching, the DNS hierarchy) that one would encounter in the public Internet. We'll cover a number of these topics and more in the second part of this book.

Lastly, we note that our example above was an integrated and holistic, but also very "nuts and bolts," view of many of the protocols that we've studied in the first part of this book. The example focused more on the "how" than the "why." For a broader, more reflective view on the design of network protocols in general, see **[Clark 1988**, **RFC 5218]**.

6.8 Summary

In this chapter, we've examined the link layer—its services, the principles underlying its operation, and a number of important specific protocols that use these principles in implementing link-layer services.

We saw that the basic service of the link layer is to move a network-layer datagram from one node (host, switch, router, WiFi access point) to an adjacent node. We saw that all link-layer protocols operate by encapsulating a network-layer datagram within a link-layer frame before transmitting the frame over the link to the adjacent node. Beyond this common framing function, however, we learned that different link-layer protocols provide very different link access, delivery, and transmission services. These differences are due in part to the wide variety of link types over which link-layer protocols must operate. A simple point-to-point link has a single sender and receiver communicating over a single "wire." A multiple access link is shared among many senders and receivers; consequently, the link-layer protocol for a multiple access channel has a protocol (its multiple access protocol) for coordinating link access. In the case of MPLS, the "link" connecting two adjacent nodes (for example, two IP routers that are adjacent in an IP sense—that they are next-hop IP routers toward some destination) may actually be a *network* in and of itself. In one sense, the idea of a network being considered as a link should not seem odd. A telephone link connecting a home modem/computer to a remote modem/router, for example, is actually a path through a sophisticated and complex telephone *network.*

Among the principles underlying link-layer communication, we examined error-detection and -correction techniques, multiple access protocols, link-layer addressing, virtualization (VLANs), and the construction of extended switched LANs and data center networks. Much of the focus today at the link layer is on these switched networks. In the case of error detection/correction, we examined how it is possible to add additional bits to a frame's header in order to detect, and in some cases correct, bit-flip errors that might occur when the frame is transmitted over the link. We covered simple parity and checksumming schemes, as well as the more robust cyclic redundancy check. We then moved on to the topic of multiple access protocols. We identified and studied three broad approaches for coordinating access to a broadcast channel: channel partitioning approaches (TDM, FDM), random access approaches (the ALOHA protocols and CSMA protocols), and taking-turns approaches (polling and token passing). We studied the cable access network and found that it uses many of these multiple access methods. We saw that a consequence of having multiple nodes share a single broadcast channel was the need to provide node addresses at the link layer. We learned that link-layer addresses were quite different from network-layer addresses and that, in the case of the Internet, a special protocol (ARP—the Address Resolution Protocol) is used to translate between these two forms of addressing and studied the hugely successful Ethernet protocol in detail. We then examined how nodes sharing a broadcast channel form a LAN and how multiple LANs can be connected together to form larger LANs—all *without* the intervention of network-layer routing to interconnect these local nodes. We also learned how multiple virtual LANs can be created on a single physical LAN infrastructure.

We ended our study of the link layer by focusing on how MPLS networks provide link-layer services when they interconnect IP routers and an overview of the network designs for today's massive data centers. We wrapped up this chapter (and indeed the first five chapters) by identifying the many protocols that are needed to fetch a simple Web page. Having covered the link layer, *our journey down the protocol stack is now ove*r! Certainly, the physical layer lies below the link layer, but the details of the physical layer are probably best left for another course (for example, in communication theory, rather than computer networking). We have, however, touched upon several aspects of the physical layer in this chapter and in **Chapter 1** (our discussion of physical media in **Section 1.2**). We'll consider the physical layer again when we study wireless link characteristics in the next chapter.

Although our journey down the protocol stack is over, our study of computer networking is not yet at an end. In the following three chapters we cover wireless networking, network security, and multimedia networking. These four topics do not fit conveniently into any one layer; indeed, each topic crosscuts many layers. Understanding these topics (billed as advanced topics in some networking texts) thus requires a firm foundation in all layers of the protocol stack—a foundation that our study of the link layer has now completed!

Homework Problems and Questions

Chapter 6 Review Questions

SECTIONS 6.1–6.2

R1. Consider the transportation analogy in **Section 6.1.1** . If the passenger is analagous to a datagram, what is analogous to the link layer frame?

R2. If all the links in the Internet were to provide reliable delivery service, would the TCP reliable delivery service be redundant? Why or why not?

R3. What are some of the possible services that a link-layer protocol can offer to the network layer? Which of these link-layer services have corresponding services in IP? In TCP?

SECTION 6.3

R4. Suppose two nodes start to transmit at the same time a packet of length *L* over a broadcast channel of rate *R*. Denote the propagation delay between the two nodes as d_{prop}. Will there be a collision if dprop<L/R? Why or why not?

R5. In **Section 6.3** , we listed four desirable characteristics of a broadcast channel. Which of these characteristics does slotted ALOHA have? Which of these characteristics does token passing have?

R6. In CSMA/CD, after the fifth collision, what is the probability that a node chooses $K=4$? The result K=4 corresponds to a delay of how many seconds on a 10 Mbps Ethernet?

R7. Describe polling and token-passing protocols using the analogy of cocktail party interactions.

R8. Why would the token-ring protocol be inefficient if a LAN had a very large perimeter?

SECTION 6.4

R9. How big is the MAC address space? The IPv4 address space? The IPv6 address space?

R10. Suppose nodes A, B, and C each attach to the same broadcast LAN (through their adapters). If A sends thousands of IP datagrams to B with each encapsulating frame addressed to the MAC address of B, will C's adapter process these frames? If so, will C's adapter pass the IP datagrams in these frames to the network layer C? How would your answers change if A sends frames with the MAC broadcast address?

R11. Why is an ARP query sent within a broadcast frame? Why is an ARP response sent within a frame with a specific destination MAC address?

R12. For the network in **Figure 6.19** , the router has two ARP modules, each with its own ARP table. Is it possible that the same MAC address appears in both tables?

R13. Compare the frame structures for 10BASE-T, 100BASE-T, and Gigabit Ethernet. How do they differ?

R14. Consider **Figure 6.15** . How many subnetworks are there, in the addressing sense of **Section 4.3** ?

R15. What is the maximum number of VLANs that can be configured on a switch supporting the 802.1Q protocol? Why?

R16. Suppose that *N* switches supporting *K* VLAN groups are to be connected via a trunking protocol. How many ports are needed to connect the switches? Justify your answer.

Problems

P1. Suppose the information content of a packet is the bit pattern 1110 0110 1001 1101 and an even parity scheme is being used. What would the value of the field containing the parity bits be for the case of a two-dimensional parity scheme? Your answer should be such that a minimum-length checksum field is used.

P2. Show (give an example other than the one in **Figure 6.5**) that two-dimensional parity checks can correct and detect a single bit error. Show (give an example of) a double-bit error that can be detected but not corrected.

P3. Suppose the information portion of a packet (*D* in **Figure 6.3**) contains 10 bytes consisting of the 8-bit unsigned binary ASCII representation of string “Networking.” Compute the Internet checksum for this data.

P4. Consider the previous problem, but instead suppose these 10 bytes contain

a. the binary representation of the numbers 1 through 10.
b. the ASCII representation of the letters B through K (uppercase).
c. the ASCII representation of the letters b through k (lowercase).
   Compute the Internet checksum for this data.

P5. Consider the 5-bit generator, $G=10011$, and suppose that D has the value 1010101010. What is the value of R?

P6. Consider the previous problem, but suppose that *D* has the value

a. 1001010101.
b. 0101101010.
c. 1010100000.

P7. In this problem, we explore some of the properties of the CRC. For the generator $G(=1001)$ given in **Section 6.2.3** , answer the following questions.

a. Why can it detect any single bit error in data D?
b. Can the above G detect any odd number of bit errors? Why?

P8. In **Section 6.3** , we provided an outline of the derivation of the efficiency of slotted ALOHA. In this problem we'll complete the derivation.

a. Recall that when there are *N* active nodes, the efficiency of slotted ALOHA is $Np(1-p)^{N-1}$. Find the value of *p* that maximizes this expression.
b. Using the value of *p* found in (a), find the efficiency of slotted ALOHA by letting *N* approach infinity. *Hint*: $(1-1/N)^N$ approaches 1/*e* as *N* approaches infinity.

P9. Show that the maximum efficiency of pure ALOHA is 1/(2*e*). *Note*: This problem is easy if you have completed the problem above!

P 10. Consider two nodes, A and B, that use the slotted ALOHA protocol to contend for a channel. Suppose node A has more data to transmit than node B, and node A's retransmission probability p_A is greater than node B's retransmission probability, p_B.

a. Provide a formula for node A's average throughput. What is the total efficiency of the protocol with these two nodes?
b. If $p_A=2p_B$, is node A's average throughput twice as large as that of node B? Why or why not? If not, how can you choose p_A and p_B to make that happen?
c. In general, suppose there are *N* nodes, among which node A has retransmission probability *2p* and all other nodes have retransmission probability *p.* Provide expressions to compute the average throughputs of node A and of any other node.

P11. Suppose four active nodes—nodes A, B, C and D—are competing for access to a channel using slotted ALOHA. Assume each node has an infinite number of packets to send. Each node attempts to transmit in each slot with probability *p*. The first slot is numbered slot 1, the second slot is numbered slot 2, and so on.

a. What is the probability that node A succeeds for the first time in slot 5?
b. What is the probability that some node (either A, B, C or D) succeeds in slot 4?
c. What is the probability that the first success occurs in slot 3?
d. What is the efficiency of this four-node system?

P12. Graph the efficiency of slotted ALOHA and pure ALOHA as a function of *p* for the following values of *N*:

a. $N=15$.
b. $N=25$.
c. $N=35$.

P13. Consider a broadcast channel with *N* nodes and a transmission rate of *R* bps. Suppose the broadcast channel uses polling (with an additional polling node) for multiple access. Suppose the amount of time from when a node completes transmission until the subsequent node is permitted to transmit (that is, the polling delay) is d_{poll}. Suppose that within a polling round, a given node is allowed to transmit at most Q bits. What is the maximum throughput of the broadcast channel?

P14. Consider three LANs interconnected by two routers, as shown in **Figure 6.33** .

a. Assign IP addresses to all of the interfaces. For Subnet 1 use addresses of the form 192.168.1.xxx; for Subnet 2 uses addresses of the form 192.168.2.xxx; and for Subnet 3 use addresses of the form 192.168.3.xxx.
b. Assign MAC addresses to all of the adapters.
c. Consider sending an IP datagram from Host E to Host B. Suppose all of the ARP tables are up to date. Enumerate all the steps, as done for the single-router example in **Section 6.4.1** .
d. Repeat (c), now assuming that the ARP table in the sending host is empty (and the other tables are up to date).

P15. Consider **Figure 6.33** . Now we replace the router between subnets 1 and 2 with a switch S1, and label the router between subnets 2 and 3 as R1.

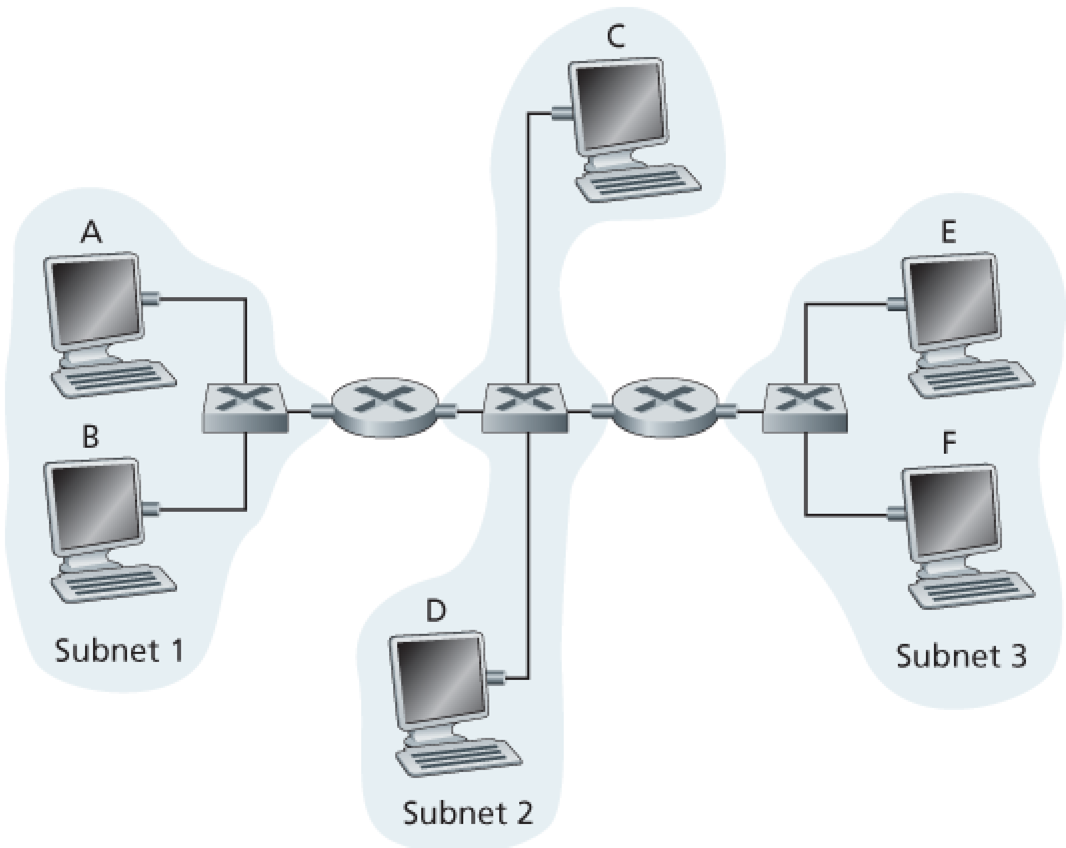


Figure 6.33 Three subnets, interconnected by routers

a. Consider sending an IP datagram from Host E to Host F. Will Host E ask router R1 to help forward the datagram? Why? In the Ethernet frame containing the IP datagram, what are the source and destination IP and MAC addresses?
b. Suppose E would like to send an IP datagram to B, and assume that E's ARP cache does not contain B's MAC address. Will E perform an ARP query to find B's MAC

address? Why? In the Ethernet frame (containing the IP datagram destined to B) that is delivered to router R1, what are the source and destination IP and MAC addresses?

c. Suppose Host A would like to send an IP datagram to Host B, and neither A's ARP cache contains B's MAC address nor does B's ARP cache contain A's MAC address. Further suppose that the switch S1's forwarding table contains entries for Host B and router R1 only. Thus, A will broadcast an ARP request message. What actions will switch S1 perform once it receives the ARP request message? Will router R1 also receive this ARP request message? If so, will R1 forward the message to Subnet 3? Once Host B receives this ARP request message, it will send back to Host A an ARP response message. But will it send an ARP query message to ask for A's MAC address? Why? What will switch S1 do once it receives an ARP response message from Host B?

P16. Consider the previous problem, but suppose now that the router between subnets 2 and 3 is replaced by a switch. Answer questions (a)–(c) in the previous problem in this new context.

P17. Recall that with the CSMA/CD protocol, the adapter waits $K \cdot 512$ bit times after a collision, where *K* is drawn randomly. For $K=100$, how long does the adapter wait until returning to Step 2 for a 10 Mbps broadcast channel? For a 100 Mbps broadcast channel?

P18. Suppose nodes A and B are on the same 10 Mbps broadcast channel, and the propagation delay between the two nodes is 325 bit times. Suppose CSMA/CD and Ethernet packets are used for this broadcast channel. Suppose node A begins transmitting a frame and, before it finishes, node B begins transmitting a frame. Can A finish transmitting before it detects that B has transmitted? Why or why not? If the answer is yes, then A incorrectly believes that its frame was successfully transmitted without a collision. *Hint*: Suppose at time $t=0$ bits, A begins transmitting a frame. In the worst case, A transmits a minimum-sized frame of 512+64 bit times. So A would finish transmitting the frame at t=512+64 bit times. Thus, the answer is no, if B's signal reaches A before bit time t=512+64 bits. In the worst case, when does B's signal reach A?

P19. Suppose nodes A and B are on the same 10 Mbps broadcast channel, and the propagation delay between the two nodes is 245 bit times. Suppose A and B send Ethernet frames at the same time, the frames collide, and then A and B choose different values of *K* in the CSMA/CD algorithm. Assuming no other nodes are active, can the retransmissions from A and B collide? For our purposes, it suffices to work out the following example. Suppose A and B begin transmission at t=0 bit times. They both detect collisions at t=245 t bit times. Suppose KA=0 and KB=1. At what time does B schedule its retransmission? At what time does A begin transmission? (*Note*: The nodes must wait for an idle channel after returning to Step 2—see protocol.) At what time does A's signal reach B? Does B refrain from transmitting at its scheduled time?

P20. In this problem, you will derive the efficiency of a CSMA/CD-like multiple access protocol. In this protocol, time is slotted and all adapters are synchronized to the slots. Unlike slotted ALOHA, however, the length of a slot (in seconds) is much less than a frame time (the time to transmit a frame). Let *S* be the length of a slot. Suppose all frames are of constant length

$L=kRS$, where *R* is the transmission rate of the channel and *k* is a large integer. Suppose there are *N* nodes, each with an infinite number of frames to send. We also assume that $d_{prop}<S$, so that all nodes can detect a collision before the end of a slot time. The protocol is as follows:

- If, for a given slot, no node has possession of the channel, all nodes contend for the channel; in particular, each node transmits in the slot with probability *p*. If exactly one node transmits in the slot, that node takes possession of the channel for the subsequent $k-1$ slots and transmits its entire frame.
- If some node has possession of the channel, all other nodes refrain from transmitting until the node that possesses the channel has finished transmitting its frame. Once this node has transmitted its frame, all nodes contend for the channel.

Note that the channel alternates between two states: the productive state, which lasts exactly *k* slots, and the nonproductive state, which lasts for a random number of slots. Clearly, the channel efficiency is the ratio of $k/(k+x)$, where *x* is the expected number of consecutive unproductive slots.

a. For fixed *N* and *p*, determine the efficiency of this protocol.
b. For fixed *N*, determine the *p* that maximizes the efficiency.
c. Using the *p* (which is a function of *N*) found in (b), determine the efficiency as *N* approaches infinity.
d. Show that this efficiency approaches 1 as the frame length becomes large.

P21. Consider **Figure 6.33** in problem P14. Provide MAC addresses and IP addresses for the interfaces at Host A, both routers, and Host F. Suppose Host A sends a datagram to Host F. Give the source and destination MAC addresses in the frame encapsulating this IP datagram as the frame is transmitted *(i)* from A to the left router, *(ii)* from the left router to the right router, *(iii)* from the right router to F. Also give the source and destination IP addresses in the IP datagram encapsulated within the frame at each of these points in time.

P22. Suppose now that the leftmost router in **Figure 6.33** is replaced by a switch. Hosts A, B, C, and D and the right router are all star-connected into this switch. Give the source and destination MAC addresses in the frame encapsulating this IP datagram as the frame is transmitted *(i)* from A to the switch, *(ii)* from the switch to the right router, *(iii)* from the right router to F. Also give the source and destination IP addresses in the IP datagram encapsulated within the frame at each of these points in time.

P23. Consider **Figure 6.15** . Suppose that all links are 100 Mbps. What is the maximum total aggregate throughput that can be achieved among the 9 hosts and 2 servers in this network? You can assume that any host or server can send to any other host or server. Why?

P24. Suppose the three departmental switches in **Figure 6.15** are replaced by hubs. All links are 100 Mbps. Now answer the questions posed in problem P23.

P25. Suppose that *all* the switches in **Figure 6.15** are replaced by hubs. All links are 100 Mbps. Now answer the questions posed in problem P23.

P26. Let's consider the operation of a learning switch in the context of a network in which 6 nodes labeled A through F are star connected into an Ethernet switch. Suppose that *(i)* B sends a frame to E, *(ii)* E replies with a frame to B, *(iii)* A sends a frame to B, *(iv)* B replies with a frame to A. The switch table is initially empty. Show the state of the switch table before and after each of these events. For each of these events, identify the link(s) on which the transmitted frame will be forwarded, and briefly justify your answers.

P27. In this problem, we explore the use of small packets for Voice-over-IP applications. One of the drawbacks of a small packet size is that a large fraction of link bandwidth is consumed by overhead bytes. To this end, suppose that the packet consists of *P* bytes and 5 bytes of header.

a. Consider sending a digitally encoded voice source directly. Suppose the source is encoded at a constant rate of 128 kbps. Assume each packet is entirely filled before the source sends the packet into the network. The time required to fill a packet is the **packetization delay**. In terms of *L*, determine the packetization delay in milliseconds.
b. Packetization delays greater than 20 msec can cause a noticeable and unpleasant echo. Determine the packetization delay for L=1,500 bytes (roughly corresponding to a maximum-sized Ethernet packet) and for L=50 (corresponding to an ATM packet).
c. Calculate the store-and-forward delay at a single switch for a link rate of R=622 Mbps for L=1,500 bytes, and for L=50 bytes.
d. Comment on the advantages of using a small packet size.

P28. Consider the single switch VLAN in **Figure 6.25** , and assume an external router is connected to switch port 1. Assign IP addresses to the EE and CS hosts and router interface. Trace the steps taken at both the network layer and the link layer to transfer an IP datagram from an EE host to a CS host (*Hint:* Reread the discussion of **Figure 6.19** in the text).

P29. Consider the MPLS network shown in **Figure 6.29** , and suppose that routers R5 and R6 are now MPLS enabled. Suppose that we want to perform traffic engineering so that packets from R6 destined for A are switched to A via R6-R4-R3-R1, and packets from R5 destined for A are switched via R5-R4-R2-R1. Show the MPLS tables in R5 and R6, as well as the modified table in R4, that would make this possible.

P30. Consider again the same scenario as in the previous problem, but suppose that packets from R6 destined for D are switched via R6-R4-R3, while packets from R5 destined to D are switched via R4-R2-R1-R3. Show the MPLS tables in all routers that would make this possible.

P31. In this problem, you will put together much of what you have learned about Internet protocols. Suppose you walk into a room, connect to Ethernet, and want to download a Web page. What are all the protocol steps that take place, starting from powering on your PC to getting the Web page? Assume there is nothing in our DNS or browser caches when you power on your PC. (*Hint:* The steps include the use of Ethernet, DHCP, ARP, DNS, TCP, and HTTP protocols.) Explicitly indicate in your steps how you obtain the IP and MAC addresses of a gateway router.

P32. Consider the data center network with hierarchical topology in **Figure 6.30** . Suppose now there are 80 pairs of flows, with ten flows between the first and ninth rack, ten flows between the second and tenth rack, and so on. Further suppose that all links in the network are 10 Gbps, except for the links between hosts and TOR switches, which are 1 Gbps.

a. Each flow has the same data rate; determine the maximum rate of a flow.
b. For the same traffic pattern, determine the maximum rate of a flow for the highly interconnected topology in **Figure 6.31** .
c. Now suppose there is a similar traffic pattern, but involving 20 hosts on each rack and 160 pairs of flows. Determine the maximum flow rates for the two topologies.

P33. Consider the hierarchical network in **Figure 6.30** and suppose that the data center needs to support e-mail and video distribution among other applications. Suppose four racks of servers are reserved for e-mail and four racks are reserved for video. For each of the applications, all four racks must lie below a single tier-2 switch since the tier-2 to tier-1 links do not have sufficient bandwidth to support the intra-application traffic. For the e-mail application, suppose that for 99.9 percent of the time only three racks are used, and that the video application has identical usage patterns.

a. For what fraction of time does the e-mail application need to use a fourth rack? How about for the video application?
b. Assuming e-mail usage and video usage are independent, for what fraction of time do (equivalently, what is the probability that) both applications need their fourth rack?
c. Suppose that it is acceptable for an application to have a shortage of servers for 0.001 percent of time or less (causing rare periods of performance degradation for users).

   Discuss how the topology in **Figure 6.31** can be used so that only seven racks are collectively assigned to the two applications (assuming that the topology can support all the traffic).

Wireshark Labs

At the Companion website for this textbook, **http://www.pearsonhighered.com/cs-resources/**, you'll find a Wireshark lab that examines the operation of the IEEE 802.3 protocol and the Wireshark frame format. A second Wireshark lab examines packet traces taken in a home network scenario.

> AN INTERVIEW WITH…
> **Simon S. Lam**
> Simon S. Lam is Professor and Regents Chair in Computer Sciences at the University of Texas at Austin. From 1971 to 1974, he was with the ARPA Network Measurement Center at UCLA, where he worked on satellite and radio packet switching. He led a research group that invented secure sockets and prototyped, in 1993, the first secure sockets layer named Secure Network Programming, which won the 2004 ACM Software System Award. His research interests are in design and analysis of network protocols and security services. He received his BSEE from

Washington State University and his MS and PhD from UCLA. He was elected to the National Academy of Engineering in 2007.

Why did you decide to specialize in networking?

When I arrived at UCLA as a new graduate student in Fall 1969, my intention was to study control theory. Then I took the queuing theory classes of Leonard Kleinrock and was very impressed by him. For a while, I was working on adaptive control of queuing systems as a possible thesis topic. In early 1972, Larry Roberts initiated the ARPAnet Satellite System project (later called Packet Satellite). Professor Kleinrock asked me to join the project. The first thing we did was to introduce a simple, yet realistic, backoff algorithm to the slotted ALOHA protocol. Shortly thereafter, I found many interesting research problems, such as ALOHA's instability problem and need for adaptive backoff, which would form the core of my thesis.

You were active in the early days of the Internet in the 1970s, beginning with your student days

at UCLA. What was it like then? Did people have any inkling of what the Internet would become?

The atmosphere was really no different from other system-building projects I have seen in industry and academia. The initially stated goal of the ARPAnet was fairly modest, that is, to provide access to expensive computers from remote locations so that many more scientists could use them. However, with the startup of the Packet Satellite project in 1972 and the Packet Radio project in 1973, ARPA's goal had expanded substantially. By 1973, ARPA was building three different packet networks at the same time, and it became necessary for Vint Cerf and Bob Kahn to develop an interconnection strategy.

Back then, all of these progressive developments in networking were viewed (I believe) as logical rather than magical. No one could have envisioned the scale of the Internet and power of personal computers today. It was a decade before appearance of the first PCs. To put things in perspective, most students submitted their computer programs as decks of punched cards for batch processing. Only some students had direct access to computers, which were typically housed in a restricted area. Modems were slow and still a rarity. As a graduate student, I had only a phone on my desk, and I used pencil and paper to do most of my work.

Where do you see the field of networking and the Internet heading in the future?

In the past, the simplicity of the Internet's IP protocol was its greatest strength in vanquishing competition and becoming the *de facto* standard for internetworking. Unlike competitors, such as X.25 in the 1980s and ATM in the 1990s, IP can run on top of any link-layer networking technology, because it offers only a best-effort datagram service. Thus, any packet network can connect to the Internet.

Today, IP's greatest strength is actually a shortcoming. IP is like a straitjacket that confines the Internet's development to specific directions. In recent years, many researchers have redirected their efforts to the application layer only. There is also a great deal of research on wireless ad hoc networks, sensor networks, and satellite networks. These networks can be viewed either as stand-alone systems or link-layer systems, which can flourish because they are outside of the IP straitjacket.

Many people are excited about the possibility of P2P systems as a platform for novel Internet applications. However, P2P systems are highly inefficient in their use of Internet resources. A concern of mine is whether the transmission and switching capacity of the Internet core will continue to increase faster than the traffic demand on the Internet as it grows to interconnect all kinds of devices and support future P2P-enabled applications. Without substantial overprovisioning of capacity, ensuring network stability in the presence of malicious attacks and congestion will continue to be a significant challenge.

The Internet's phenomenal growth also requires the allocation of new IP addresses at a rapid rate to network operators and enterprises worldwide. At the current rate, the pool of unallocated IPv4 addresses would be depleted in a few years. When that happens, large contiguous blocks of address space can only be allocated from the IPv6 address space. Since adoption of IPv6 is off to a slow start, due to lack of incentives for early adopters, IPv4 and IPv6 will most likely co-exist on the Internet for many years to come. Successful migration from an IPv4-dominant Internet to an IPv6-dominant Internet will require a substantial global effort.

What is the most challenging part of your job?

The most challenging part of my job as a professor is teaching and motivating *every* student in my class, and *every* doctoral student under my supervision, rather than just the high achievers. The very bright and motivated may require a little guidance but not much else. I often learn more from these students than they learn from me. Educating and motivating the underachievers present a major challenge.

What impacts do you foresee technology having on learning in the future?

Eventually, almost all human knowledge will be accessible through the Internet, which will be the most powerful tool for learning. This vast knowledge base will have the potential of leveling the

> playing field for students all over the world. For example, motivated students in any country will be able to access the best-class Web sites, multimedia lectures, and teaching materials. Already, it was said that the IEEE and ACM digital libraries have accelerated the development of computer science researchers in China. In time, the Internet will transcend all geographic barriers to learning.