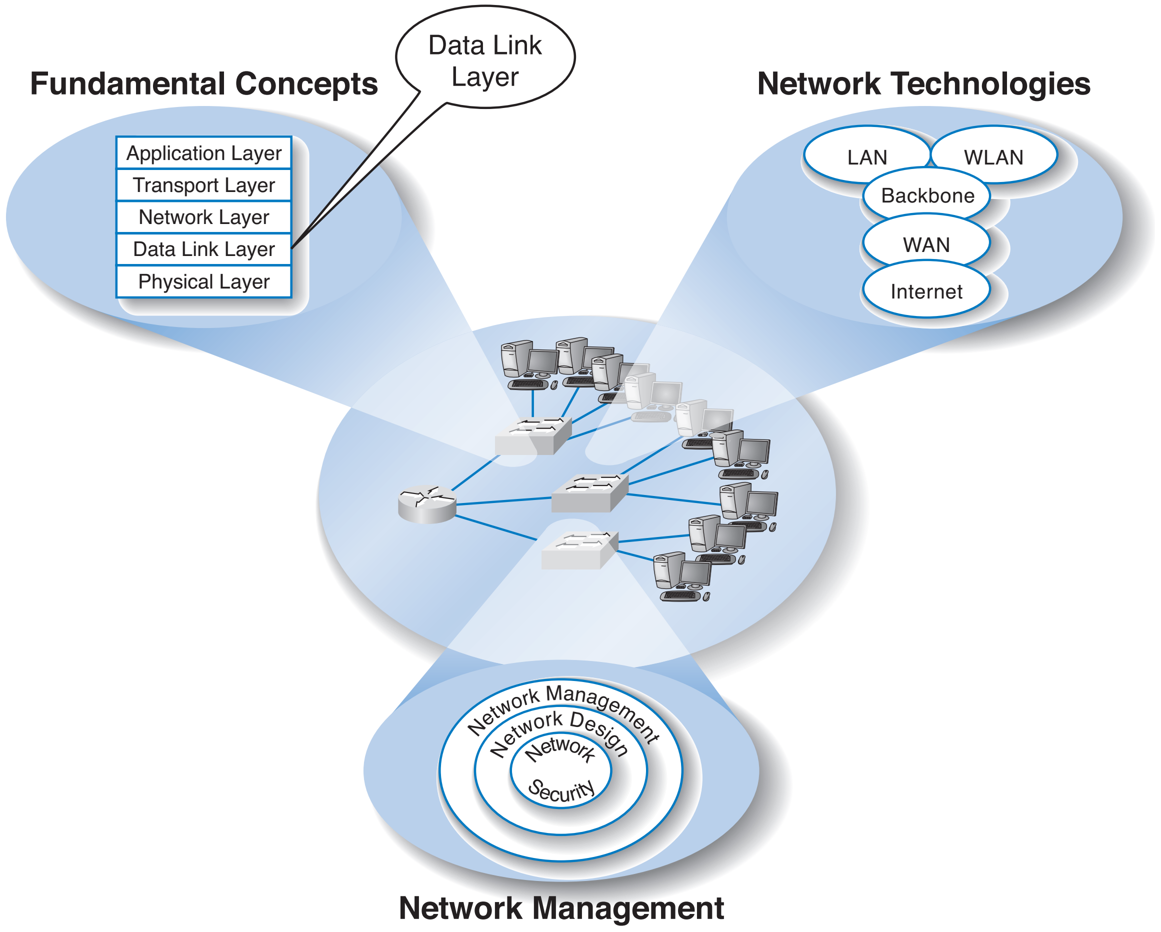
CHAPTER 4

DATA LINK LAYER



The Three Faces of Networking

THE DATA link layer (also called layer 2) is responsible for moving a message from one computer or network device to the next computer or network device in the overall path from sender or receiver. It controls the way messages are sent on the physical media. Both the sender and receiver have to agree on the rules or *protocols* that govern how they will communicate with each other. A *data link protocol* determines who can transmit at what time, where a message begins and ends, and how a receiver recognizes and corrects a transmission error. In this chapter, we discuss these processes, as well as several important sources of errors.

OBJECTIVES

- Understand the role of the data link layer
- Become familiar with two basic approaches to controlling access to the media
- Become familiar with common sources of error and their prevention
- Understand three common error detection and correction methods
- Become familiar with several commonly used data link protocols

CHAPTER OUTLINE

INTRODUCTION

In Chapter 1, we introduced the concept of layers in data communications. The data link layer sits between the physical layer (hardware such as the circuits, computers, and multiplexers described in Chapter 3) and the network layer (that performs addressing and routing, as described in Chapter 5).

The data link layer accepts messages from the network layer and controls the hardware that actually transmits them. The data link layer is responsible for getting a message from one computer to another without errors. The data link layer also accepts streams of bits from the physical layer and organizes them into coherent messages that it passes to the network layer.

Both the sender and receiver have to agree on the rules or *protocols* that govern how their data link layers will communicate with each other. A *data link protocol* performs three functions:

- Controls when computers transmit (*media access control*)
- Detects and corrects transmission errors (*error control*)
- Identifies the start and end of a message (*message delineation*)

MEDIA ACCESS CONTROL

Media access control refers to the need to control when computers transmit. With point-to-point full-duplex configurations, media access control is unnecessary because there are only two computers on the circuit and full duplex permits either computer to transmit at any time.

Media access control becomes important when several computers share the same communication circuit, such as a point-to-point configuration with a half-duplex configuration that requires computers to take turns, or a multipoint configuration in which several computers share the same circuit. Here, it is critical to ensure that no two computers attempt to transmit data at the same time—but if they do, there must be a way to recover from the problem. There are two fundamental approaches to media access control: controlled access and contention.

Controlled Access

Most computer networks managed by a host mainframe computer use controlled access. In this case, the mainframe controls the circuit and determines which clients can access media at what time.

Polling is the process of sending a signal to a client (a computer or terminal) that gives it permission to transmit or asks it to receive. With polling, the clients store all messages that need to be transmitted. Periodically, the server (usually a mainframe computer) *polls* the client to see if it has data to send. If the client has data to send, it does so. If the client has no data to send, it responds negatively, and the server asks another client if it has data to send.

In other words, polling is analogous to a classroom situation in which the instructor calls on the students who raise their hands. The instructor acts like the server. To gain access to the media, students raise their hands and the instructor recognizes them so they can contribute. When they have finished, the instructor again takes charge and allows someone else to comment.

There are several types of polling. With *roll-call polling,* the server works consecutively through a list of clients, first polling client 1, then client 2, and so on, until all are polled. Roll-call polling can be modified to select clients in priority so that some get polled more often than others. For example, one could increase the priority of client 1 by using a polling sequence such as 1, 2, 3, 1, 4, 5, 1, 6, 7, 1, 8, 9.

Typically, roll-call polling involves some waiting because the server has to poll a client and then wait for a response. The response might be an incoming message that was waiting to be sent, a negative response indicating nothing is to be sent, or the full "time-out period" may expire because the client is temporarily out of service (e.g., it is malfunctioning or the user has turned it off). Usually, a timer "times out" the client after waiting several seconds without getting a response. If some sort of fail-safe time-out is not used, the system poll might lock up indefinitely on an out-of-service client.

With *hub polling* (often called *token passing*), one computer starts the poll and passes it to the next computer on the multipoint circuit, which sends its message and passes the poll to the next. That computer then passes the poll to the next, and so on, until it reaches the first computer, which restarts the process again.

Contention

Contention is the opposite of *controlled access.* Computers wait until the circuit is free (i.e., no other computers are transmitting) and then transmit whenever they have data to send. Contention is commonly used in Ethernet LANs.

As an analogy, suppose that you are talking with some friends. Each person tries to get the floor when the previous speaker finishes. Usually, the others yield to the first person who jumps in at the precise moment the previous speaker stops. Sometimes two people attempt to talk at the same time, so there must be some technique to continue the conversation after such a verbal collision occurs.

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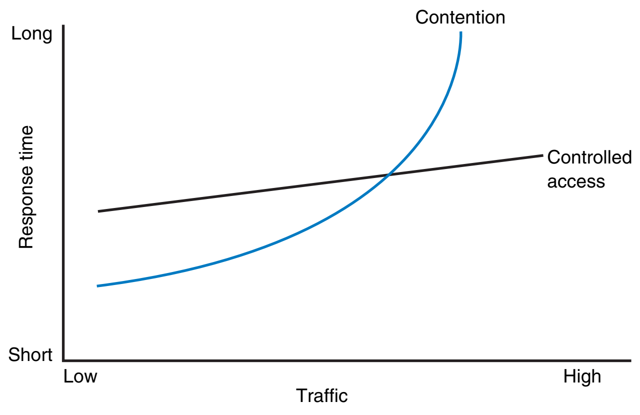


FIGURE 4.1 Relative response times.

Relative Performance

Which media access control approach is best: controlled access or contention? There is no simple answer. The key consideration is throughput—which approach will permit the most amount of user data to be transmitted through the network.

In general, contention approaches work better than controlled approaches for small networks that have low usage. In this case, each computer can transmit when necessary, without waiting for permission. Because usage is low, there is little chance of a collision. In contrast, computers in a controlled access environment must wait for permission, so even if no other computer needs to transmit, they must wait for the poll.

The reverse is true for large networks with high usage: controlled access works better. In high-volume networks, many computers want to transmit, and the probability of a collision using contention is high. Collisions are very costly in terms of throughput because they waste circuit capacity during the collision and require both computers to retransmit later. Controlled access prevents collisions and makes more efficient use of the circuit, and although response time does increase, it does so more gradually (Figure 4.1).

The key to selecting the best access control technique is to find the crossover point between controlled and contention. Although there is no one correct answer, because it depends on how many messages the computers in the network transmit, most experts believe that the crossover point is often around 20 computers (lower for busy computers, higher for less-busy computers). For this reason, when we build shared multipoint circuits like those often used in LANs, we try to put no more than 20 computers on any one shared circuit.

ERROR CONTROL

Before learning the control mechanisms that can be implemented to protect a network from errors, you should realize that there are *human errors* and *network errors.* Human errors, such as a mistake in typing a number, usually are controlled through the application program. Network errors, such as those that occur during transmission, are controlled by the network hardware and software.

There are two categories of network errors: *corrupted data* (data that have been changed) and *lost data.* Networks should be designed to (1) prevent, (2) detect, and (3) correct both corrupted data and lost data. We begin by examining the sources of errors and how to prevent them and then turn to error detection and correction.

Network errors are a fact of life in data communications networks. Depending on the type of circuit, they may occur every few hours, minutes, or seconds because of noise on the lines. No network can eliminate all errors, but most errors can be prevented, detected, and corrected by proper design. IXCs that provide data transmission circuits provide statistical measures specifying typical error rates and the pattern of errors that can be expected on the circuits they lease. For example, the error rate might be stated as 1 in 500,000, meaning there is 1 bit in error for every 500,000 bits transmitted.

Normally, errors appear in bursts. In a *burst error,* more than 1 data bit is changed by the error-causing condition. In other words, errors are not uniformly distributed in time. Although an error rate might be stated as 1 in 500,000, errors are more likely to occur as 100 bits every 50,000,000 bits. The fact that errors tend to be clustered in bursts rather than evenly dispersed is both good and bad. If the errors were not clustered, an error rate of 1 bit in 500,000 would make it rare for 2 erroneous bits to occur in the same character. Consequently, simple character-checking schemes would be effective at detecting errors. When errors are #ore or less evenly distrib#ted, it is not di#ficult to gras# the me#ning even when the error #ate is high, as it is in this #entence (1 charac#er in 20). But burst errors are the rule rather than the exception, often obliterating 100 or more bits at a time. This makes it more difficult to recover the meaning, so more reliance must be placed on special #######[1] or numeric error detection and correction methods. The positive side is that there are long periods of error-free transmission, meaning that very few messages encounter errors.

Sources of Errors

Line noise and *distortion* can cause data communication errors. The focus in this section is on electrical media such as twisted-pair wire and coaxial cable, because they are more likely to suffer from noise than are optical media such as fiber-optic cable. In this case, noise is undesirable electrical signals (for fiber-optic cable, it is undesirable light). Noise is introduced by equipment or natural disturbances, and it degrades the performance of a communication circuit. Noise manifests itself as extra bits, missing bits, or bits that have been “flipped” (i.e., changed from 1 to 0 or vice versa). Figure 4.2 summarizes the major sources of error and ways to prevent them. The first six sources listed there are the most important; the last three are more common in analog rather than digital circuits.

Line outages are a catastrophic cause of errors and incomplete transmission. Occasionally, a communication circuit fails for a brief period. This type of failure may be caused by faulty telephone end office equipment, storms, loss of the carrier signal, and

[1]In case you could not guess, the word is *logical.*

Source of Error	What Causes It	How to Prevent It
Line outages	Storms, accidents	
White noise	Movement of electrons	Increase signal strength
Impulse noise	Sudden increases in electricity (e.g., lightning)	Shield or move the wires
Cross-talk	Multiplexer guardbands too small or wires too close together	Increase the guardbands or move or shield the wires
Echo	Poor connections	Fix the connections or tune equipment
Attenuation	Gradual decrease in signal over distance	Use repeaters or amplifiers
Intermodulation noise	Signals from several circuits combine	Move or shield the wires
Jitter	Analog signals change phase	Tune equipment
Harmonic distortion	Amplifier changes phase	Tune equipment

FIGURE 4.2 Sources of errors and ways to minimize them.

any other failure that causes a short circuit. The most common cause of line outages are storms that cause damage to circuits or facilities.

White noise or *Gaussian noise* (the familiar background hiss or static on radios and telephones) is caused by the thermal agitation of electrons and therefore is inescapable. Even if the equipment were perfect and the wires were perfectly insulated from any and all external interference, there still would be some white noise. White noise usually is not a problem unless it becomes so strong that it obliterates the transmission. In this case, the strength of the electrical signal is increased so it overpowers the white noise; in technical terms, we increase the signal-to-noise ratio.

Impulse noise (sometimes called *spikes*) is the primary source of errors in data communications. Impulse noise is heard as a click or a crackling noise and can last as long as $\frac{1}{100}$ of a second. Such a click does not really affect voice communications, but it can obliterate a group of data, causing a burst error. At 1.5 Mbps, 15,000 bits would be changed by a spike of $\frac{1}{100}$ of a second. Some of the sources of impulse noise are voltage changes in adjacent lines, lightning flashes during thunderstorms, fluorescent lights, and poor connections in circuits.

Cross-talk occurs when one circuit picks up signals in another. You experience cross-talk during telephone calls when you hear other conversations in the background. It occurs between pairs of wires that are carrying separate signals, in multiplexed links carrying many discrete signals, or in microwave links in which one antenna picks up a minute reflection from another antenna. Cross-talk between lines increases with increased communication distance, increased proximity of the two wires, increased signal strength, and higher-frequency signals. Wet or damp weather can also increase cross-talk. Like white noise, cross-talk has such a low signal strength that it normally is not bothersome.

Echoes can cause errors. Echoes are caused by poor connections that cause the signal to reflect back to the transmitting equipment. If the strength of the echo is strong enough to be detected, it causes errors. Echoes, like cross-talk and white noise, have such a low signal strength that they normally are not bothersome. Echoes can also occur in fiber-optic cables when connections between cables are not properly aligned.

Attenuation is the loss of power a signal suffers as it travels from the transmitting computer to the receiving computer. Some power is absorbed by the medium or is lost before it reaches the receiver. As the medium absorbs power, the signal becomes weaker, and the receiving equipment has less and less chance of correctly interpreting the data. This power loss is a function of the transmission method and circuit medium. High frequencies lose power more rapidly than do low frequencies during transmission, so the received signal can thus be distorted by unequal loss of its component frequencies. Attenuation increases as frequency increases or as the diameter of the wire decreases.

Intermodulation noise is a special type of cross-talk. The signals from two circuits combine to form a new signal that falls into a frequency band reserved for another signal. This type of noise is similar to harmonics in music. On a multiplexed line, many different signals are amplified together, and slight variations in the adjustment of the equipment can cause intermodulation noise. A maladjusted modem may transmit a strong frequency tone when not transmitting data, thus producing this type of noise.

Jitter may affect the accuracy of the data being transmitted because minute variations in amplitude, phase, and frequency always occur. The generation of a pure carrier signal in an analog circuit is impossible. The signal may be impaired by continuous and rapid gain and/or phase changes. This jitter may be random or periodic. Phase jitter during a telephone call causes the voice to fluctuate in volume.

Harmonic distortion usually is caused by an amplifier on a circuit that does not correctly represent its output with what was delivered to it on the input side. *Phase hits* are short-term shifts “out of phase,” with the possibility of a shift back into phase.

Error Prevention

There are many techniques to prevent errors (or at least reduce them), depending on the situation. *Shielding* (protecting wires by covering them with an insulating coating) is one of the best ways to prevent impulse noise, cross-talk, and intermodulation noise. Many different types of wires and cables are available with different amounts of shielding. In general, the greater the shielding, the more expensive the cable and the more difficult it is to install.

Moving cables away from sources of noise (especially power sources) can also reduce impulse noise, cross-talk, and intermodulation noise. For impulse noise, this means avoiding lights and heavy machinery. Locating communication cables away from power cables is always a good idea. For cross-talk, this means physically separating the cables from other communication cables.

Cross-talk and intermodulation noise is often caused by improper multiplexing. Changing multiplexing techniques (e.g., from FDM to TDM) or changing the frequencies or size of the guardbands in FDM can help.

Many types of noise (e.g., echoes, white noise, jitter, harmonic distortion) can be caused by poorly maintained equipment or poor connections and splices among cables. This is particularly true for echo in fiber-optic cables, which is almost always caused by poor connections. The solution here is obvious: Tune the transmission equipment and redo the connections.

To avoid attenuation, telephone circuits have *repeaters* or *amplifiers* spaced throughout their length. The distance between them depends on the amount of power lost per unit length of the transmission line. An amplifier takes the incoming signal, increases

MANAGEMENT FOCUS

4-1 FINDING THE SOURCE OF IMPULSE NOISE

Several years ago, the University of Georgia radio station received FCC (Federal Communications Commission) approval to broadcast using a stronger signal. Immediately after the station started broadcasting with the new signal, the campus backbone network (BN) became unusable because of impulse noise. It took 2 days to link the impulse noise to the radio station, and when the radio station returned to its usual broadcast signal, the problem disappeared.

However, this was only the first step in the problem. The radio station wanted to broadcast at full strength, and there was no good reason for why the stronger broadcast should affect the BN in this way. After 2 weeks of effort, the problem was discovered. A short section of the BN ran above ground between two buildings. It turned out that the specific brand of outdoor cable we used was particularly tasty to squirrels. They had eaten the outer insulating coating off of the cable, making it act like an antennae to receive the radio signals. The cable was replaced with a steel-coated armored cable so the squirrels could not eat the insulation. Things worked fine when the radio station returned to its stronger signal.

its strength, and retransmits it on the next section of the circuit. They are typically used on analog circuits such as the telephone company's voice circuits. The distance between the amplifiers depends on the amount of attenuation, although 1- to 10-mile intervals are common. On analog circuits, it is important to recognize that the noise and distortion are *also* amplified, along with the signal. This means some noise from a previous circuit is regenerated and amplified each time the signal is amplified.

Repeaters are commonly used on digital circuits. A repeater receives the incoming signal, translates it into a digital message, and retransmits the message. Because the message is recreated at each repeater, noise and distortion from the previous circuit are not amplified. This provides a much cleaner signal and results in a lower error rate for digital circuits.

If the circuit is provided by a common carrier such as the telephone company, you can lease a more expensive *conditioned* circuit. A conditioned circuit is one that has been certified by the carrier to experience fewer errors. There are several levels of conditioning that provide increasingly fewer errors at increasingly higher cost. Conditioned circuits employ a variety of the techniques described previously (e.g., shielding) to provide less noise.

Error Detection

It is possible to develop data transmission methodologies that give very high *error detection and correction* performance. The only way to do error detection and correction is to send extra data with each message. These error detection data are added to each message by the data link layer of the sender on the basis of some mathematical calculations performed on the message (in some cases, error-detection methods are built into the hardware itself). The receiver performs the same mathematical calculations on the message it receives and matches its results against the error-detection data that were transmitted with the message. If the two match, the message is assumed to be correct. If they don't match, an error has occurred.

In general, the larger the amount of error-detection data sent, the greater the ability to detect an error. However, as the amount of error-detection data is increased, the throughput of useful data is reduced, because more of the available capacity is used to transmit these error-detection data and less is used to transmit the actual message itself. Therefore, the efficiency of data throughput varies inversely as the desired amount of error detection is increased.

Three well-known *error-detection methods* are parity checking, checksum, and cyclic redundancy checking.

Parity Checking One of the oldest and simplest error-detection methods is *parity.* With this technique, one additional bit is added to each byte in the message. The value of this additional *parity bit* is based on the number of 1's in each byte transmitted. This parity bit is set to make the total number of 1's in the byte (including the parity bit) either an even number or an odd number. Figure 4.3 gives an example.

A little thought will convince you that any single error (a switch of a 1 to a 0 or vice versa) will be detected by parity, but it cannot determine which bit was in error. You will know an error occurred, but not what the error was. But if *two* bits are switched, the *parity check* will not detect any error. It is easy to see that parity can detect errors only when an odd number of bits have been switched; any even number of errors cancel one another out. Therefore, the probability of detecting an error, given that one has occurred, is only about 50 percent. Many networks today do not use parity because of its low error-detection rate. When parity is used, protocols are described as having *odd parity* or *even parity.*

Checksum With the *checksum* technique, a checksum (typically 1 byte) is added to the end of the message. The checksum is calculated by adding the decimal value of each character in the message, dividing the sum by 255, and using the remainder as the checksum. The receiver calculates its own checksum in the same way and compares it with the transmitted checksum. If the two values are equal, the message is presumed to contain no errors. Use of checksum detects close to 95 percent of the errors for multiple-bit burst errors.

Cyclical Redundancy Check One of the most popular error-checking schemes is *cyclical redundancy check (CRC).* It adds 8, 16, 24, or 32 bits to the message. With CRC,

Assume we are using even parity with 8-bit ASCII.
The letter *V* in 8-bit ASCII is encoded as 01101010.
Because there are four 1's (an even number), parity is set to 0.
This would be transmitted as 011010100.

Assume we are using even parity with 8-bit ASCII.
The letter *W* in 8-bit ASCII is encoded as 00011010.
Because there are three 1's (an odd number), parity is set to 1.
This would be transmitted as 000110101.

FIGURE 4.3 Using parity for error detection.

a message is treated as one long binary number, *P.* Before transmission, the data link layer (or hardware device) divides *P* by a fixed binary number, *G,* resulting in a whole number, *Q,* and a remainder, *R/G.* So, $P/G = Q + R/G$. For example, if $P = 58$ and $G = 8$, then $Q = 7$ and $R = 2$. *G* is chosen so that the remainder, *R,* will be either 8 bits, 16 bits, 24 bits, or 32 bits.[2]

The remainder, *R,* is appended to the message as the error-checking characters before transmission. The receiving hardware divides the received message by the same *G,* which generates an *R.* The receiving hardware checks to ascertain whether the received *R* agrees with the locally generated *R.* If it does not, the message is assumed to be in error.

CRC performs quite well. The most commonly used CRC codes are CRC-16 (a 16-bit version), CRC-CCITT (another 16-bit version), and CRC-32 (a 32-bit version). The probability of detecting an error is 100 percent for all errors of the same length as the CRC or less. For example, CRC-16 is guaranteed to detect errors if 16 or fewer bits are affected. If the burst error is longer than the CRC, then CRC is not perfect but is close to it. CRC-16 will detect about 99.998 percent of all burst errors longer than 16 bits, whereas CRC-32 will detect about 99.99999998 percent of all burst errors longer than 32 bits.

Error Correction via Retransmission

Once error has been detected, it must be corrected. The simplest, most effective, least expensive, and most commonly used method for error correction is retransmission. With retransmission, a receiver that detects an error simply asks the sender to retransmit the message until it is received without error. This is often called Automatic Repeat reQuest (ARQ). There are two types of ARQ: stop-and-wait and continuous.

Stop-and-Wait ARQ With *stop-and-wait ARQ,* the sender stops and waits for a response from the receiver after each data packet. After receiving a packet, the receiver sends either an *acknowledgement (ACK),* if the packet was received without error, or a *negative acknowledgment (NAK),* if the message contained an error. If it is an NAK, the sender resends the previous message. If it is an ACK, the sender continues with the next message. Stop-and-wait ARQ is by definition a half-duplex transmission technique (Figure 4.4).

Continuous ARQ With *continuous ARQ,* the sender does not wait for an acknowledgment after sending a message; it immediately sends the next one. Although the messages are being transmitted, the sender examines the stream of returning acknowledgments. If it receives an NAK, the sender retransmits the needed messages. The packets that are retransmitted may be only those containing an error (called *Link Access Protocol for Modems [LAP-M]*) or may be the first packet with an error and all those that followed it (called *Go-Back-N ARQ*). LAP-M is better because it is more efficient.

Continuous ARQ is by definition a full-duplex transmission technique, because both the sender and the receiver are transmitting simultaneously. (The sender is sending mes-

[2]CRC is actually more complicated than this because it uses polynominal division, not "normal" division as illustrated here. Ross Willams provides an excellent tutorial on CRC at www.ross.net/crc/crcpaper.html.

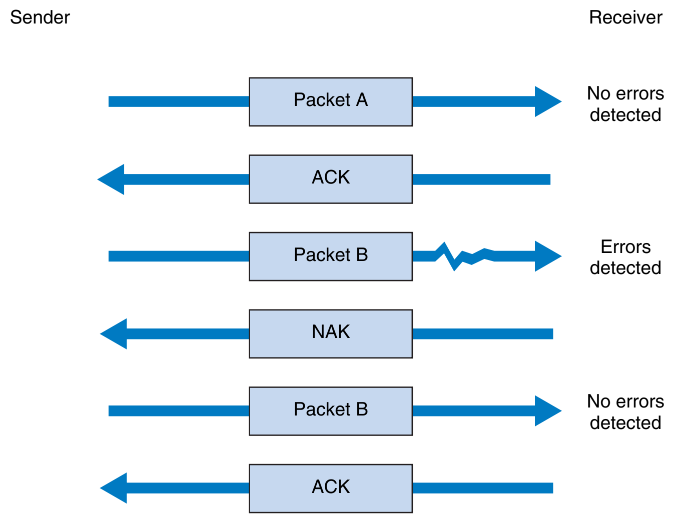


FIGURE 4.4 Stop-and-wait ARQ (Automatic Repeat reQuest). ACK = acknowledgment; NAK = negative acknowledgment.

sages, and the receiver is sending ACKs and NAKs.) Figure 4.5 illustrates the flow of messages on a communication circuit using continuous ARQ. Continuous ARQ is sometimes called *sliding window* because of the visual imagery the early network designers used to think about continuous ARQ. Visualize the sender having a set of messages to send in memory stacked in order from first to last. Now imagine a window that moves through the stack from first to last. As a message is sent, the window expands to cover it, meaning that the sender is waiting for an ACK for the message. As an ACK is received for a message, the window moves forward, dropping the message out of the bottom of the window, indicating that it has been sent and received successfully.

Both stop-and-wait ARQ and continuous ARQ are also important in providing *flow control,* which means ensuring that the computer sending the message is not transmitting too quickly for the receiver. For example, if a client computer was sending information too quickly for a server computer to store a file being uploaded, the server might run out of memory to store the file. By using ACKs and NAKs, the receiver can control the rate at which it receives information. With stop-and-wait ARQ, the receiver does not send an ACK until it is ready to receive more packets. In continuous ARQ, the sender and receiver usually agree on the size of the sliding window. Once the sender has transmitted the maximum number of packets permitted in the sliding window, it cannot send any more packets until the receiver sends an ACK.

Forward Error Correction

Forward error correction uses codes containing sufficient redundancy to prevent errors by detecting and correcting them at the receiving end *without* retransmission of the original message. The redundancy, or extra bits required, varies with different schemes. It ranges

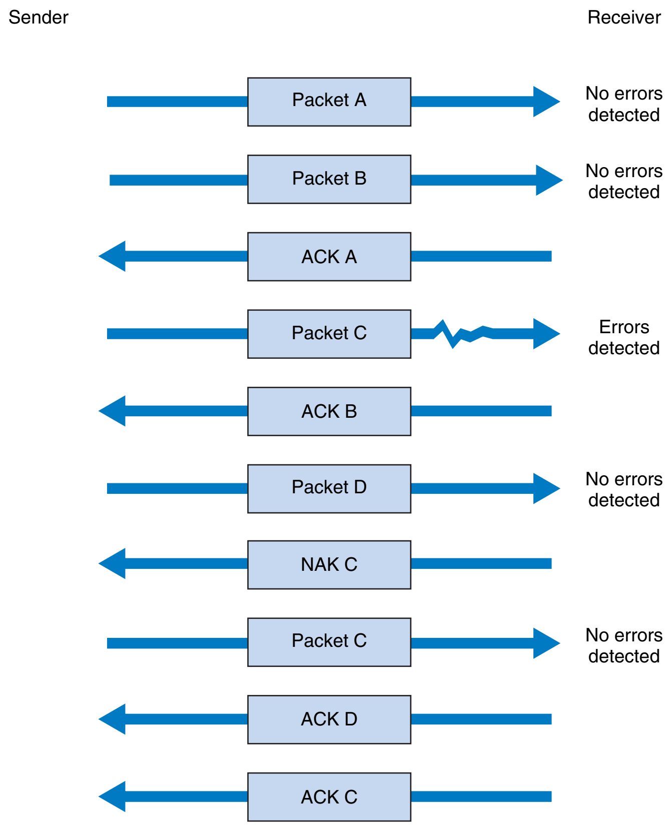


FIGURE 4.5 Continuous ARQ (Automatic Repeat reQuest). ACK = acknowledgment; NAK = negative acknowledgment.

from a small percentage of extra bits to 100 percent redundancy, with the number of error-detecting bits roughly equaling the number of data bits. One of the characteristics of many error-correcting codes is that there must be a minimum number of error-free bits between bursts of errors.

Forward error correction is commonly used in satellite transmission. A round trip from the earth station to the satellite and back includes a significant delay. Error rates can fluctuate depending on the condition of equipment, sunspots, or the weather. Indeed, some weather conditions make it impossible to transmit without some errors, making forward error correction essential. Compared with satellite equipment costs, the additional cost of forward error correction is insignificant.

Error Control in Practice

In the OSI model (see Chapter 1), error control is defined to be a layer-2 function—it is the responsibility of the data link layer. However, in practice, we have moved away from

TECHNICAL FOCUS

4-1 How Forward Error Correction Works

To see how error-correcting codes work, consider the example of a forward error checking code in Figure 4.6, called a *Hamming code,* after its inventor, R. W. Hamming. This code is a very simple approach, capable of correcting 1-bit errors. More sophisticated techniques (e.g., Reed–Solomon) are commonly used today, but this will give you a sense of how they work.

The Hamming code associates even parity bits with unique combinations of data bits. With a 4-data-bit code as an example, a character might be represented by the data-bit configuration 1010. Three parity bits, P_1, P_2, and P_4, are added, resulting in a 7-bit code, shown in the upper half of Figure 4.6. Notice that the data bits (D_3, D_5, D_6, D_7) are 1010 and the parity bits (P_1, P_2, P_4) are 101.

As depicted in the upper half of Figure 4.6, parity bit P_1 applies to data bits D_3, D_5, and D_7. Parity bit P_2 applies to data bits D_3, D_6, and D_7. Parity bit P_4 applies to data bits D_5, D_6, and D_7. For the example, in which D_3, D_5, D_6, D_7 = 1010, P_1 must equal 1 because there is only a single 1 among D_3, D_5 and D_7 and parity must be even. Similarly, P_2 must be 0 because D_3 and D_6 are 1's. P_4 is 1 because D_6 is the only 1 among D_5, D_6, and D_7.

Now, assume that during the transmission, data bit D_7 is changed from a 0 to a 1 by line noise. Because this data bit is being checked by P_1, P_2, and P_4, all 3 parity bits now show odd parity instead of the correct even parity. D_7 is the only data bit that is monitored by all 3 parity bits; therefore, when D_7 is in error, all 3 parity bits show an incorrect parity. In this way, the receiving equipment can determine which bit was in error and reverse its state, thus correcting the error without retransmission.

The lower half of the figure is a table that determines the location of the bit in error. A 1 in the table means that the corresponding parity bit indicates a parity error. Conversely, a 0 means the parity check is correct. These 0's and 1's form a binary number that indicates the numeric location of the erroneous bit. In the previous example, P_1, P_2, and P_4 checks all failed, yielding 111, or a decimal 7, the subscript of the erroneous bit.

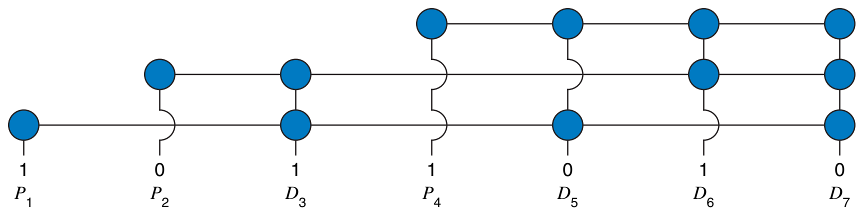
this. Most network hardware and software available today provide an error control function at the data link layer, but it is turned off. Most network cables are very reliable and errors are far less common than they were in the 1980s.

Therefore, most data link layer software today is configured to detect errors, but *not* correct them. Any time a packet with an error is discovered, it is simply discarded. The exceptions to this tend to be wireless technologies and a few WAN technologies where errors are more common.

The implication from this is that error correction must be performed by software at higher layers. This software must be able to detect lost packets (i.e., those that have been discarded) and request the sender to retransmit them. This is commonly done by the transport layer using continuous ARQ as we shall see in the next chapter.

DATA LINK PROTOCOLS

In this section, we outline several commonly used data link layer protocols, which are summarized in Figure 4.7. Here we focus on message delineation, which indicates where



Checking relations between parity bits (P) and data bits (D)

0 = Corresponding parity check is correct

1 = Corresponding parity check fails

P_4	P_2	P_1	Determines in which bit the error occured
0	0	0	no error
0	0	1	P_1
0	1	0	P_2
0	1	1	D_3
1	0	0	P_4
1	0	1	D_5
1	1	0	D_6
1	1	1	D_7

Interpreting parity bit patterns

FIGURE 4.6 Hamming code for forward error correction.

Protocol	Size	Error Detection	Retransmission	Media Access
Asynchronous transmission	1	Parity	Continuous ARQ	Full Duplex
File transfer protocols				
Xmodem	132	8-bit Checksum	Stop-and-wait ARQ	Controlled Access
Xmodem-CRC	132	8-bit CRC	Stop-and-wait ARQ	Controlled Access
Xmodem-1K	1,028	8-bit CRC	Stop-and-wait ARQ	Controlled Access
Zmodem	*	32-bit CRC	Continuous ARQ	Controlled Access
Synchronous protocols				
SDLC	*	16-bit CRC	Continuous ARQ	Controlled Access
HDLC	*	16-bit CRC	Continuous ARQ	Controlled Access
Ethernet	*	32-bit CRC	Stop-and-wait ARQ	Contention
PPP	*	16-bit CRC	Continuous ARQ	Full Duplex

*Varies depending on the message length.

ARQ = Automatic Repeat reQuest; CRC = cyclical redundancy check; HDLC = high-level data link control; PPP = Point-to-Point Protocol; SDLC = synchronous data link control.

FIGURE 4.7 Protocol summary.

a message starts and stops, and the various parts or *fields* within the message. For example, you must clearly indicate which part of a message or packet of data is the error-control portion; otherwise, the receiver cannot use it properly to determine if an error has occurred.

Asynchronous Transmission

Asynchronous transmission often is referred to as start–stop transmission because the transmitting computer can transmit a character whenever it is convenient, and the receiving computer will accept that character. It is typically used on point-to-point full-duplex circuits (i.e., circuits that have only two computers on them), so media access control is not a concern. If you use VT100 protocol, or connect to a UNIX or Linux computer using Telnet, chances are you are using asynchronous transmission.

With *asynchronous transmission,* each character is transmitted independently of all other characters. To separate the characters and synchronize transmission, a *start bit* and a *stop bit* are put on the front and back of *each* individual character. For example, if we are using 7-bit ASCII with even parity, the total transmission is 10 bits for each character (1 start bit, 7 bits for the letter, 1 parity bit, 1 stop bit).

The start bit and stop bit are the opposite of each other. Typically, the start bit is a 0 and the stop bit is a 1. There is no fixed distance between characters because the terminal transmits the character as soon as it is typed, which varies with the speed of the typist. The recognition of the start and stop of each message (called *synchronization*) takes place for each individual character because the start bit is a signal that tells the receiver to start sampling the incoming bits of a character so the data bits can be interpreted into their proper character structure. A stop bit informs the receiver that the character has been received and resets it for recognition of the next start bit.

When the sender is waiting for the user to type the next character, no data is sent; the communication circuit is idle. This idle time really is artificial—some signal always must be sent down the circuit. For example, suppose we are using a unipolar digital signaling technique where +5 volts indicates a 1 and 0 volts indicates a 0 (see Chapter 3). Even if we send 0 volts, we are still sending a signal, a 0 in this case. Asynchronous transmission defines the *idle signal* (the signal that is sent down the circuit when no data are being transmitted) as the same as the stop bit. When the sender finishes transmitting a letter and is waiting for more data to send, it sends a continuous series of stop bits. Figure 4.8 shows an example of asynchronous transmission.

Some older protocols have two stop bits instead of the traditional single stop bit. The use of both a start bit and a stop bit is changing; some protocols have eliminated the stop bit altogether.

Asynchronous File Transfer Protocols

Today, data transmission by microcomputers often means the transfer of data files. In general, microcomputer file transfer protocols are used on asynchronous point-to-point circuits, typically across telephone lines via a modem. All file transfer protocols have two characteristics in common. First, these protocols are designed to transmit error-free data from one computer to another. Second, because there is a large amount of data to be trans-

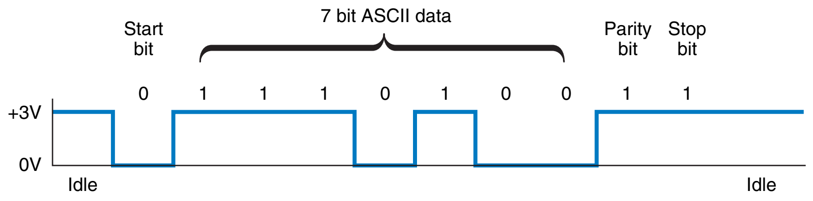


FIGURE 4.8 Asynchronous transmission. ASCII = United States of America Standard Code for Information Interchange.

mitted, it makes more sense to group the data together into blocks of data that are transmitted at the same time, rather than sending each character individually via standard asynchronous transmission. This section discusses the structure of the data blocks (also called packets or frames) used by several common protocols.

Xmodem The *Xmodem* protocol takes the data being transmitted and divides it into blocks (Figure 4.9). Each block has a start-of-header (SOH) character, a 1-byte block number, 128 bytes of data, and a 1-byte checksum for error checking. Even though this protocol was developed for microcomputer-to-microcomputer communications, it often is used for microcomputer-to-mainframe communications. Xmodem uses stop-and-wait ARQ.

Xmodem-CRC improves error detection accuracy of the Xmodem protocol. It replaces the checksum with a more rigorous 1-byte cyclical redundancy check (CRC-8).

Xmodem-1K increases the efficiency of Xmodem-CRC by using data blocks of 1,024 bytes instead of the 128-character blocks of the original Xmodem. Efficiency and throughput are discussed in more detail later in this chapter.

Zmodem *Zmodem* is a newer protocol and not a subset of Xmodem. It incorporates features of several protocols. It uses a more powerful error-detection method (CRC-32) with continuous ARQ. Zmodem also dynamically adjusts its packet size according to communication circuit conditions to increase efficiency. Usually Zmodem is preferred to Xmodem.

Synchronous Transmission

With *synchronous transmission,* all the letters or data in one group of data is transmitted at one time as a block of data. This block of data is called a *frame* or *packet,* depending on

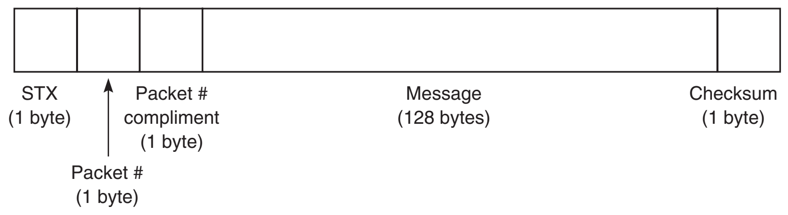


FIGURE 4.9 Xmodem format.

the protocol, but the meaning is the same. For example, a terminal or microcomputer will save all the keystrokes typed by the user and transmit them only when the user presses a special "transmit" key. In this case, the start and end of the entire packet must be marked, not the start and end of each letter. Synchronous transmission is often used on both point-to-point and multipoint. For multipoint circuits, each packet must include a destination address and a source address, and media access control is important.

The start and end of each packet (synchronization) sometimes is established by adding synchronization characters (SYN) to the start of the packet. Depending on the protocol, there may be anywhere from one to eight SYN characters. After the SYN characters, the transmitting computer sends a long stream of data that may contain thousands of bits. Knowing what code is being used, the receiving computer counts off the appropriate number of bits for the first character, assumes this is the first character, and passes it to the computer. It then counts off the bits for the second character, and so on.

In summary, asynchronous data transmission means each character is transmitted as a totally independent entity with its own start and stop bits to inform the receiving computer that the character is beginning and ending. Synchronous transmission means whole blocks of data are transmitted as packets after the sender and the receiver have been synchronized.

There are many protocols for synchronous transmission. They fall into three broad categories: byte-oriented protocols, bit-oriented protocols, and byte-count protocols. In this next section, we discuss four common synchronous data link protocols.

Synchronous Data Link Control *Synchronous data link control (SDLC)* is a mainframe protocol developed by IBM in 1972 that is still in use today. SDLC is a *bit-oriented protocol,* because the data contained in the frame do not have to be in 8-bit bytes. SDLC is therefore more flexible than byte-oriented protocols. It uses a controlled-access media access protocol. If you use a 3270 protocol, you're using SDLC.

Figure 4.10 shows a typical SDLC packet (or *frame,* as it is called). Each SDLC frame begins and ends with a special bit pattern (01111110), known as the *flag.* The *address field* identifies the destination. The length of the address field is usually 8 bits but can be set at 16 bits; all computers on the same network must use the same length. The *control field* identifies the kind of frame that is being transmitted, either information or supervisory. An *information frame* is used for the transfer and reception of messages, frame numbering of contiguous frames, and the like. A *supervisory frame* is used to transmit acknowledgments (ACKs and NAKs). The *message field* is of variable length and is the user's message. The *frame check sequence field* is a 32-bit CRC code (some older versions use a 16-bit CRC).

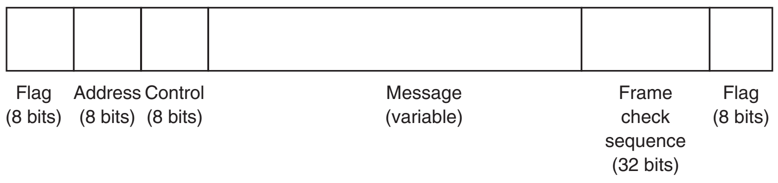


FIGURE 4.10 SDLC (synchronous data link control) format.

SDLC and other bit-oriented protocols suffer from a *transparency problem;* that is, the protocol is not "transparent" because it cannot automatically send all types of data with any bit patterns. It is possible that the user's data to be transmitted contains the same bit pattern as the flag (01111110). If this is not prevented, the receiver will mistakenly believe that this data marks the end of the frame and ignore all the data that follows it. The solution is called *bit stuffing.* Anytime the sender encounters five 1's in a row in the *user's data* to be transmitted, the sender "stuffs" one extra bit, a 0, into the message and continues to transmit. Anytime the receiver encounters five 1's followed by a 0 (i.e., 111110), the receiver automatically deletes the 0 and continues to process the data stream. Conversely, if the receiver encounters five 1's followed by a 1 (i.e., 111111) it knows to expect another 0 as part of the flag. This technique works, but it increases the complexity of the protocol.

High-Level Data Link Control *High-level data link control (HDLC)* is a formal standard developed by the ISO. HDLC is essentially the same as SDLC, except that the address and control fields can be longer. HDLC also has several additional benefits that are beyond the scope of this book, such as a larger sliding window for continuous ARQ. It uses a controlled-access media access protocol. One variant, *Link Access Protocol–Balanced (LAP-B)*, uses the same structure as HDLC but is a scaled-down version of HDLC (i.e., provides fewer of those benefits mentioned that are "beyond the scope of this book").

Ethernet (IEEE 802.3) Ethernet is a very popular LAN protocol, conceived by Bob Metcalfe in 1973 and developed jointly by Digital, Intel, and Xerox in the 1970s. Since then, Ethernet has been further refined and developed into a formal standard called IEEE 802.3 ac.[3] Ethernet is a *byte-count protocol* because instead of using special characters or bit patterns to mark the end of a packet, it includes a field that specifies the length of the message portion of the packet. Unlike SDLC and HDLC, Ethernet has no transparency problems. Any bit pattern can be transmitted, because Ethernet uses the number of bytes, not control characters, to delineate the message. Ethernet uses a contention media access protocol.

Figure 4.11 shows a typical Ethernet packet. The packet starts with a 7-byte *preamble* which is a repeating pattern of ones and zeros (10101010). This is followed by a *start of frame delimiter,* which acts like the flag in SDLC to mark the start of the frame. The *destination address* specifies the receiver, whereas the *source address* specifies the sender. The *length* indicates the length in 8-bit bytes of the message portion of the packet. The

Preamble	Start of Frame	Destination Address	Source Address	VLAN Tag	Length	DSAP	SSAP	Control	Data	Frame check sequence
7 bytes	1 byte	6 bytes	6 bytes	4 bytes	2 bytes	1 byte	1 byte	1-2 bytes	43-1497 bytes	4 bytes

FIGURE 4.11 Ethernet 802.3ac packet layout.

[3]A competing version of Ethernet called Ethernet II is also available. Ethernet II and IEEE 802.3 Ethernet are similar but differ enough to be incompatible. In this book, we discuss only IEEE 802.3 Ethernet.

VLAN tag field is an optional 4-byte address field used by virtual LANs (VLANs), which are discussed in Chapter 8. The Ethernet packet uses this field only when VLANs are in use; otherwise the field is omitted, and the length field immediately follows the source address field. When the VLAN tag field is in use, the first 2 bytes are set to the number 24,832 (hexadecimal 81-00), which is obviously an impossible packet length. When Ethernet sees this length, it knows that the VLAN tag field is in use. When the length is some other value, it assumes that VLAN tags are not in use and that the length field immediately follows the source address field. The *DSAP* and *SSAP* are used to pass control information between the sender and receiver. These are often used to indicate the type of network layer protocol the packet contains (e.g., TCP/IP or IPX/SPX, as described in Chapter 5). The *control field* is used to hold the packet sequence numbers and ACKs and NAKs used for error control, as well as to enable the data link layers of communicating computers to exchange other control information. The last 2 bits in the first byte are used to indicate the type of control information being passed and whether the control field is 1 or 2 bytes (e.g., if the last 2 bits of the control field are 11, then the control field is 1 byte in length). In most cases, the control field is 1-byte long. The maximum length of the message is 1,500 bytes. The packet ends with a CRC-32 *frame check sequence* used for error detection.

A Day in the Life: Network Support Technician

When a help call arrives at the help desk, the help desk staff (first-level support) spends up to 10 minutes attempting to solve the problem. If they can't, then the problem is passed to the second-level support, the network support technician.

A typical day in the life of a network support technician starts by working on computers from the day before. Troubleshooting usually begins with a series of diagnostic tests to eliminate hardware problems. The next step, for a laptop, is to remove the hard disk and replace it with a hard disk containing a correct standard image. If the computer passes those tests then the problem is usually software. Then the fun begins.

Once a computer has been fixed it is important to document all the hardware and/or software changes to help track problem computers or problem software. Sometimes a problem is new but relatively straightforward to correct once it has been diagnosed. In this case, the technician will change the standard support process followed by the technicians working at the help desk to catch the problem before it is escalated to the network support technicians. In other cases, a new entry is made into the organization's technical support knowledge base so that if another technician (or user) encounters the problem it is easier for him or her to diagnose and correct the problem. About 10% of the time of the network technician is spent documenting solutions to problems.

Network support technicians also are the ones who manage new inventory and set up and configure new computers as they arrive from the manufacturer. They are also the ones responsible for deploying new software and standard desktop images across the network. Many companies also set aside standard times for routine training; in our case, every Friday, several hours is devoted to regular training.

With thanks to Doug Strough

Point-to-Point Protocol *Point-to-Point Protocol (PPP)* is a byte-oriented protocol developed in the early 1990s that is used to dial up from home computers to an ISP. It is designed to transfer data over a point-to-point telephone line but provides an address so that it can be used on multipoint circuits. The *message* may be up to 1,500 bytes in length. PPP uses CRC-16 for error control.

TRANSMISSION EFFICIENCY

One objective of a data communication network is to move the highest possible volume of accurate information through the network. The higher the volume, the greater the resulting network's efficiency and the lower the cost. Network efficiency is affected by characteristics of the circuits such as error rates and maximum transmission speed, as well as by the speed of transmitting and receiving equipment, the error-detection and control methodology, and the protocol used by the data link layer.

Each protocol we discussed uses some bits or bytes to delineate the start and end of each message and to control error. These bits and bytes are necessary for the transmission to occur, but they are not part of the message. They add no value to the user, but they count against the total number of bits that can be transmitted.

Each communication protocol has both information bits and overhead bits. *Information bits* are those used to convey the user's meaning. *Overhead bits* are used for purposes such as error checking and marking the start and end of characters and packets. A parity bit used for error checking is an overhead bit because it is not used to send the user's data; if you did not care about errors, the overhead error checking bit could be omitted and the users could still understand the message.

Transmission efficiency is defined as the total number of information bits (i.e., bits in the message sent by the user) divided by the total bits in transmission (i.e., information bits plus overhead bits). For example, let's calculate the transmission efficiency of asynchronous transmission. Assume we are using 7-bit ASCII. We have 1 bit for parity, plus 1 start bit and 1 stop bit. Therefore, there are 7 bits of information in each letter, but the total bits per letter is 10 (7 + 3). The efficiency of the asynchronous transmission system is 7 bits of information divided by 10 total bits, or 70 percent.

In other words, with asynchronous transmission, only 70 percent of the data rate is available for the user; 30 percent is used by the transmission protocol. If we have a communication circuit using a dial-up modem receiving 56 Kbps, the user sees an effective data rate (or throughput) of 39.2 Kbps. This is very inefficient.

We can improve efficiency by reducing the number of overhead bits in each message or by increasing the number of information bits. For example, if we remove the stop bits from asynchronous transmission, efficiency increases to $\frac{7}{9}$, or 77.8 percent. The throughput of a dial-up modem at 56 Kbps would increase 43.6 Kbps, which is not great but is at least a little better.

The same basic formula can be used to calculate the efficiency of asynchronous file transfer or synchronous transmission. For example, suppose we are using SDLC. The number of information bits is calculated by determining how many information characters are in the message. If the message portion of the frame contains 100 information characters and we are using an 8-bit code, then there are $100 \times 8 = 800$ bits of information. The total number of bits is the 800 information bits plus the overhead bits that are inserted for delineation and error control. Figure 4.10 shows that SDLC has a beginning flag (8 bits), an address (8 bits), a control field (8 bits), a frame check sequence (assume we use a CRC-32 with 32 bits), and an ending flag (8 bits). This is a total of 64 overhead bits; thus, efficiency is 800/(800 + 64) = 92.6 percent. If the circuit provides a data rate of 56 Kbps, then the effective data rate available to the user is about 51.9 Kbps.

This example shows that synchronous networks usually are more efficient than asynchronous networks and some protocols are more efficient than others. The longer the message (1,000 characters as opposed to 100), the more efficient the protocol. For example, suppose the message in the SDLC example were 1,000 bytes. The efficiency here would be 99.2 percent, or 8,000/(8000 + 64), giving an effective data rate of about 55.6 Kbps.

This example should also show why Zmodem (with a message length of 1,024 bytes) is more efficient than Xmodem (with a message length of 128 bytes). The general rule is that the larger the message field, the more efficient the protocol.

So why not have 10,000-byte or even 100,000-byte packets to really increase efficiency? The answer is that anytime a packet is received containing an error, the entire packet must be retransmitted. Thus, if an entire file is sent as one large packet (e.g., 100K) and 1 bit is received in error, all 100,000 bytes must be sent again. Clearly, this is a waste of capacity. Furthermore, the probability that a packet contains an error increases with the size of the packet; larger packets are more likely to contain errors than are smaller ones, simply due to the laws of probability.

Thus, in designing a protocol, there is a trade-off between large and small packets. Small packets are less efficient but are less likely to contain errors and cost less (in terms of circuit capacity) to retransmit if there is an error (Figure 4.12).

Throughput is the total number of information bits received per second, after taking into account the overhead bits and the need to retransmit packets containing errors. Generally speaking, small packets provide better throughput for circuits with more errors, whereas larger packets provide better throughput in less-error-prone networks. Fortu-

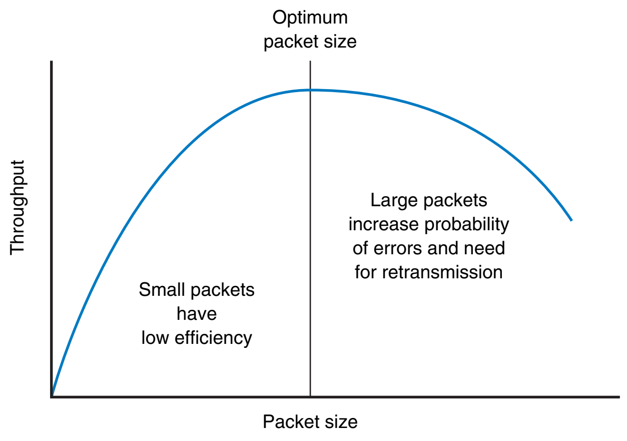


FIGURE 4.12 Packet size effects on throughput.

MANAGEMENT FOCUS

4-2 SLEUTHING FOR THE RIGHT PACKET SIZE

Optimizing performance in a network, particularly a client–server network, can be difficult because few network managers realize the importance of the packet size. Selecting the right—or the wrong—packet size can have greater effects on performance than anything you might do to the server.

Standard Commercial, a multinational tobacco and agricultural company, noticed a decrease in network performance when they upgraded to a new server. They tested the effects of using packet sizes between 500 bytes to 32,000 bytes. In their tests, a packet size of 512 bytes required a total of 455,000 bytes transmitted over their network to transfer the test messages. In contrast, the 32,000-byte packets were far more efficient, cutting the total data by 44 percent to 257,000 bytes.

However, the problem with 32,000-byte packets was a noticeable response time delay because messages were saved until the 32,000-byte packets were full before transmitting.

The ideal packet size depends on the specific application and the pattern of messages it generates. For Standard Commercial, the ideal packet size appeared to be between 4,000 and 8,000. Unfortunately, not all network software packages enable network managers to fine-tune packet sizes in this way.

SOURCE: "Sleuthing for the Right Packet Size," *InfoWorld,* January 16, 1995.

nately, in most real networks, the curve shown in Figure 4.12 is very flat on top, meaning that there is a range of packet sizes that provide almost optimum performance. Packet sizes vary greatly among different networks, but most packet sizes tend to be between 2,000 and 4,000 bytes.

Calculating the actual *throughput* of a data communications network is complex because it depends not only on the efficiency of the data link protocol but also on the error rate and number of retransmissions that occur. *Transmission rate of information bits (TRIB)* is a measure of the effective number of information bits that is transmitted over a communication circuit per unit of time. The basic TRIB equation from ANSI is shown in Figure 4.13, along with an example.

IMPLICATIONS FOR MANAGEMENT

You can think of the data link layer protocol as the fundamental "language" spoken by networks. This protocol must be compatible with the physical cables that are used, but in many cases the physical cables can support a variety of different protocols. Each device on the network speaks a particular data link layer protocol. In the past, there were literally dozens of protocols that were used; each protocol was custom-tailored to specific needs of the devices and application software in use. Where different devices or cables from different parts of the organization were connected, we used a translator to convert from the data link protocol spoken by one device into the protocol spoken by another device.

As the Internet has become more prominent and as it has become more important to move data from one part of an organization to the other, the need to translate among different data link layer protocols has become more and more costly. It is now more important to

FORMULA FOR CALCULATING TRIB

$$\text{TRIB} = \frac{\text{Number of information bits accepted}}{\text{Total time required to get the bits accepted}}$$

$$\text{TRIB} = \frac{K(M - C)\,(1 - P)}{(M/R) + T}$$

where K = information bits per character
M = packet length in characters
R = data transmission rate in characters per second
C = average number of noninformation characters per block (control characters)
P = probability that a block will require retransmission because of error
T = time between blocks in seconds, such as modem delay/turnaround time on half duplex, echo suppressor delay on dial-up, and propogation delay on satellite transmission. This is the time required to reverse the direction of transmission from send to receive or receive to send on a half-duplex circuit. It can be obtained from the modem specification book and may be referred to as *reclocking time*.

The following TRIB example shows the calculation of throughput assuming a 4,800-bps half-duplex circuit.

$$\text{TRIB} = \frac{7(400 - 10)\,(1 - 0.01)}{(400/600) + 0.025} = 3{,}908 \text{ bits per second}$$

where K = 7 bits per character (information)
M = 400 characters per block
R = 600 characters per second (derived from 4,800 bps divided by 8 bits/character)
C = 10 control characters per block
P = 0.01 (10^{-2}) or 1 retransmission per 100 blocks transmitted—1%
T = 25 milliseconds (0.025) turnaround time

If all factors in the calculation remain constant except for the circuit, which is changed to full duplex (no turnaround time delays, T = 0), then the TRIB increases to 4,504 bps.

Look at the equation where the turnaround value (T) is 0.025. If therre is a further propagation delay time of 475 milliseconds (0.475), this figure changes to 0.500. For demonstrating how a satellite channel affects TRIB, the total delay time is now 500 milliseconds. Still using the figures above (except for the new 0.500 delay time), we reduce the TRIB for our half-duplex satellite link to 2,317 bps, which is almost half of the full-duplex (no turnaround time) 4,054 bps.

FIGURE 4.13 Calculating TRIB (transmission rate of information bits).

provide a few widely used protocols for all networks than to custom tailor protocols to the needs of specific devices or applications. Today, businesses are moving rapidly to reduce the number of different protocols spoken by their networking equipment and converge on a few standard protocols that used widely throughout the network.

We still do use different protocols in different parts of the network where there are important reasons for doing so. For example, local area networks often have different needs than wide area networks, so their data link layer protocols typically are still different, but even here we are seeing a few organizations move to standardize protocols.

This move to standardize data link layer protocols means that networking equipment and networking staff need to understand fewer protocols—their job is becoming simpler, which in turn means that the cost to buy and maintain network equipment and to train networking staff is gradually decreasing (and the side benefit to students is that there are fewer protocols to learn!). The downside, of course, is that some applications may take longer to run over protocols are not perfectly suited to them. As network capacities in the physical layer continue to increase, this has proven to be far less important than the significant cost savings that can be realized from standardization.

SUMMARY

Media Access Control Media access control refers to controlling when computers transmit. There are three basic approaches. With roll-call polling, the server polls client computers to see if they have data to send; computers can transmit only when they have been polled. With hub polling or token passing, the computers themselves manage when they can transmit by passing a token to one other; no computer can transmit unless it has the token. With contention, computers listen and transmit only when no others are transmitting. In general, contention approaches work better for small networks that have low levels of usage, whereas polling approaches work better for networks with high usage.

Sources and Prevention of Error Errors occur in all networks. Errors tend to occur in groups (or bursts) rather than 1 bit at a time. The primary sources of errors are impulse noises (e.g., lightning), cross-talk, echo, and attenuation. Errors can be prevented (or at least reduced) by shielding the cables; moving cables away from sources of noise and power sources; using repeaters (and, to a lesser extent, amplifiers); and improving the quality of the equipment, media, and their connections.

Error Detection and Correction All error-detection schemes attach additional error-detection data, based on a mathematical calculation, to the user's message. The receiver performs the same calculation on incoming messages, and if the results of this calculation do not match the error-detection data on the incoming message, an error has occurred. Parity, LRC, and CRC are the most common error-detection schemes. The most common error-correction technique is simply to ask the sender to retransmit the message until it is received without error. A different approach, forward error correction, includes sufficient information to allow the receiver to correct the error in most cases without asking for a retransmission.

Message Delineation Message delineation means to indicate the start and end of a message. Asynchronous transmission uses start and stop bits on each letter to mark where they begin and end. Synchronous techniques (e.g., SDLC, HDLC, token ring, Ethernet, PPP) or file transfer protocols (e.g., Xmodem, Zmodem) group blocks of data together into packets or frames that use special characters or bit patterns to mark the start and end of entire messages.

Transmission Efficiency and Throughput Every protocol adds additional bits to the user's message before sending it (e.g., for error detection). These bits are called overhead bits because they add no value to the user; they simply ensure correct data transfer. The efficiency of a transmission protocol is the number of information bits sent by the user divided by the total number of bits transferred (information bits plus overhead bits). Synchronous transmission provides greater efficiency than does asynchronous transmission. In general, protocols with larger packet sizes provide greater efficiency than do those with small packet sizes. The drawback to large packet sizes is that they are more likely to be affected by errors and thus require more retransmission. Small packet sizes are therefore better suited to error-prone circuits, and large packets, to error-free circuits.

KEY TERMS

acknowledgement (ACK)
amplifier
asynchronous transmission
attenuation
Automatic Repeat reQuest (ARQ)
block check character (BCC)
burst error
checksum
contention
continuous ARQ
cyclical redundancy check (CRC)
echo
efficiency
error detection with retransmission
error prevention
error rate
Ethernet (IEEE 802.3)
even parity
flow control
forward error correction
frame
Gaussian noise
Go-Back-N ARQ
Hamming code
harmonic distortion
high-level data link control (HDLC)
hub polling
impulse noise
information bits
intermodulation noise
jitter
line noise
line outage
Link Access Protocol-Balanced (LAP-B)
Link Access Protocol for Modems (LAP-M)
media access control
negative acknowledgment (NAK)
odd parity
overhead bits
packet
parity bit
parity checking
Point-to-Point Protocol (PPP)
polling
repeater
roll-call polling
Serial Line Internet Protocol (SLIP)
sliding window
start bit
stop-and-wait ARQ
stop bit
synchronization
synchronous data link control (SDLC)
synchronous transmission
throughput
token passing
token ring (IEEE 802.5)
transmission efficiency
transmission rate of information bits (TRIB)
white noise
Xmodem
Zmodem

QUESTIONS

1. What does the data link layer do?
2. What is media access control, and why is it important?
3. Under what conditions is media access control unimportant?
4. Compare and contrast roll-call polling, hub polling (or token passing), and contention.
5. Which is better, hub polling or contention? Explain.
6. Define two fundamental types of errors.
7. Errors normally appear in _____, which is when more than 1 data bit is changed by the error-causing condition.
8. Is there any difference in the error rates of lower-speed lines and higher-speed lines?
9. Briefly define *noise.*
10. Describe four types of noise. Which is likely to pose the greatest problem to network managers?
11. How do amplifiers differ from repeaters?
12. What are three ways of reducing errors and the types of noise they affect?
13. Describe three approaches to *detecting* errors, including how they work, the probability of detecting an error, and any other benefits or limitations.
14. Briefly describe how even parity and odd parity work.
15. Briefly describe how checksum works.
16. How does CRC work?
17. How does forward error correction work? How is it different from other error-correction methods?
18. Under what circumstances is forward error correction desirable?
19. Compare and contrast stop-and-wait ARQ and continuous ARQ.

20. Which is the simplest (least sophisticated) protocol described in this chapter?

21. How do the various types of Xmodem differ from Zmodem?

22. Describe the packet layouts for SDLC and Ethernet.

23. What is transparency, and why is this a problem with SDLC?

24. How does SDLC overcome transparency problems?

25. Explain why Ethernet does not suffer from transparency problems.

26. Why do SDLC packets need an address?

27. What is transmission efficiency?

28. How do information bits differ from overhead bits?

29. Are stop bits necessary in asynchronous transmission? Explain using a diagram.

30. During the 1990s, there was intense competition between two technologies (10-Mbps Ethernet and 16-Mbps token ring) for the LAN market. Ethernet was promoted by a consortium of vendors, whereas token ring was primarily an IBM product, even though it was standardized. Ethernet won, and no one talks about token ring anymore. Token ring used a hub-polling–based approach. Outline a number of reasons why Ethernet might have won. Hint: The reasons were both technical and business.

31. Under what conditions does a data link layer protocol need an address?

32. Are large packet sizes better than small packet sizes? Expain.

33. What media access control technique does your class use?

34. Show how the word "HI" would be sent using asynchronous transmission using even parity (make assumptions about the bit patterns needed). Show how it would be sent using Ethernet.

EXERCISES

4-1. Draw how a series of four separate messages would be *successfully* sent from one computer to another if the first message was transferred without error, the second was initially transmitted with an error, the third was initially lost, and the ACK for the fourth was initially lost.

4-2. How efficient would a 6-bit code be in asynchronous transmission if it had 1 parity bit, 1 start bit, and 2 stop bits? (Some old equipment uses 2 stop bits.)

4-3. What is the transmission rate of information bits if you use ASCII (8 bits plus 1 parity bit), a 1,000-character block, 56 Kbps modem transmission speed, 20 control characters per block, an error rate of 1 percent, and a 30-millisecond turnaround time? What is the TRIB if you add a half-second delay to the turnaround time because of satellite delay?

4-4. Search the Web to find a software vendor that sells a package that supports each of the following protocols: Zmodem, SDLC, HDLC, Ethernet, and PPP (i.e., one package that supports SDLC, another [or the same] for Zmodem, and so on).

4-5. Investigate the network at your organization (or a service offered by an IXC) to find out the average error rates.

4-6. What is the efficiency if a 100-byte file is transmitted using Ethernet? A 10,000-byte file?

4-7. What is the propagation delay on a circuit using a LEO satellite orbiting 500 miles above the earth if the speed of the signal is 186,000 miles per second? If the satellite is 22,000 miles above the earth?

4-8. Suppose you are going to connect the computers in your house or apartment. What media would you use? Why? Would this change if you were building a new house?

117-147_Fitzg04.qxd 7/5/06 6:26 PM Page 144

MINI-CASES

I. Smith, Smith, Smith, and Smith

Smith, Smith, Smith, and Smith is a regional accounting firm that is putting up a new headquarters building. The building will have a backbone network that connects eight LANs (two on each floor). The company is very concerned with network errors. What advice would you give regarding the design of the building and network cable planning that would help reduce network errors?

II. Worldwide Charity

Worldwide Charity is a charitable organization whose mission is to improve education levels in developing countries. In each country where it is involved, the organization has a small headquarters and usually 5 to 10 offices in outlying towns. Staff members communicate with one another via e-mail on older computers donated to the organization. Because Internet service is not reliable in many of the towns in these countries, the staff members usually phone headquarters and use a very simple Linux e-mail system that uses a server-based network architecture. They also upload and download files. What data link layer protocols should they use for the file transfer? What range of packet sizes is likely to be used?

III. Industrial Products

Industrial Products is a small light-manufacturing firm that produces a variety of control systems for heavy industry. They have a network that connects their office building and warehouse that has functioned well for the last year, but over the past week, users have begun to complain that the network is slow. Clarence Hung, the network manager, did a quick check of number of orders over the past week and saw no real change, suggesting that there has been no major increase in network traffic. What would you suggest that Clarence do next?

IV. Alpha Corp.

Alpha Corp. is trying to decide the size of the connection it needs to the Internet. They estimate that they will send and receive a total of about 1,000 e-mails per hour and that each e-mail message is about 1,500 bytes in size. They also estimate that they will send and receive a total of about 3,000 Web pages per hour and that each page is about 40,000 bytes in size. 1. Without considering transmission efficiency, how large an Internet connection would you recommend in terms of bits per second (assuming that each byte is 8 bits in length)? 2. Assuming they use a synchronous data link layer protocol with an efficiency of about 90%, how large an Internet connection would you recommend? 3. Suppose Alpha wants to be sure that its Internet connection will provide sufficient capacity the next 2 years, how large an Internet connection would you recommend?

CASE STUDY

NEXT-DAY AIR SERVICE

See the Web site.

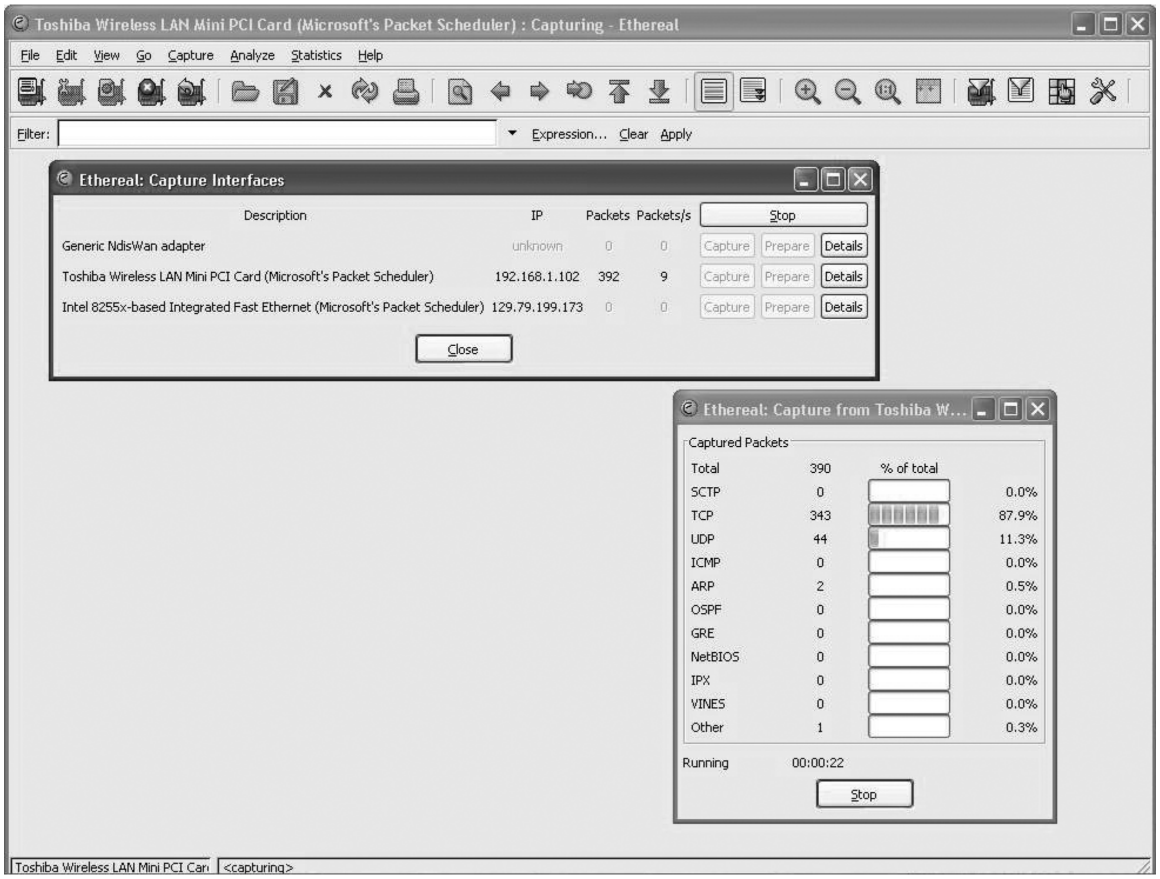
HANDS-ON ACTIVITY

Capturing Packets on Your Network

In this chapter, we discussed several data link layer protocols, such as SDLC and Ethernet. The objective of this Activity is for you to see the data link layer packets in action on your network.

Ethereal is one of the many tools that permit users to examine the packets in their network. It is called a packet sniffer because it enables you to see inside the packets that your computer sends, as well as packets sent by other users on your LAN. In other words, you can eavesdrop on the other users on your LAN to see what Web sites they visit and even the e-mail they send. We don't recommend using it for this reason, but it is important that you understand that someone else could be using Ethereal to sniff your packets to see and record what you are doing on the Internet.

1. Use your browser to connect to www.ethereal.com and download and install the Ethereal software.
2. When you start Ethereal you will see a screen like that in Figure 4.14, minus the two smaller windows on top.
   a. Click Capture
   b. Click Interfaces
   c. Click the Capture button beside your Ethernet connection (wireless LAN or traditional LAN).


Toshiba Wireless LAN Mini PCI Card (Microsoft's Packet Scheduler) : Capturing - Ethereal
File Edit View Go Capture Analyze Statistics Help
Filter:
Expression... Clear Apply
Ethereal: Capture Interfaces
Close
Ethereal: Capture from Toshiba W...
Running 00:00:22
Stop
Toshiba Wireless LAN Mini PCI Card
<capturing>

Description	IP	Packets	Packets/s	Stop		
Generic NdisWan adapter	unknown	0	0	Capture	Prepare	Details
Toshiba Wireless LAN Mini PCI Card (Microsoft's Packet Scheduler)	192.168.1.102	392	9	Capture	Prepare	Details
Intel 8255x-based Integrated Fast Ethernet (Microsoft's Packet Scheduler)	129.79.199.173	0	0	Capture	Prepare	Details

Captured Packets

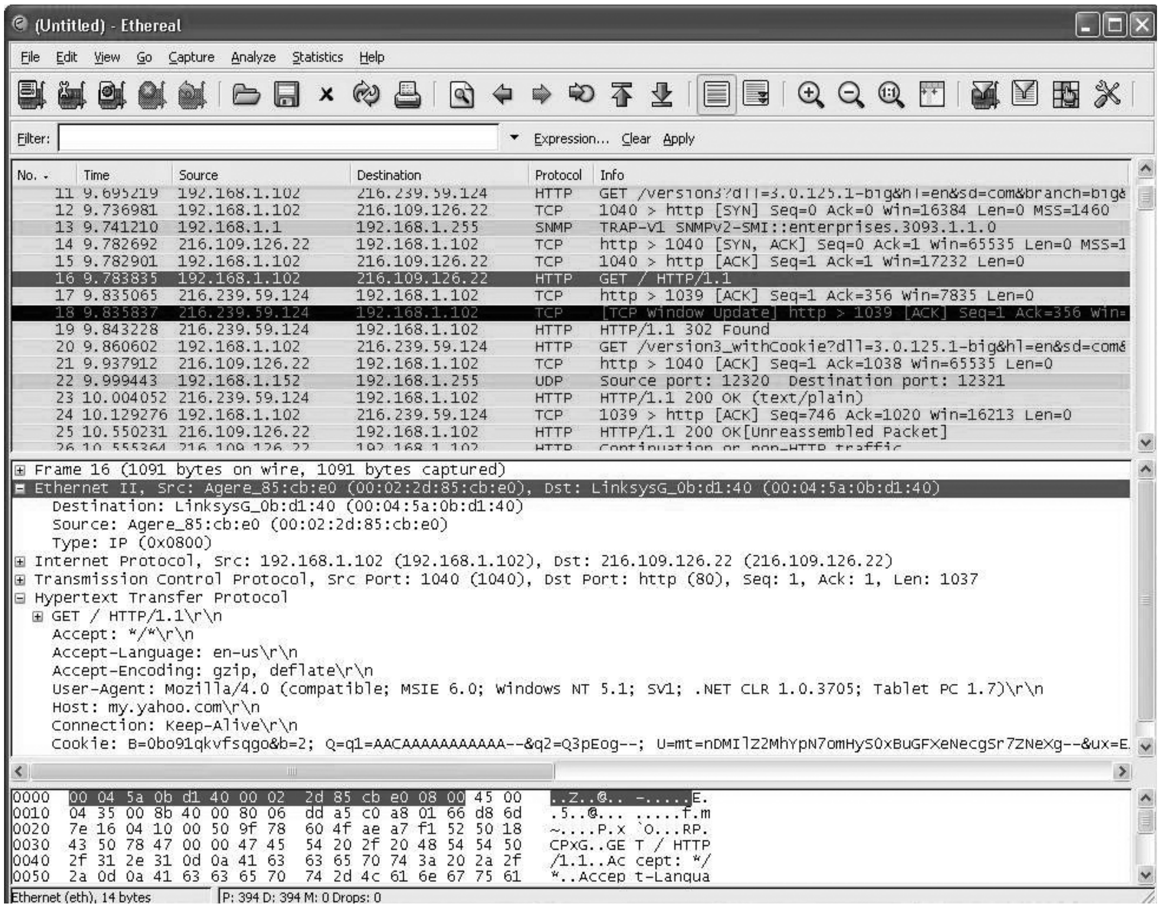
		% of total
Total	390	
SCTP	0	0.0%
TCP	343	87.9%
UDP	44	11.3%
ICMP	0	0.0%
ARP	2	0.5%
OSPF	0	0.0%
GRE	0	0.0%
NetBIOS	0	0.0%
IPX	0	0.0%
VINES	0	0.0%
Other	1	0.3%

FIGURE 4.14 Capturing packets with Ethereal.

3. Ethereal will capture all packets moving through your LAN. To make sure you have something to see, open your Web browser and visit one or two Web sites. After you have captured packets for 30–60 seconds, return to Ethereal and click Stop.
4. Figure 4.15 shows the packets captured on my home network. The top window in Ethereal displays the complete list of packets in chronological order. Each packet is numbered; I've scrolled the window, so the first packet shown is packet 11. Ethereal lists the time, the source IP address, the destination IP address, the protocol, and some additional information about each packet. The IP addresses will be explained in more detail in the next chapter.

For the moment, look at packet number 16, the second HTTP packet from the top. I've clicked on this packet, so the middle window shows the inside of the packet. The first line in this second window says the frame (or packet if you prefer) is 1091 bytes long. It contains an Ethernet II packet, an Internet Protocol (IP) packet, a Transmission Control Protocol (TCP) Packet, and a Hypertext Transfer Protocol (HTTP) packet. Remember in Chapter 1 that Figure 1.4 described how each packet was placed inside another packet as the message moved through the layers and was transmitted.

Click on the plus sign (+) in front of the HTTP packet to expand it. Ethereal shows the contents of


(Untitled) - Ethereal
File Edit View Go Capture Analyze Statistics Help
Filter:
Expression... Clear Apply

```
No. -  Time        Source          Destination     Protocol Info
  11 9.695219    192.168.1.102   216.239.59.124  HTTP     GET /version3?dll=3.0.125.1-big&hl=en&sd=com&branch=big&
  12 9.736981    192.168.1.102   216.109.126.22  TCP      1040 > http [SYN] Seq=0 Ack=0 Win=16384 Len=0 MSS=1460
  13 9.741210    192.168.1.1     192.168.1.255   SNMP     TRAP-V1 SNMPv2-SMI::enterprises.3093.1.1.0
  14 9.782692    216.109.126.22  192.168.1.102   TCP      http > 1040 [SYN, ACK] Seq=0 Ack=1 Win=65535 Len=0 MSS=1
  15 9.782901    192.168.1.102   216.109.126.22  TCP      1040 > http [ACK] Seq=1 Ack=1 Win=17232 Len=0
  16 9.783835    192.168.1.102   216.109.126.22  HTTP     GET / HTTP/1.1
  17 9.835065    216.239.59.124  192.168.1.102   TCP      http > 1039 [ACK] Seq=1 Ack=356 Win=7835 Len=0
  18 9.835837    216.239.59.124  192.168.1.102   TCP      [TCP Window Update] http > 1039 [ACK] Seq=1 Ack=356 Win=
  19 9.843228    216.239.59.124  192.168.1.102   HTTP     HTTP/1.1 302 Found
  20 9.860602    192.168.1.102   216.239.59.124  HTTP     GET /version3_withCookie?dll=3.0.125.1-big&hl=en&sd=com&
  21 9.937912    216.109.126.22  192.168.1.102   TCP      http > 1040 [ACK] Seq=1 Ack=1038 Win=65535 Len=0
  22 9.999443    192.168.1.152   192.168.1.255   UDP      Source port: 12320  Destination port: 12321
  23 10.004052   216.239.59.124  192.168.1.102   HTTP     HTTP/1.1 200 OK (text/plain)
  24 10.129276   192.168.1.102   216.239.59.124  TCP      1039 > http [ACK] Seq=746 Ack=1020 Win=16213 Len=0
  25 10.550231   216.109.126.22  192.168.1.102   HTTP     HTTP/1.1 200 OK[Unreassembled Packet]
  26 10.555364   216.109.126.22  192.168.1.102   HTTP     Continuation or non-HTTP traffic
```

```
⊞ Frame 16 (1091 bytes on wire, 1091 bytes captured)
⊟ Ethernet II, Src: Agere_85:cb:e0 (00:02:2d:85:cb:e0), Dst: LinksysG_0b:d1:40 (00:04:5a:0b:d1:40)
    Destination: LinksysG_0b:d1:40 (00:04:5a:0b:d1:40)
    Source: Agere_85:cb:e0 (00:02:2d:85:cb:e0)
    Type: IP (0x0800)
⊞ Internet Protocol, Src: 192.168.1.102 (192.168.1.102), Dst: 216.109.126.22 (216.109.126.22)
⊞ Transmission Control Protocol, Src Port: 1040 (1040), Dst Port: http (80), Seq: 1, Ack: 1, Len: 1037
⊟ Hypertext Transfer Protocol
  ⊞ GET / HTTP/1.1\r\n
    Accept: */*\r\n
    Accept-Language: en-us\r\n
    Accept-Encoding: gzip, deflate\r\n
    User-Agent: Mozilla/4.0 (compatible; MSIE 6.0; Windows NT 5.1; SV1; .NET CLR 1.0.3705; Tablet PC 1.7)\r\n
    Host: my.yahoo.com\r\n
    Connection: Keep-Alive\r\n
    Cookie: B=0bo91qkvfsqgo&b=2; Q=q1=AACAAAAAAAAAAA--&q2=Q3pEog--; U=mt=nDMIlZ2MhYpN7omHyS0xBuGFXeNecgSr7ZNeXg--&ux=E.
```

```
0000  00 04 5a 0b d1 40 00 02  2d 85 cb e0 08 00 45 00   ..Z..@.. -.....E.
0010  04 35 00 8b 40 00 80 06  dd a5 c0 a8 01 66 d8 6d   .5..@... .....f.m
0020  7e 16 04 10 00 50 9f 78  60 4f ae a7 f1 52 50 18   ~....P.x `O...RP.
0030  43 50 78 47 00 00 47 45  54 20 2f 20 48 54 54 50   CPxG..GE T / HTTP
0040  2f 31 2e 31 0d 0a 41 63  63 65 70 74 3a 20 2a 2f   /1.1..Ac cept: */
0050  2a 0d 0a 41 63 63 65 70  74 2d 4c 61 6e 67 75 61   *..Accep t-Langua
```

```
Ethernet (eth), 14 bytes    P: 394 D: 394 M: 0 Drops: 0
```

FIGURE 4.15 Analyzing packets with Ethereal.

the HTTP packet. By reading the data inside the HTTP packet, you can see that this packet was an HTTP request to my.yahoo.com that contained a cookie. If you look closely, you'll see that the sending computer was a Tablet PC—that's some of the optional information my Web browser (Internet Explorer) included in the HTTP header.

The bottom window in Figure 4.15 shows the exact bytes that were captured. The section highlighted in grey shows the HTTP packet. The numbers on the left show the data in hexadecimal format while the data on the right show the text version. The data before the highlighted section is the TCP packet.

From Chapter 2, you know that the client sends an HTTP request packet to request a Web page, and the Web server sends back an HTTP response packet. Packet number 25 in the top window in Figure 4.15 is the HTTP response sent back to my computer by the Yahoo server. You can see that the destination IP address in my HTTP request is the source IP address of this HTTP packet.

5. Figure 4.15 also shows what happen when you click the plus sign (+) in front of the Ethernet II packet to expand it. You can see that this Ethernet packet has a destination address and source address (e.g., 00:02:2d:85:cb:e0).