

Data Link Layer

Objectives

The data link layer transforms the physical layer, a raw transmission facility, to a link responsible for node-to-node (hop-to-hop) communication. Specific responsibilities of the data link layer include *framing*, *addressing*, *flow control*, *error control*, and *media access control*. The data link layer divides the stream of bits received from the network layer into manageable data units called frames. The data link layer adds a header to the frame to define the addresses of the sender and receiver of the frame. If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender, the data link layer imposes a flow control mechanism to avoid overwhelming the receiver. The data link layer also adds reliability to the physical layer by adding mechanisms to detect and retransmit damaged, duplicate, or lost frames. When two or more devices are connected to the same link, data link layer protocols are necessary to determine which device has control over the link at any given time.

In Part 3 of the book, we first discuss services provided by the data link layer. We then discuss the implementation of these services in local area networks (LANs). Finally we discuss how wide area networks (WANs) use these services.

Part 3 of the book is devoted to the data link layer and the services provided by this layer.

Chapters

This part consists of nine chapters: Chapters 10 to 18.

Chapter 10

Chapter 10 discusses error detection and correction. Although the quality of devices and media have been improved during the last decade, we still need to check for errors and correct them in most applications.

Chapter 11

Chapter 11 is named data link control, which involves flow and error control. It discusses some protocols that are designed to handle the services required from the data link layer in relation to the network layer.

Chapter 12

Chapter 12 is devoted to access control, the duties of the data link layer that are related to the use of the physical layer.

Chapter 13

This chapter introduces wired local area networks. A wired LAN, viewed as a link, is mostly involved in the physical and data link layers. We have devoted the chapter to the discussion of Ethernet and its evolution, a dominant technology today.

Chapter 14

This chapter introduces wireless local area networks. The wireless LAN is a growing technology in the Internet. We devote one chapter to this topic.

Chapter 15

After discussing wired and wireless LANs, we show how they can be connected together using connecting devices.

Chapter 16

This is the first chapter on wide area networks (WANs). We start with wireless WANs and then move on to satellite networks and mobile telephone networks.

Chapter 17

To demonstrate the operation of a high-speed wide area network that can be used as a backbone for other WANs or for the Internet, we have chosen to devote all of Chapter 17 to SONET, a wide area network that uses fiber-optic technology.

Chapter 18

This chapter concludes our discussion on wide area networks. Two switched WANs, Frame Relay and ATM, are discussed here.

CHAPTER 10

Error Detection and Correction

Networks must be able to transfer data from one device to another with acceptable accuracy. For most applications, a system must guarantee that the data received are identical to the data transmitted. Any time data are transmitted from one node to the next, they can become corrupted in passage. Many factors can alter one or more bits of a message. Some applications require a mechanism for detecting and correcting errors.

Data can be corrupted during transmission.
Some applications require that errors be detected and corrected.

Some applications can tolerate a small level of error. For example, random errors in audio or video transmissions may be tolerable, but when we transfer text, we expect a very high level of accuracy.

10.1 INTRODUCTION

Let us first discuss some issues related, directly or indirectly, to error detection and correction.

Types of Errors

Whenever bits flow from one point to another, they are subject to unpredictable changes because of interference. This interference can change the shape of the signal. In a single-bit error, a 0 is changed to a 1 or a 1 to a 0. In a burst error, multiple bits are changed. For example, a 11100 s burst of impulse noise on a transmission with a data rate of 1200 bps might change all or some of the 12 bits of information.

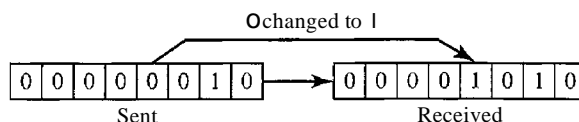
Single-Bit Error

The term *single-bit error* means that only 1 bit of a given data unit (such as a byte, character, or packet) is changed from 1 to 0 or from 0 to 1.

In a single-bit error, only 1 bit in the data unit has changed.

Figure 10.1 shows the effect of a single-bit error on a data unit. To understand the impact of the change, imagine that each group of 8 bits is an ASCII character with a 0 bit added to the left. In Figure 10.1, 00000010 (ASCII *STX*) was sent, meaning *start of text*, but 00001010 (ASCII *LF*) was received, meaning *line feed*. (For more information about ASCII code, see Appendix A.)

:Figure 10.1 *Single-bit error*



Single-bit errors are the least likely type of error in serial data transmission. To understand why, imagine data sent at 1 Mbps. This means that each bit lasts only $1/1,000,000$ s, or $1\ \mu\text{s}$. For a single-bit error to occur, the noise must have a duration of only $1\ \mu\text{s}$, which is very rare; noise normally lasts much longer than this.

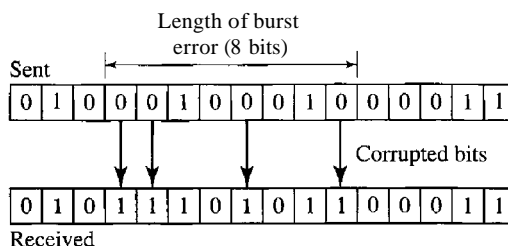
Burst Error

The term *burst error* means that 2 or more bits in the data unit have changed from 1 to 0 or from 0 to 1.

A burst error means that 2 or more bits in the data unit have changed.

Figure 10.2 shows the effect of a burst error on a data unit. In this case, 0100010001000011 was sent, but 0101110101100011 was received. Note that a burst error does not necessarily mean that the errors occur in consecutive bits. The length of the burst is measured from the first corrupted bit to the last corrupted bit. Some bits in between may not have been corrupted.

Figure 10.2 *Burst error of length 8*



A burst error is more likely to occur than a single-bit error. The duration of noise is normally longer than the duration of 1 bit, which means that when noise affects data, it affects a set of bits. The number of bits affected depends on the data rate and duration of noise. For example, if we are sending data at 1 kbps, a noise of 11100 s can affect 10 bits; if we are sending data at 1 Mbps, the same noise can affect 10,000 bits.

Redundancy

The central concept in detecting or correcting errors is redundancy. To be able to detect or correct errors, we need to send some extra bits with our data. These redundant bits are added by the sender and removed by the receiver. Their presence allows the receiver to detect or correct corrupted bits.

To detect or correct errors, we need to send **extra** (redundant) bits with data.

Detection Versus Correction

The correction of errors is more difficult than the detection. In error detection, we are looking only to see if any error has occurred. The answer is a simple yes or no. We are not even interested in the number of errors. A single-bit error is the same for us as a burst error.

In error correction, we need to know the exact number of bits that are corrupted and more importantly, their location in the message. The number of the errors and the size of the message are important factors. If we need to correct one single error in an 8-bit data unit, we need to consider eight possible error locations; if we need to correct two errors in a data unit of the same size, we need to consider 28 possibilities. You can imagine the receiver's difficulty in finding 10 errors in a data unit of 1000 bits.

Forward Error Correction Versus Retransmission

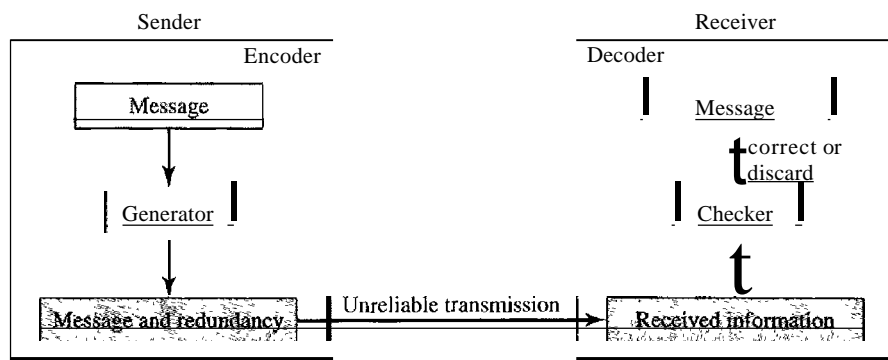
There are two main methods of error correction. Forward error correction is the process in which the receiver tries to guess the message by using redundant bits. This is possible, as we see later, if the number of errors is small. Correction by retransmission is a technique in which the receiver detects the occurrence of an error and asks the sender to resend the message. Resending is repeated until a message arrives that the receiver believes is error-free (usually, not all errors can be detected).

Coding

Redundancy is achieved through various coding schemes. The sender adds redundant bits through a process that creates a relationship between the redundant bits and the actual data bits. The receiver checks the relationships between the two sets of bits to detect or correct the errors. The ratio of redundant bits to the data bits and the robustness of the process are important factors in any coding scheme. Figure 10.3 shows the general idea of coding.

We can divide coding schemes into two broad categories: block coding and convolution coding. In this book, we concentrate on block coding; convolution coding is more complex and beyond the scope of this book.

Figure 10.3 The structure of encoder and decoder



In this book, we concentrate on block codes; we leave convolution codes to advanced texts.

Modular Arithmetic

Before we finish this section, let us briefly discuss a concept basic to computer science in general and to error detection and correction in particular: modular arithmetic. Our intent here is not to delve deeply into the mathematics of this topic; we present just enough information to provide a background to materials discussed in this chapter.

In modular arithmetic, we use only a limited range of integers. We define an upper limit, called a modulus N . We then use only the integers 0 to $N - 1$, inclusive. This is *modulo- N* arithmetic. For example, if the modulus is 12, we use only the integers 0 to 11, inclusive. An example of modulo arithmetic is our clock system. It is based on modulo-12 arithmetic, substituting the number 12 for 0. In a *modulo- N* system, if a number is greater than N , it is divided by N and the remainder is the result. If it is negative, as many N s as needed are added to make it positive. Consider our clock system again. If we start a job at 11 A.M. and the job takes 5 h, we can say that the job is to be finished at 16:00 if we are in the military, or we can say that it will be finished at 4 P.M. (the remainder of $16/12$ is 4).

In modulo- N arithmetic, we use only the integers in the range 0 to $N - 1$, inclusive.

Addition and subtraction in modulo arithmetic are simple. There is no carry when you add two digits in a column. There is no carry when you subtract one digit from another in a column.

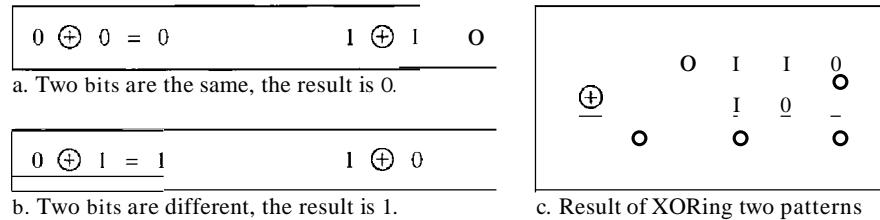
Modulo-2 Arithmetic

Of particular interest is modulo-2 arithmetic. In this arithmetic, the modulus N is 2. We can use only 0 and 1. Operations in this arithmetic are very simple. The following shows how we can add or subtract 2 bits.

Adding:	$0+0=0$	$0+1=1$	$1+0=1$	$1+1=0$
Subtracting:	$0-0=0$	$0-1=1$	$1-0=1$	$1-1=0$

Notice particularly that addition and subtraction give the same results. In this arithmetic we use the XOR (exclusive OR) operation for both addition and subtraction. The result of an XOR operation is 0 if two bits are the same; the result is 1 if two bits are different. Figure 10.4 shows this operation.

Figure 10.4 *XORing of two single bits or two words*



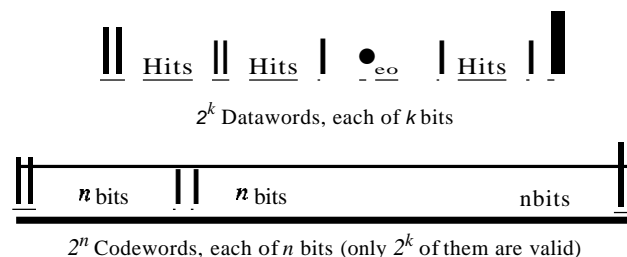
Other Modulo Arithmetic

We also use, modulo- N arithmetic through the book. The principle is the same; we use numbers between 0 and $N - 1$. If the modulus is not 2, addition and subtraction are distinct. If we get a negative result, we add enough multiples of N to make it positive.

10.2 BLOCK CODING

In block coding, we divide our message into blocks, each of k bits, called datawords. We add r redundant bits to each block to make the length $n = k + r$. The resulting n -bit blocks are called codewords. How the extra r bits is chosen or calculated is something we will discuss later. For the moment, it is important to know that we have a set of datawords, each of size k , and a set of codewords, each of size of n . With k bits, we can create a combination of 2^k datawords; with n bits, we can create a combination of 2^n codewords. Since $n > k$, the number of possible codewords is larger than the number of possible datawords. The block coding process is one-to-one; the same dataword is always encoded as the same codeword. This means that we have $2^n - 2^k$ codewords that are not used. We call these codewords invalid or illegal. Figure 10.5 shows the situation.

Figure 10.5 *Datawords and codewords in block coding*



Example 10.1

The 4B/5B block coding discussed in Chapter 4 is a good example of this type of coding. In this coding scheme, $k=4$ and $n=5$. As we saw, we have $2^k=16$ datawords and $2^n=32$ codewords. We saw that 16 out of 32 codewords are used for message transfer and the rest are either used for other purposes or unused.

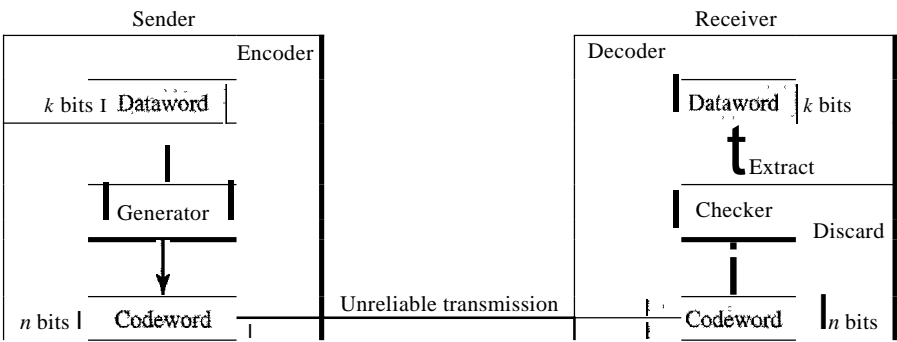
Error Detection

How can errors be detected by using block coding? If the following two conditions are met, the receiver can detect a change in the original codeword.

- 1. The receiver has (or can find) a list of valid codewords.
- 2. The original codeword has changed to an invalid one.

Figure 10.6 shows the role of block coding in error detection.

Figure 10.6 *Process of error detection in block coding*



The sender creates codewords out of datawords by using a generator that applies the rules and procedures of encoding (discussed later). Each codeword sent to the receiver may change during transmission. If the received codeword is the same as one of the valid codewords, the word is accepted; the corresponding dataword is extracted for use. If the received codeword is not valid, it is discarded. However, if the codeword is corrupted during transmission but the received word still matches a valid codeword, the error remains undetected. This type of coding can detect only single errors. Two or more errors may remain undetected.

Example 10.2

Let us assume that $k=2$ and $n=3$. Table 10.1 shows the list of datawords and codewords. Later, we will see how to derive a codeword from a dataword.

Table 10.1 *A code for error detection (Example 10.2)*

<i>Datawords</i>	<i>Codewords</i>
00	000
01	011
10	101
11	110

Assume the sender encodes the dataword 01 as 011 and sends it to the receiver. Consider the following cases:

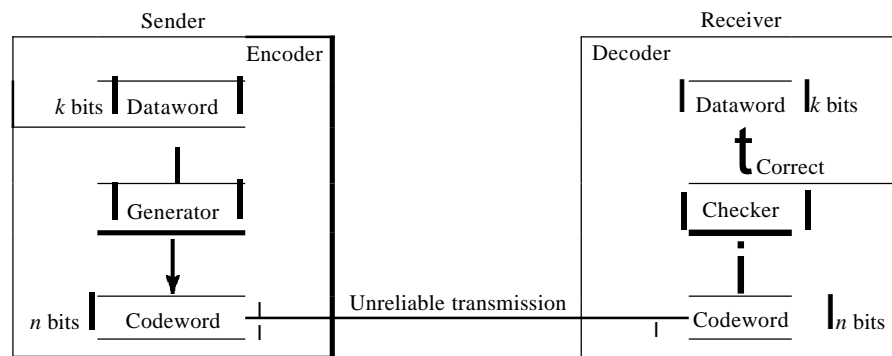
1. The receiver receives 011. It is a valid codeword. The receiver extracts the dataword 01 from it.
2. The codeword is corrupted during transmission, and 111 is received (the leftmost bit is corrupted). This is not a valid codeword and is discarded.
3. The codeword is corrupted during transmission, and 000 is received (the right two bits are corrupted). This is a valid codeword. The receiver incorrectly extracts the dataword 00. Two corrupted bits have made the error undetectable.

An error-detecting code can detect only the types of errors for which it is designed; other types of errors may remain undetected.

Error Correction

As we said before, error correction is much more difficult than error detection. In error detection, the receiver needs to know only that the received codeword is invalid; in error correction the receiver needs to find (or guess) the original codeword sent. We can say that we need more redundant bits for error correction than for error detection. Figure 10.7 shows the role of block coding in error correction. We can see that the idea is the same as error detection but the checker functions are much more complex.

Figure 10.7 Structure of encoder and decoder in error correction



Example 10.3

Let us add more redundant bits to Example 10.2 to see if the receiver can correct an error without knowing what was actually sent. We add 3 redundant bits to the 2-bit dataword to make 5-bit codewords. Again, later we will show how we chose the redundant bits. For the moment let us concentrate on the error correction concept. Table 10.2 shows the datawords and codewords.

Assume the dataword is 01. The sender consults the table (or uses an algorithm) to create the codeword 01011. The codeword is corrupted during transmission, and 01001 is received (error in the second bit from the right). First, the receiver finds that the received codeword is not in the table. This means an error has occurred. (Detection must come before correction.) The receiver, assuming that there is only 1 bit corrupted, uses the following strategy to guess the correct dataword.

Table 10.2 A code for error correction (Example 10.3)

<i>Dataword</i>	<i>Codeword</i>
00	00000
01	01011
10	10101
11	11110

1. Comparing the received codeword with the first codeword in the table (01001 versus 00000), the receiver decides that the first codeword is not the one that was sent because there are two different bits.
2. By the same reasoning, the original codeword cannot be the third or fourth one in the table.
3. The original codeword must be the second one in the table because this is the only one that differs from the received codeword by 1 bit. The receiver replaces 01001 with 01011 and consults the table to find the dataword 01.

Hamming Distance

One of the central concepts in coding for error control is the idea of the Hamming distance. The Hamming distance between two words (of the same size) is the number of differences between the corresponding bits. We show the Hamming distance between two words x and y as $d(x, y)$.

The Hamming distance can easily be found if we apply the XOR operation (\oplus) on the two words and count the number of 1s in the result. Note that the Hamming distance is a value greater than zero.

The Hamming distance between two words is the number
of differences between corresponding bits.

Example 10.4

Let us find the Hamming distance between two pairs of words.

1. The Hamming distance $d(000, 011)$ is 2 because $000 \oplus 011$ is 011 (two 1s).
2. The Hamming distance $d(10101, 11110)$ is 3 because $10101 \oplus 11110$ is 01011 (three 1s).

Minimum Hamming Distance

Although the concept of the Hamming distance is the central point in dealing with error detection and correction codes, the measurement that is used for designing a code is the minimum Hamming distance. In a set of words, the minimum Hamming distance is the smallest Hamming distance between all possible pairs. We use d_{\min} to define the minimum Hamming distance in a coding scheme. To find this value, we find the Hamming distances between all words and select the smallest one.

The minimum Hamming distance is the smallest Hamming
distance between all possible pairs in a set of words.

Example 10.5

Find the minimum Hamming distance of the coding scheme in Table 10.1.

Solution

We first find all Hamming distances.

$$\begin{array}{llll} d(000, 011) = 2 & d(000, 101) = 2 & d(0a0, 110) = 2 & d(011, 101) = 2 \\ d(011, 110) = 2 & d(W1, 110) = 2 & & \end{array}$$

The d_{min} in this case is 2.

Example 10.6

Find the minimum Hamming distance of the coding scheme in Table 10.2.

Solution

We first find all the Hamming distances.

$$\begin{array}{lll} d(00000, 01011) = 3 & d(00000, 10101) = 3 & d(00000, 11110) = 4 \\ d(01011, 10101) = 4 & d(01011, 11110) = 3 & d(10101, 11110) = 3 \end{array}$$

The d_{min} in this case is 3.

Three Parameters

Before we continue with our discussion, we need to mention that any coding scheme needs to have at least three parameters: the codeword size n , the dataword size k , and the minimum Hamming distance d_{min} . A coding scheme C is written as $C(n, k)$ with a separate expression for d_{min} . For example, we can call our first coding scheme $C(3, 2)$ with $d_{min} = 2$ and our second coding scheme $C(5, 2)$ with $d_{min} = 3$.

Hamming Distance and Error

Before we explore the criteria for error detection or correction, let us discuss the relationship between the Hamming distance and errors occurring during transmission. When a codeword is corrupted during transmission, the Hamming distance between the sent and received codewords is the number of bits affected by the error. In other words, the Hamming distance between the received codeword and the sent codeword is the number of bits that are corrupted during transmission. For example, if the codeword 00000 is sent and 01101 is received, 3 bits are in error and the Hamming distance between the two is $d(00000, 01101) = 3$.

Minimum Distance for Error Detection

Now let us find the minimum Hamming distance in a code if we want to be able to detect up to s errors. If s errors occur during transmission, the Hamming distance between the sent codeword and received codeword is s . If our code is to detect up to s errors, the minimum distance between the valid codes must be $s + 1$, so that the received codeword does not match a valid codeword. In other words, if the minimum distance between all valid codewords is $s + 1$, the received codeword cannot be erroneously mistaken for another codeword. The distances are not enough ($s + 1$) for the receiver to accept it as valid. The error will be detected. We need to clarify a point here: Although a code with $d_{min} = s + 1$

may be able to detect more than s errors in some special cases, only s or fewer errors are guaranteed to be detected.

To guarantee the detection of up to s errors in all cases, the minimum Hamming distance in a block code must be $d_{\min} = s + 1$.

Example 10.7

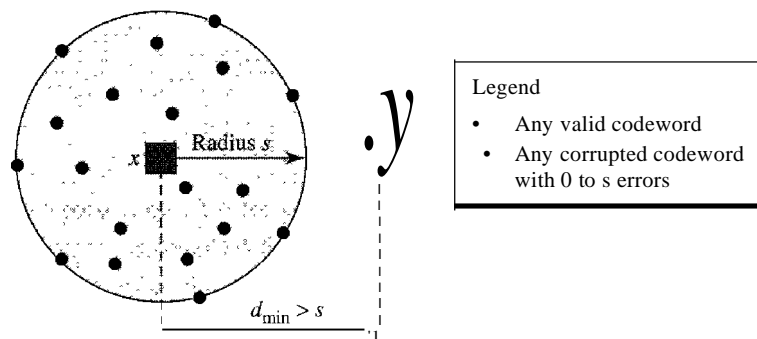
The minimum Hamming distance for our first code scheme (Table 10.1) is 2. This code guarantees detection of only a single error. For example, if the third codeword (101) is sent and one error occurs, the received codeword does not match any valid codeword. If two errors occur, however, the received codeword may match a valid codeword and the errors are not detected.

Example 10.8

Our second block code scheme (Table 10.2) has $d_{\min} = 3$. This code can detect up to two errors. Again, we see that when any of the valid codewords is sent, two errors create a codeword which is not in the table of valid codewords. The receiver cannot be fooled. However, some combinations of three errors change a valid codeword to another valid codeword. The receiver accepts the received codeword and the errors are undetected.

We can look at this geometrically. Let us assume that the sent codeword x is at the center of a circle with radius s . All other received codewords that are created by 1 to s errors are points inside the circle or on the perimeter of the circle. All other valid codewords must be outside the circle, as shown in Figure 10.8.

Figure 10.8 Geometric concept for finding d_{\min} in error detection



In Figure 10.8, d_{\min} must be an integer greater than s ; that is, $d_{\min} = s + 1$.

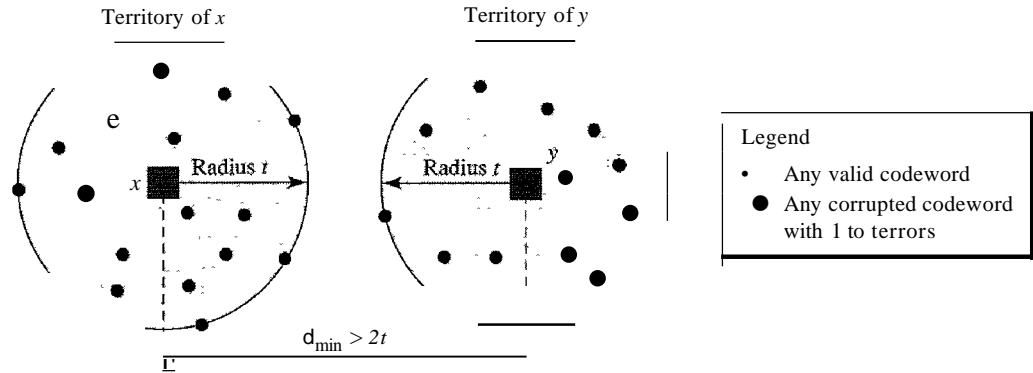
Minimum Distance for Error Correction

Error correction is more complex than error detection; a decision is involved. When a received codeword is not a valid codeword, the receiver needs to decide which valid codeword was actually sent. The decision is based on the concept of territory, an exclusive area surrounding the codeword. Each valid codeword has its own territory.

We use a geometric approach to define each territory. We assume that each valid codeword has a circular territory with a radius of t and that the valid codeword is at the

center. For example, suppose a codeword x is corrupted by t bits or less. Then this corrupted codeword is located either inside or on the perimeter of this circle. If the receiver receives a codeword that belongs to this territory, it decides that the original codeword is the one at the center. Note that we assume that only up to t errors have occurred; otherwise, the decision is wrong. Figure 10.9 shows this geometric interpretation. Some texts use a sphere to show the distance between all valid block codes.

Figure 10.9 Geometric concept for finding d_{\min} in error correction



In Figure 10.9, $d_{\min} > 2t$; since the next integer increment is 1, we can say that $d_{\min} = 2t + 1$.

To guarantee correction of up to t errors in all cases, the minimum Hamming distance in a block code must be $d_{\min} = 2t + 1$.

Example 10.9

A code scheme has a Hamming distance $d_{\min} = 4$. What is the error detection and correction capability of this scheme?

Solution

This code guarantees the detection of up to three errors ($s = 3$), but it can correct up to one error. In other words, if this code is used for error correction, part of its capability is wasted. Error correction codes need to have an odd minimum distance (3, 5, 7, ...).

10.3 LINEAR BLOCK CODES

Almost all block codes used today belong to a subset called linear block codes. The use of nonlinear block codes for error detection and correction is not as widespread because their structure makes theoretical analysis and implementation difficult. We therefore concentrate on linear block codes.

The formal definition of linear block codes requires the knowledge of abstract algebra (particularly Galois fields), which is beyond the scope of this book. We therefore give an informal definition. For our purposes, a linear block code is a code in which the exclusive OR (addition modulo-2) of two valid codewords creates another valid codeword.

In a linear block code, the exclusive OR (XOR) of any two valid codewords creates another valid codeword.

Example 10.10

Let us see if the two codes we defined in Table 10.1 and Table 10.2 belong to the class of linear block codes.

1. The scheme in Table 10.1 is a linear block code because the result of XORing any codeword with any other codeword is a valid codeword. For example, the XORing of the second and third codewords creates the fourth one.
2. The scheme in Table 10.2 is also a linear block code. We can create all four codewords by XORing two other codewords.

Minimum Distance for Linear Block Codes

It is simple to find the minimum Hamming distance for a linear block code. The minimum Hamming distance is the number of 1s in the nonzero valid codeword with the smallest number of 1s.

Example 10.11

In our first code (Table 10.1), the numbers of 1s in the nonzero codewords are 2, 2, and 2. So the minimum Hamming distance is $d_{\min} = 2$. In our second code (Table 10.2), the numbers of 1s in the nonzero codewords are 3, 3, and 4. So in this code we have $d_{\min} = 3$.

Some Linear Block Codes

Let us now show some linear block codes. These codes are trivial because we can easily find the encoding and decoding algorithms and check their performances.

Simple Parity-Check Code

Perhaps the most familiar error-detecting code is the simple parity-check code. In this code, a k -bit dataword is changed to an n -bit codeword where $n = k + 1$. The extra bit, called the parity bit, is selected to make the total number of 1s in the codeword even. Although some implementations specify an odd number of 1s, we discuss the even case. The minimum Hamming distance for this category is $d_{\min} = 2$, which means that the code is a single-bit error-detecting code; it cannot correct any error.

A simple parity-check code is a single-bit error-detecting code in which $n = k + 1$ with $d_{\min} = 2$.

Our first code (Table 10.1) is a parity-check code with $k = 2$ and $n = 3$. The code in Table 10.3 is also a parity-check code with $k = 4$ and $n = 5$.

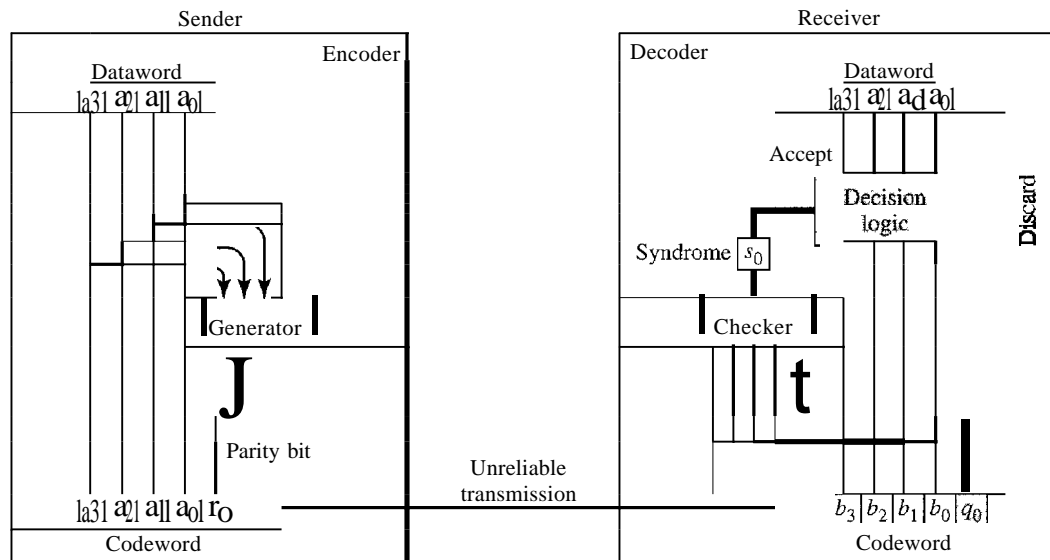
Figure 10.10 shows a possible structure of an encoder (at the sender) and a decoder (at the receiver).

The encoder uses a generator that takes a copy of a 4-bit dataword (a_0, a_1, a_2 , and a_3) and generates a parity bit p_0 . The dataword bits and the parity bit create the 5-bit codeword. The parity bit that is added makes the number of 1s in the codeword even.

Table 10.3 Simple parity-check code $C(5, 4)$

Datawords	Codewords	Datawords	Codewords
0000	00000	1000	10001
0001	00011	1001	10010
0010	00101	1010	10100
0011	00110	1011	10111
0100	01001	1100	11000
0101	01010	1101	11011
0110	01100	1110	11101
0111	01111	1111	11110

Figure 10.10 Encoder and decoder for simple parity-check code



This is normally done by adding the 4 bits of the dataword (modulo-2); the result is the parity bit. In other words,

$$r_0 = a_3 + a_2 + a_1 + a_0 \quad (\text{modulo-2})$$

If the number of 1s is even, the result is 0; if the number of 1s is odd, the result is 1. In both cases, the total number of 1s in the codeword is even.

The sender sends the codeword which may be corrupted during transmission. The receiver receives a 5-bit word. The checker at the receiver does the same thing as the generator in the sender with one exception: The addition is done over all 5 bits. The result, which is called the syndrome, is just 1 bit. The syndrome is 0 when the number of 1s in the received codeword is even; otherwise, it is 1.

$$s_0 = b_3 + b_2 + b_1 + b_0 + q_0 \quad (\text{modulo-2})$$

The syndrome is passed to the decision logic analyzer. If the syndrome is 0, there is no error in the received codeword; the data portion of the received codeword is accepted as the dataword; if the syndrome is 1, the data portion of the received codeword is discarded. The dataword is not created.

Example 10.12

Let us look at some transmission scenarios. Assume the sender sends the dataword 1011. The codeword created from this dataword is 10111, which is sent to the receiver. We examine five cases:

1. No error occurs; the received codeword is 10111. The syndrome is 0. The dataword 1011 is created.
2. One single-bit error changes $a1'$. The received codeword is 10011. The syndrome is 1. No dataword is created.
3. One single-bit error changes 100 . The received codeword is 10110. The syndrome is 1. No dataword is created. Note that although none of the dataword bits are corrupted, no dataword is created because the code is not sophisticated enough to show the position of the corrupted bit.
4. An error changes 10 and a second error changes $a3'$. The received codeword is 00110. The syndrome is 0. The dataword 0011 is created at the receiver. Note that here the dataword is wrongly created due to the syndrome value. The simple parity-check decoder cannot detect an even number of errors. The errors cancel each other out and give the syndrome a value of 0.
5. Three bits— $a3$, $a2$, and $a1$ —are changed by errors. The received codeword is 01011. The syndrome is 1. The dataword is not created. This shows that the simple parity check, guaranteed to detect one single error, can also find any odd number of errors.

A simple parity-check code can detect an odd number of errors.

A better approach is the two-dimensional parity check. In this method, the dataword is organized in a table (rows and columns). In Figure 10.11, the data to be sent, five 7-bit bytes, are put in separate rows. For each row and each column, 1 parity-check bit is calculated. The whole table is then sent to the receiver, which finds the syndrome for each row and each column. As Figure 10.11 shows, the two-dimensional parity check can detect up to three errors that occur anywhere in the table (arrows point to the locations of the created nonzero syndromes). However, errors affecting 4 bits may not be detected.

Hamming Codes

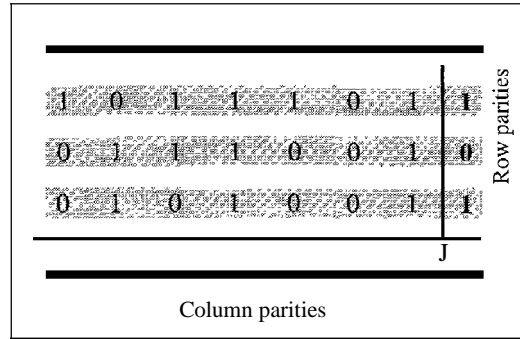
Now let us discuss a category of error-correcting codes called Hamming codes. These codes were originally designed with $d_{\min} = 3$, which means that they can detect up to two errors or correct one single error. Although there are some Hamming codes that can correct more than one error, our discussion focuses on the single-bit error-correcting code.

First let us find the relationship between n and k in a Hamming code. We need to choose an integer $m \geq 3$. The values of n and k are then calculated from $n = 2^m - 1$ and $k = n - m$. The number of check bits $r = m$.

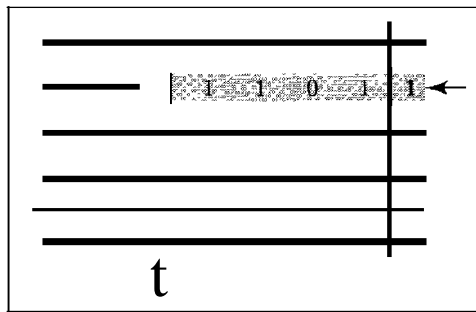
All Hamming codes discussed in this book have $d_{\min} = 3$.
The relationship between m and n in these codes is $n = 2^m - 1$.

For example, if $m = 3$, then $n = 7$ and $k = 4$. This is a Hamming code $C(7, 4)$ with $d_{\min} = 3$. Table 10.4 shows the datawords and codewords for this code.

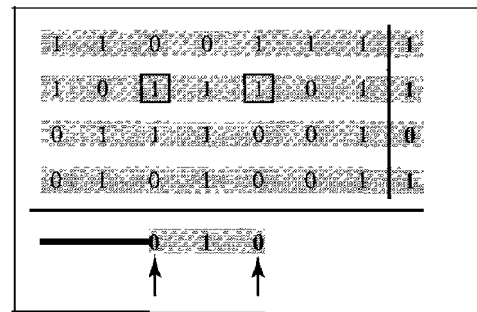
Figure 10.11 Two-dimensional parity-check code



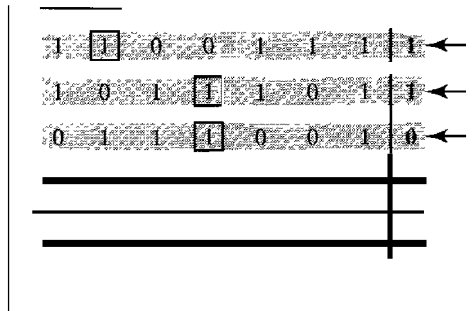
a. Design of row and column parities



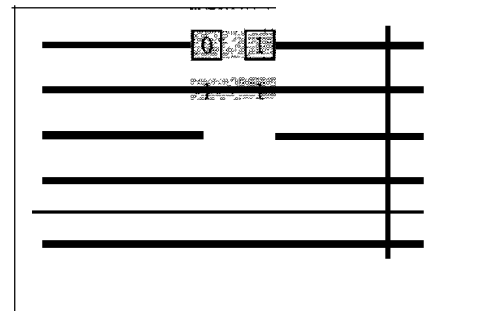
b. One error affects two parities



c. Two errors affect two parities



d. Three errors affect four parities



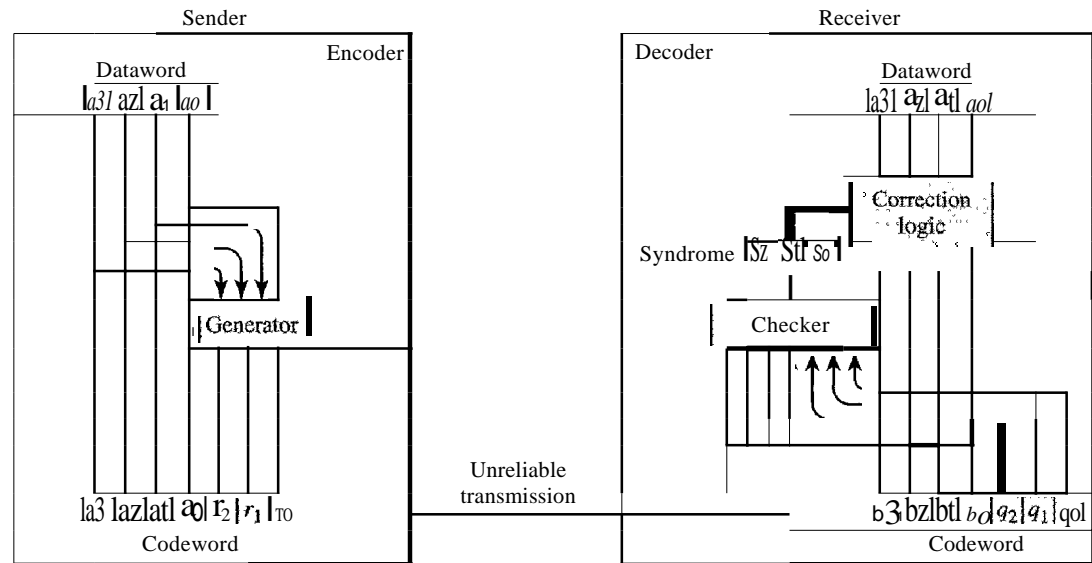
e. Four errors cannot be detected

Table 10.4 Hamming code $C(7, 4)$

Datawords	Codewords	Datawords	Codewords
0000	0000000	1000	1000110
0001	0001101	1001	1001011
0010	0010111	1010	1010001
0011	0011010	1011	1011100
0100	0100011	1100	1100101
0101	0101110	1101	1101000
0110	0110100	1110	1110010
0111	0111001	1111	1111111

Figure 10.12 shows the structure of the encoder and decoder for this example.

Figure 10.12 The structure of the encoder and decoder for a Hamming code



A copy of a 4-bit dataword is fed into the generator that creates three parity checks r_0 , r_1 and r_2 as shown below:

$$r_0 = a_2 + a_1 + a_0 \quad \text{modulo-2}$$

$$r_1 = a_3 + a_2 + a_1 \quad \text{modulo-2}$$

$$r_2 = a_1 + a_0 + a_3 \quad \text{modulo-2}$$

In other words, each of the parity-check bits handles 3 out of the 4 bits of the dataword. The total number of 1s in each 4-bit combination (3 dataword bits and 1 parity bit) must be even. We are not saying that these three equations are unique; any three equations that involve 3 of the 4 bits in the dataword and create independent equations (a combination of two cannot create the third) are valid.

The checker in the decoder creates a 3-bit syndrome ($s_2s_1s_0$) in which each bit is the parity check for 4 out of the 7 bits in the received codeword:

$$s_0 = b_2 + b_1 + b_0 + q_0 \quad \text{modulo-2}$$

$$s_1 = b_3 + b_2 + b_1 + q_1 \quad \text{modulo-2}$$

$$s_2 = b_1 + b_0 + b_3 + q_2 \quad \text{modulo-2}$$

The equations used by the checker are the same as those used by the generator with the parity-check bits added to the right-hand side of the equation. The 3-bit syndrome creates eight different bit patterns (000 to 111) that can represent eight different conditions. These conditions define a lack of error or an error in 1 of the 7 bits of the received codeword, as shown in Table 10.5.

Table 10.5 Logical decision made by the correction logic analyzer at the decoder

Syndrome	000	001	010	011	100	101	110	111
Error	None	q_0	q_1	b_2	q_2	b_0	b_3	b_1

Note that the generator is not concerned with the four cases shaded in Table 10.5 because there is either no error or an error in the parity bit. In the other four cases, 1 of the bits must be flipped (changed from 0 to 1 or 1 to 0) to find the correct dataword.

The syndrome values in Table 10.5 are based on the syndrome bit calculations. For example, if q_0 is in error, s_0 is the only bit affected; the syndrome, therefore, is 001. If b_2 is in error, s_0 and s_1 are the bits affected; the syndrome, therefore is 011. Similarly, if b_1 is in error, all 3 syndrome bits are affected and the syndrome is 111.

There are two points we need to emphasize here. First, if two errors occur during transmission, the created dataword might not be the right one. Second, if we want to use the above code for error detection, we need a different design.

Example 10.13

Let us trace the path of three datawords from the sender to the destination:

1. The dataword 0100 becomes the codeword 0100011. The codeword 0100011 is received. The syndrome is 000 (no error), the final dataword is 0100.
2. The dataword 0111 becomes the codeword 0111001. The codeword 0011001 is received. The syndrome is 011. According to Table 10.5, b_2 is in error. After flipping b_2 (changing the 1 to 0), the final dataword is 0111.
3. The dataword 1101 becomes the codeword 1101000. The codeword 0001000 is received (two errors). The syndrome is 101, which means that b_0 is in error. After flipping b_0 we get 0000, the wrong dataword. This shows that our code cannot correct two errors.

Example 10.14

We need a dataword of at least 7 bits. Calculate values of k and n that satisfy this requirement.

Solution

We need to make $k = n - m$ greater than or equal to 7, or $2^m - 1 - m \geq 7$.

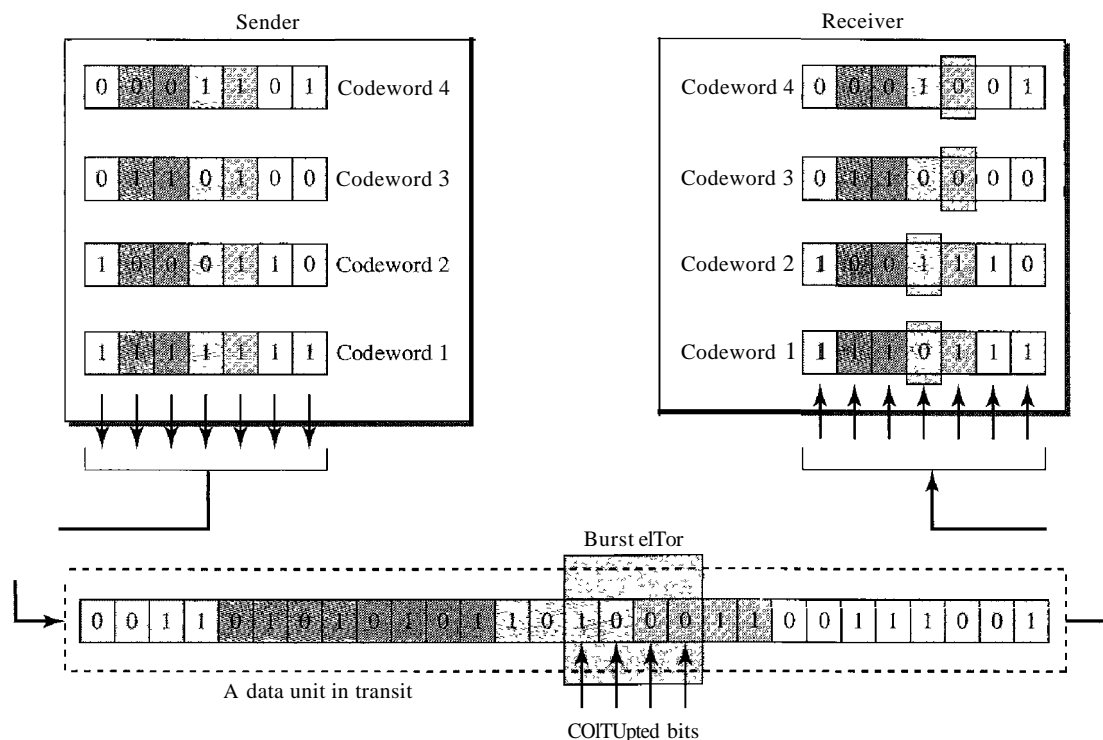
1. If we set $m = 3$, the result is $n = 2^3 - 1 = 7$ and $k = 7 - 3 = 4$, which is not acceptable.
2. If we set $m = 4$, then $n = 2^4 - 1 = 15$ and $k = 15 - 4 = 11$, which satisfies the condition. So the code is $C(15, 11)$. There are methods to make the dataword a specific size, but the discussion and implementation are beyond the scope of this book.

Performance

A Hamming code can only correct a single error or detect a double error. However, there is a way to make it detect a burst error, as shown in Figure 10.13.

The key is to split a burst error between several codewords, one error for each codeword. In data communications, we normally send a packet or a frame of data. To make the Hamming code respond to a burst error of size N , we need to make N codewords out of our frame. Then, instead of sending one codeword at a time, we arrange the codewords in a table and send the bits in the table a column at a time. In Figure 10.13, the bits are sent column by column (from the left). In each column, the bits are sent from the bottom to the top. In this way, a frame is made out of the four codewords and sent to the receiver. Figure 10.13 shows

Figure 10.13 Burst error correction using Hamming code



that when a burst error of size 4 corrupts the frame, only 1 bit from each codeword is corrupted. The corrupted bit in each codeword can then easily be corrected at the receiver.

10.4 CYCLIC CODES

Cyclic codes are special linear block codes with one extra property. In a cyclic code, if a codeword is cyclically shifted (rotated), the result is another codeword. For example, if 1011000 is a codeword and we cyclically left-shift, then 0110001 is also a codeword. In this case, if we call the bits in the first word a_0 to a_6 and the bits in the second word b_0 to b_6 , we can shift the bits by using the following:

$$b_1 = a_0 \quad b_2 = a_1 \quad b_3 = a_2 \quad b_4 = a_3 \quad b_5 = a_4 \quad b_6 = a_5 \quad b_0 = a_6$$

In the rightmost equation, the last bit of the first word is wrapped around and becomes the first bit of the second word.

Cyclic Redundancy Check

We can create cyclic codes to correct errors. However, the theoretical background required is beyond the scope of this book. In this section, we simply discuss a category of cyclic codes called the cyclic redundancy check (CRC) that is used in networks such as LANs and WANs.

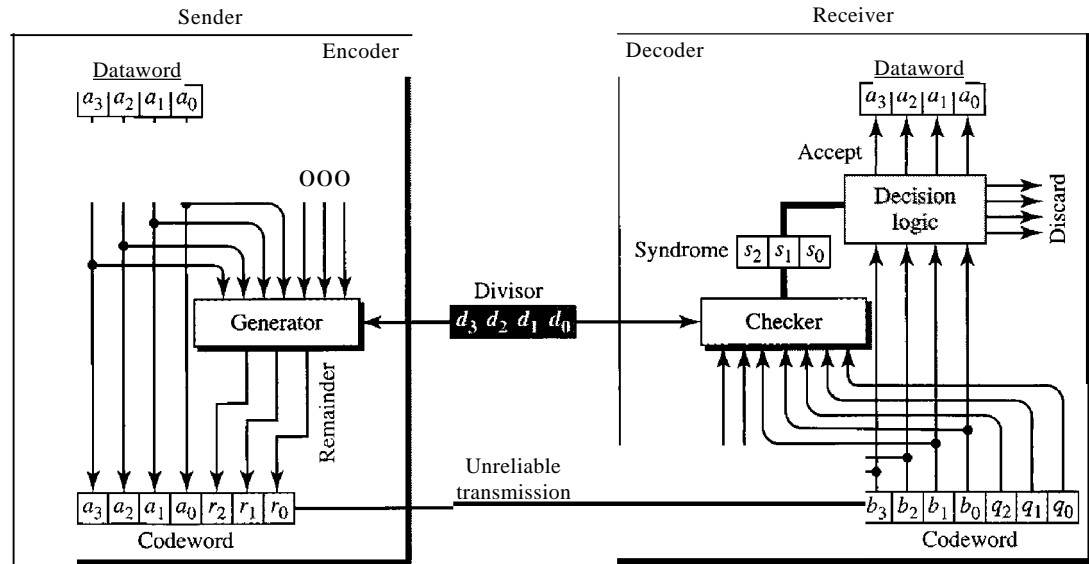
Table 10.6 shows an example of a CRC code. We can see both the linear and cyclic properties of this code.

Table 10.6 A CRC code with $C(7, 4)$

Dataword	Codeword	Dataword	Codeword
0000	0000000	1000	1000101
0001	0001011	1001	1001110
0010	0010110	1010	1010011
0011	0011101	1011	1011000
0100	0100111	1100	1100010
0101	0101100	1101	1101001
0110	0110001	1110	1110100
0111	0111010	1111	1111111

Figure 10.14 shows one possible design for the encoder and decoder.

Figure 10.14 CRC encoder and decoder



In the encoder, the dataword has k bits (4 here); the codeword has n bits (7 here). The size of the dataword is augmented by adding $n - k$ (3 here) 0s to the right-hand side of the word. The n -bit result is fed into the generator. The generator uses a divisor of size $n - k + 1$ (4 here), predefined and agreed upon. The generator divides the augmented dataword by the divisor (modulo-2 division). The quotient of the division is discarded; the remainder ($r_2r_1r_0$) is appended to the dataword to create the codeword.

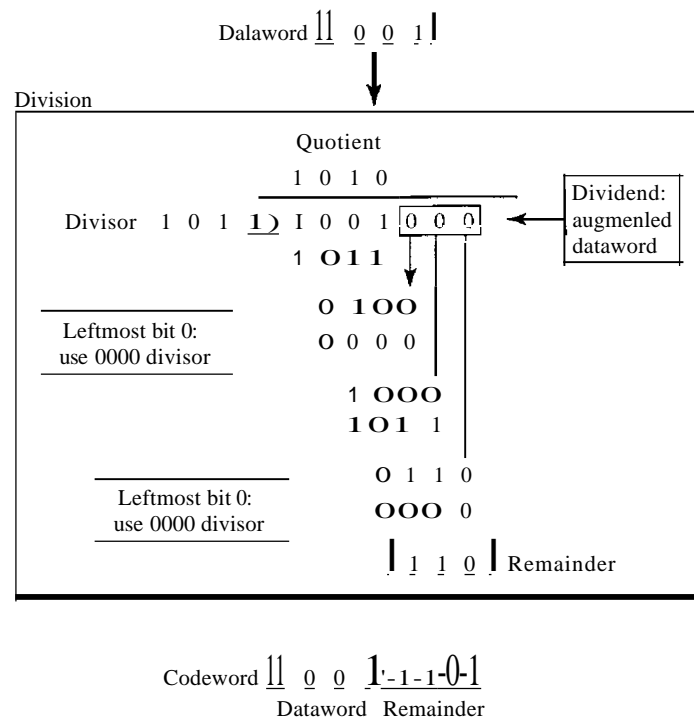
The decoder receives the possibly corrupted codeword. A copy of all n bits is fed to the checker which is a replica of the generator. The remainder produced by the checker

is a syndrome of $n - k$ (3 here) bits, which is fed to the decision logic analyzer. The analyzer has a simple function. If the syndrome bits are all as, the 4 leftmost bits of the codeword are accepted as the dataword (interpreted as no error); otherwise, the 4 bits are discarded (error).

Encoder

Let us take a closer look at the encoder. The encoder takes the dataword and augments it with $n - k$ number of as. It then divides the augmented dataword by the divisor, as shown in Figure 10.15.

Figure 10.15 Division in CRC encoder



The process of modulo-2 binary division is the same as the familiar division process we use for decimal numbers. However, as mentioned at the beginning of the chapter, in this case addition and subtraction are the same. We use the XOR operation to do both.

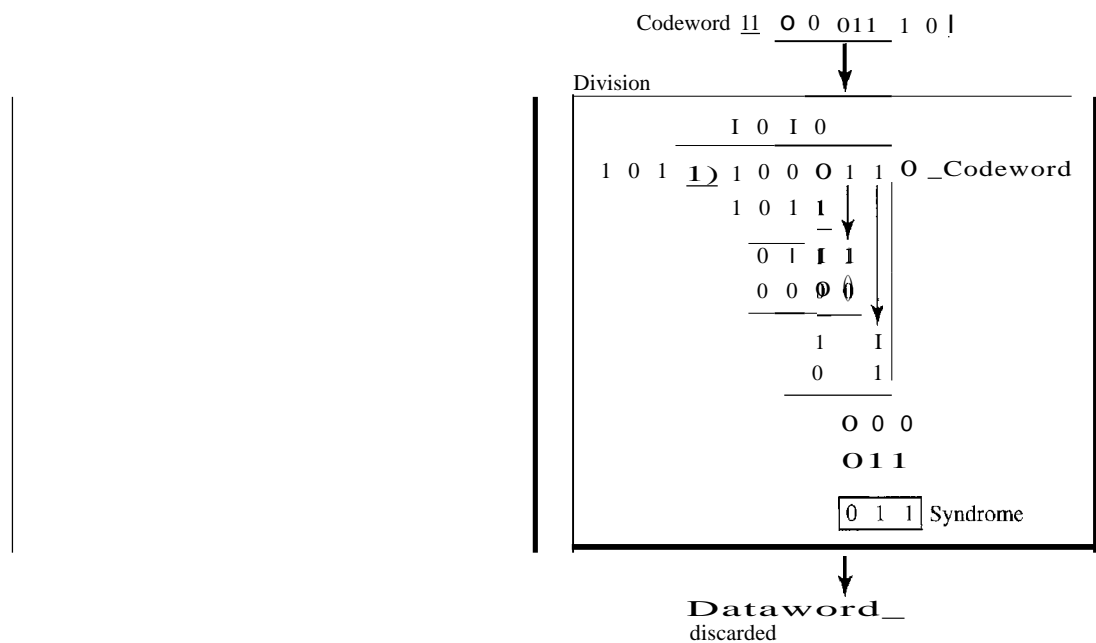
As in decimal division, the process is done step by step. In each step, a copy of the divisor is XORed with the 4 bits of the dividend. The result of the XOR operation (remainder) is 3 bits (in this case), which is used for the next step after 1 extra bit is pulled down to make it 4 bits long. There is one important point we need to remember in this type of division. If the leftmost bit of the dividend (or the part used in each step) is 0, the step cannot use the regular divisor; we need to use an all-Os divisor.

When there are no bits left to pull down, we have a result. The 3-bit remainder forms the check bits (r_2' , r_1' and r_0). They are appended to the dataword to create the codeword.

Decoder

The codeword can change during transmission. The decoder does the same division process as the encoder. The remainder of the division is the syndrome. If the syndrome is all 0s, there is no error; the dataword is separated from the received codeword and accepted. Otherwise, everything is discarded. Figure 10.16 shows two cases: The left-hand figure shows the value of syndrome when no error has occurred; the syndrome is 000. The right-hand part of the figure shows the case in which there is one single error. The syndrome is not all 0s (it is 011).

Figure 10.16 Division in the CRC decoder for two cases



Divisor

You may be wondering how the divisor 011 is chosen. Later in the chapter we present some criteria, but in general it involves abstract algebra.

Hardware Implementation

One of the advantages of a cyclic code is that the encoder and decoder can easily and cheaply be implemented in hardware by using a handful of electronic devices. Also, a hardware implementation increases the rate of check bit and syndrome bit calculation. In this section, we try to show, step by step, the process. The section, however, is optional and does not affect the understanding of the rest of the chapter.

Divisor

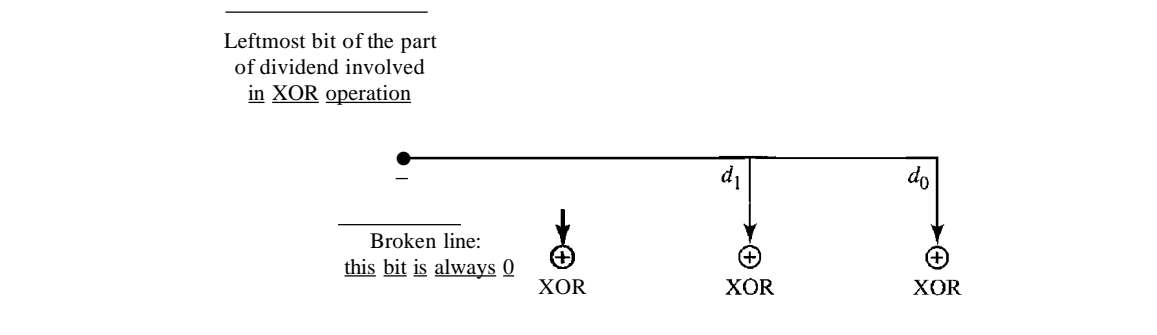
Let us first consider the divisor. We need to note the following points:

1. The divisor is repeatedly XORed with part of the dividend.

2. The divisor has $n - k + 1$ bits which either are predefined or are all 0s. In other words, the bits do not change from one dataword to another. In our previous example, the divisor bits were either 1011 or 0000. The choice was based on the leftmost bit of the part of the augmented data bits that are active in the XOR operation.
3. A close look shows that only $n - k$ bits of the divisor is needed in the XOR operation. The leftmost bit is not needed because the result of the operation is always 0, no matter what the value of this bit. The reason is that the inputs to this XOR operation are either both 0s or both 1s. In our previous example, only 3 bits, not 4, is actually used in the XOR operation.

Using these points, we can make a fixed (hardwired) divisor that can be used for a cyclic code if we know the divisor pattern. Figure 10.17 shows such a design for our previous example. We have also shown the XOR devices used for the operation.

Figure 10.17 Hardwired design of the divisor in CRC



Note that if the leftmost bit of the part of dividend to be used in this step is 1, the divisor bits ($d_2d_1d_0$) are all 1; if the leftmost bit is 0, the divisor bits are 000. The design provides the right choice based on the leftmost bit.

Augmented Dataword

In our paper-and-pencil division process in Figure 10.15, we show the augmented dataword as fixed in position with the divisor bits shifting to the right, 1 bit in each step. The divisor bits are aligned with the appropriate part of the augmented dataword. Now that our divisor is fixed, we need instead to shift the bits of the augmented dataword to the left (opposite direction) to align the divisor bits with the appropriate part. There is no need to store the augmented dataword bits.

Remainder

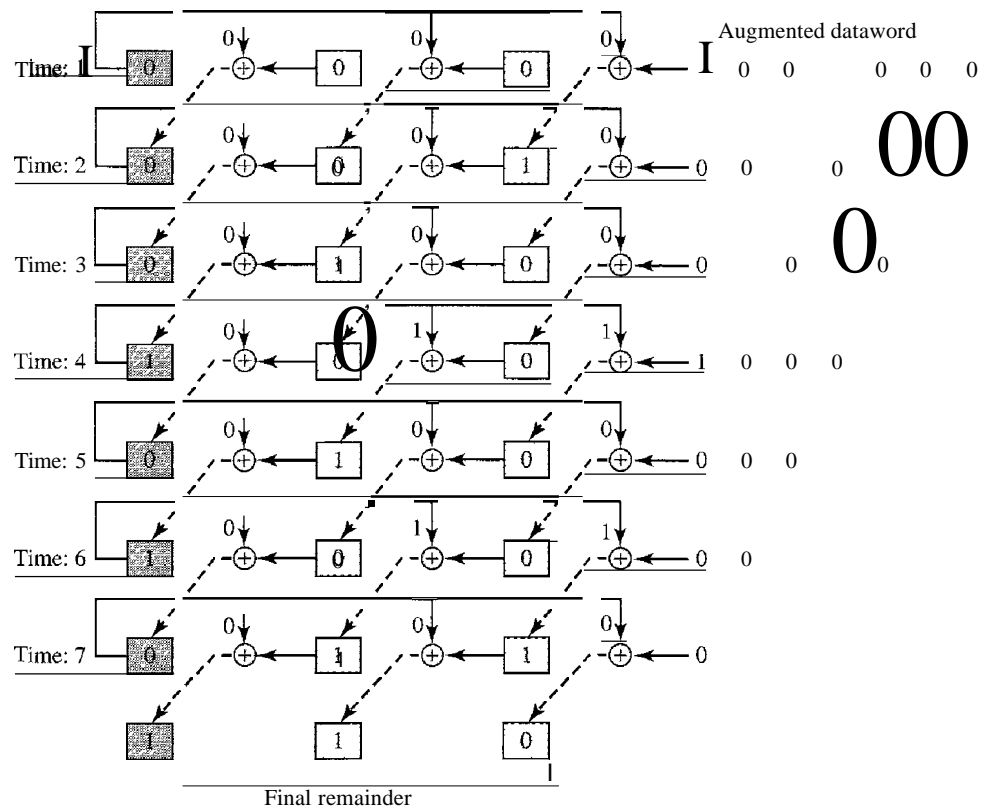
In our previous example, the remainder is 3 bits ($n - k$ bits in general) in length. We can use three registers (single-bit storage devices) to hold these bits. To find the final remainder of the division, we need to modify our division process. The following is the step-by-step process that can be used to simulate the division process in hardware (or even in software).

1. We assume that the remainder is originally all 0s (000 in our example).

2. At each time click (arrival of 1 bit from an augmented dataword), we repeat the following two actions:
 - a. We use the leftmost bit to make a decision about the divisor (011 or 000).
 - b. The other 2 bits of the remainder and the next bit from the augmented dataword (total of 3 bits) are XORed with the 3-bit divisor to create the next remainder.

Figure 10.18 shows this simulator, but note that this is not the final design; there will be more improvements.

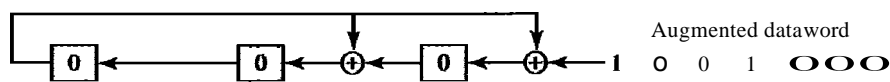
Figure 10.18 *Simulation of division in CRC encoder*



At each clock tick, shown as different times, one of the bits from the augmented dataword is used in the XOR process. If we look carefully at the design, we have seven steps here, while in the paper-and-pencil method we had only four steps. The first three steps have been added here to make each step equal and to make the design for each step the same. Steps 1, 2, and 3 push the first 3 bits to the remainder registers; steps 4, 5, 6, and 7 match the paper-and-pencil design. Note that the values in the remainder register in steps 4 to 7 exactly match the values in the paper-and-pencil design. The final remainder is also the same.

The above design is for demonstration purposes only. It needs simplification to be practical. First, we do not need to keep the intermediate values of the remainder bits; we need only the final bits. We therefore need only 3 registers instead of 24. After the XOR operations, we do not need the bit values of the previous remainder. Also, we do

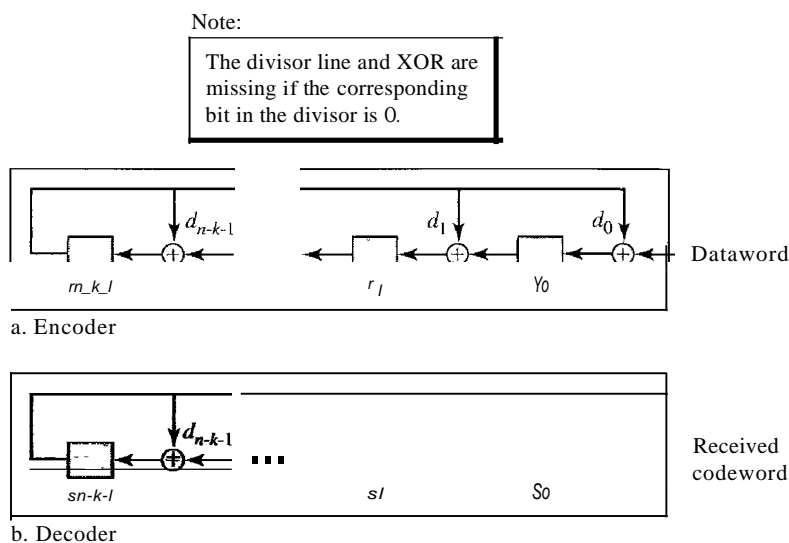
not need 21 XOR devices; two are enough because the output of an XOR operation in which one of the bits is 0 is simply the value of the other bit. This other bit can be used as the output. With these two modifications, the design becomes tremendously simpler and less expensive, as shown in Figure 10.19.

Figure 10.19 *The CRC encoder design using shift registers*

We need, however, to make the registers shift registers. A 1-bit shift register holds a bit for a duration of one clock time. At a time click, the shift register accepts the bit at its input port, stores the new bit, and displays it on the output port. The content and the output remain the same until the next input arrives. When we connect several 1-bit shift registers together, it looks as if the contents of the register are shifting.

General Design

A general design for the encoder and decoder is shown in Figure 10.20.

Figure 10.20 *General design of encoder and decoder of a CRC code*

Note that we have $n - k$ I-bit shift registers in both the encoder and decoder. We have up to $n - k$ XOR devices, but the divisors normally have several 0s in their pattern, which reduces the number of devices. Also note that, instead of augmented datawords, we show the dataword itself as the input because after the bits in the dataword are all fed into the encoder, the extra bits, which all are 0s, do not have any effect on the right-most XOR. Of course, the process needs to be continued for another $n - k$ steps before

the check bits are ready. This fact is one of the criticisms of this design. Better schemes have been designed to eliminate this waiting time (the check bits are ready after k steps), but we leave this as a research topic for the reader. In the decoder, however, the entire codeword must be fed to the decoder before the syndrome is ready.

Polynomials

A better way to understand cyclic codes and how they can be analyzed is to represent them as polynomials. Again, this section is optional.

A pattern of 0s and 1s can be represented as a **polynomial** with coefficients of 0 and 1. The power of each term shows the position of the bit; the coefficient shows the value of the bit. Figure 10.21 shows a binary pattern and its polynomial representation. In Figure 10.21a we show how to translate a binary pattern to a polynomial; in Figure 10.21b we show how the polynomial can be shortened by removing all terms with zero coefficients and replacing x^1 by x and x^0 by 1.

Figure 10.21 A polynomial to represent a binary word

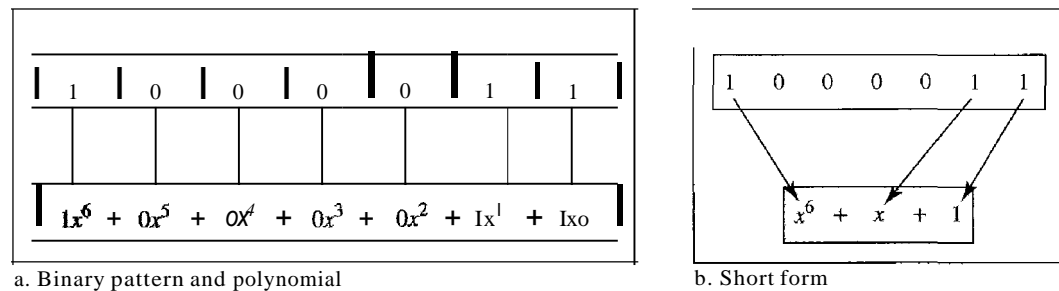


Figure 10.21 shows one immediate benefit; a 7-bit pattern can be replaced by three terms. The benefit is even more conspicuous when we have a polynomial such as $x^{23} + x^3 + 1$. Here the bit pattern is 24 bits in length (three 1s and twenty-one 0s) while the polynomial is just three terms.

Degree of a Polynomial

The degree of a polynomial is the highest power in the polynomial. For example, the degree of the polynomial $x^6 + x + 1$ is 6. Note that the degree of a polynomial is 1 less than the number of bits in the pattern. The bit pattern in this case has 7 bits.

Adding and Subtracting Polynomials

Adding and subtracting polynomials in mathematics are done by adding or subtracting the coefficients of terms with the same power. In our case, the coefficients are only 0 and 1, and adding is in modulo-2. This has two consequences. First, addition and subtraction are the same. Second, adding or subtracting is done by combining terms and deleting pairs of identical terms. For example, adding $x^5 + x^4 + x^2$ and $x^6 + x^4 + x^2$ gives just $x^6 + x^5$. The terms x^4 and x^2 are deleted. However, note that if we add, for example, three polynomials and we get x^2 three times, we delete a pair of them and keep the third.

Multiplying or Dividing Terms

In this arithmetic, multiplying a term by another term is very simple; we just add the powers. For example, $x^3 \times x^4$ is x^7 . For dividing, we just subtract the power of the second term from the power of the first. For example, x^5/x^2 is x^3 .

Multiplying Two Polynomials

Multiplying a polynomial by another is done term by term. Each term of the first polynomial must be multiplied by all terms of the second. The result, of course, is then simplified, and pairs of equal terms are deleted. The following is an example:

$$\begin{aligned}(x^5 + x^3 + x^2 + x)(x^2 + x + 1) \\= x^7 + x^6 + x^5 + x^5 + x^4 + x^3 + x^4 + x^3 + x^2 + x^3 + x^2 + x \\= x^7 + x^6 + x^3 + x\end{aligned}$$

Dividing One Polynomial by Another

Division of polynomials is conceptually the same as the binary division we discussed for an encoder. We divide the first term of the dividend by the first term of the divisor to get the first term of the quotient. We multiply the term in the quotient by the divisor and subtract the result from the dividend. We repeat the process until the dividend degree is less than the divisor degree. We will show an example of division later in this chapter.

Shifting

A binary pattern is often shifted a number of bits to the right or left. Shifting to the left means adding extra 0s as rightmost bits; shifting to the right means deleting some rightmost bits. Shifting to the left is accomplished by multiplying each term of the polynomial by x^m , where m is the number of shifted bits; shifting to the right is accomplished by dividing each term of the polynomial by x^m . The following shows shifting to the left and to the right. Note that we do not have negative powers in the polynomial representation.

$$\begin{array}{lll}\text{Shifting left 3 bits:} & 10011 \text{ becomes } 10011000 & x^4 + x + 1 \text{ becomes } x^7 + x^4 + x^3 \\ \text{Shifting right 3 bits:} & 10011 \text{ becomes } 10 & x^4 + x + 1 \text{ becomes } x\end{array}$$

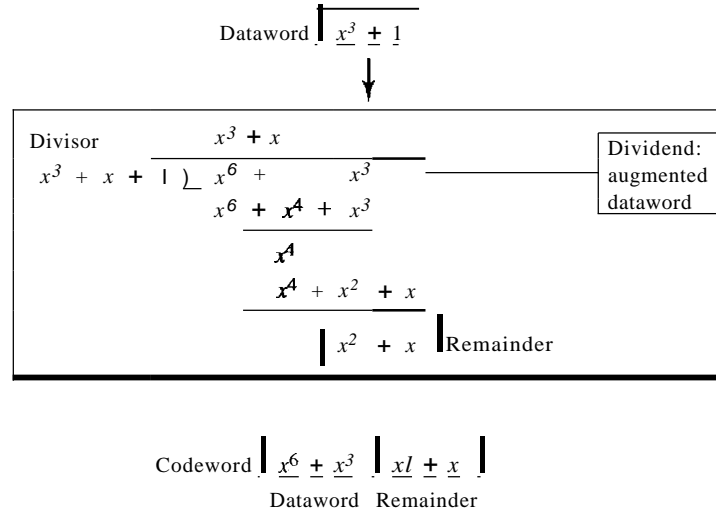
When we augmented the dataword in the encoder of Figure 10.15, we actually shifted the bits to the left. Also note that when we concatenate two bit patterns, we shift the first polynomial to the left and then add the second polynomial.

Cyclic Code Encoder Using Polynomials

Now that we have discussed operations on polynomials, we show the creation of a codeword from a dataword. Figure 10.22 is the polynomial version of Figure 10.15. We can see that the process is shorter. The dataword 1001 is represented as $x^3 + 1$. The divisor 1011 is represented as $x^3 + x + 1$. To find the augmented dataword, we have left-shifted the dataword 3 bits (multiplying by x^3). The result is $x^6 + x^3$. Division is straightforward. We divide the first term of the dividend, x^6 , by the first term of the divisor, x^3 . The first term of the quotient is then x^6/x^3 , or x^3 . Then we multiply x^3 by the divisor and subtract (according to our previous definition of subtraction) the result from the dividend. The

result is x^4 , with a degree greater than the divisor's degree; we continue to divide until the degree of the remainder is less than the degree of the divisor.

Figure 10.22 CRC division using polynomials



It can be seen that the polynomial representation can easily simplify the operation of division in this case, because the two steps involving all-Os divisors are not needed here. (Of course, one could argue that the all-Os divisor step can also be eliminated in binary division.) In a polynomial representation, the divisor is normally referred to as the generator polynomial $t(x)$.

The divisor in a cyclic code is normally called the generator polynomial or simply the generator.

Cyclic Code Analysis

We can analyze a cyclic code to find its capabilities by using polynomials. We define the following, where $f(x)$ is a polynomial with binary coefficients.

Dataword: $d(x)$	Codeword: $c(x)$	Generator: $g(x)$
Syndrome: $s(x)$	Error: $e(x)$	

If $s(x)$ is not zero, then one or more bits is corrupted. However, if $s(x)$ is zero, either no bit is corrupted or the decoder failed to detect any errors.

In a cyclic code,

1. If $s(x) \neq 0$, one or more bits is corrupted.
2. If $s(x) = 0$, either
 - a. No bit is corrupted. or
 - b. Some bits are corrupted, but the decoder failed to detect them.

In our analysis we want to find the criteria that must be imposed on the generator, $g(x)$ to detect the type of error we especially want to be detected. Let us first find the relationship among the sent codeword, error, received codeword, and the generator. We can say

$$\text{Received codeword} = c(x) + e(x)$$

In other words, the received codeword is the sum of the sent codeword and the error. The receiver divides the received codeword by $g(x)$ to get the syndrome. We can write this as

$$\frac{\text{Received codeword}}{g(x)} = \frac{c(x)}{g(x)} + \frac{e(x)}{g(x)}$$

The first term at the right-hand side of the equality does not have a remainder (according to the definition of codeword). So the syndrome is actually the remainder of the second term on the right-hand side. If this term does not have a remainder (syndrome = 0), either $e(x)$ is 0 or $e(x)$ is divisible by $g(x)$. We do not have to worry about the first case (there is no error); the second case is very important. Those errors that are divisible by $g(x)$ are not caught.

In a cyclic code, those $e(x)$ errors that are divisible by $g(x)$ are not caught.

Let us show some specific errors and see how they can be caught by a well-designed $g(x)$.

Single-Bit Error

What should be the structure of $g(x)$ to guarantee the detection of a single-bit error? A single-bit error is $e(x) = x^i$, where i is the position of the bit. If a single-bit error is caught, then x^i is not divisible by $g(x)$. (Note that when we say *not divisible*, we mean that there is a remainder.) If $g(x)$ has at least two terms (which is normally the case) and the coefficient of x^0 is not zero (the rightmost bit is 1), then $e(x)$ cannot be divided by $g(x)$.

If the generator has more than one term and the coefficient of x^0 is 1,
all single errors can be caught.

Example 10.15

Which of the following $g(x)$ values guarantees that a single-bit error is caught? For each case, what is the error that cannot be caught?

- a. $x + 1$
- b. x^3
- c. 1

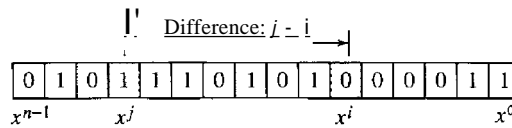
Solution

- No x^i can be divisible by $x + 1$. In other words, $x^i/(x + 1)$ always has a remainder. So the syndrome is nonzero. Any single-bit error can be caught.
- If i is equal to or greater than 3, x^i is divisible by $g(x)$. The remainder of x^i/x^3 is zero, and the receiver is fooled into believing that there is no error, although there might be one. Note that in this case, the corrupted bit must be in position 4 or above. All single-bit errors in positions 1 to 3 are caught.
- All values of i make x^i divisible by $g(x)$. No single-bit error can be caught. In addition, this $g(x)$ is useless because it means the codeword is just the dataword augmented with $n - k$ zeros.

Two Isolated Single-Bit Errors

Now imagine there are two single-bit isolated errors. Under what conditions can this type of error be caught? We can show this type of error as $e(x) = x^j + x^i$. The values of i and j define the positions of the errors, and the difference $j - i$ defines the distance between the two errors, as shown in Figure 10.23.

Figure 10.23 Representation of two isolated single-bit errors using polynomials



We can write $e(x) = x^i(x^{j-i} + 1)$. If $g(x)$ has more than one term and one term is x^0 , it cannot divide x^t , as we saw in the previous section. So if $g(x)$ is to divide $e(x)$, it must divide $x^{j-i} + 1$. In other words, $g(x)$ must not divide $x^t + 1$, where t is between 0 and $n - 1$. However, $t = 0$ is meaningless and $t = 1$ is needed as we will see later. This means t should be between 2 and $n - 1$.

If a generator cannot divide $x^t + 1$ (t between 0 and $n - 1$), then all isolated double errors can be detected.

Example 10.16

Find the status of the following generators related to two isolated, single-bit errors.

- $x + 1$
- $x^4 + 1$
- $x^7 + x^6 + 1$
- $x^{15} + x^{14} + 1$

Solution

- This is a very poor choice for a generator. Any two errors next to each other cannot be detected.
- This generator cannot detect two errors that are four positions apart. The two errors can be anywhere, but if their distance is 4, they remain undetected.
- This is a good choice for this purpose.
- This polynomial cannot divide any error of type $x^t + 1$ if t is less than 32,768. This means that a codeword with two isolated errors that are next to each other or up to 32,768 bits apart can be detected by this generator.

Odd Numbers of Errors

A generator with a factor of $x + 1$ can catch all odd numbers of errors. This means that we need to make $x + 1$ a factor of any generator. Note that we are not saying that the generator itself should be $x + 1$; we are saying that it should have a factor of $x + 1$. If it is only $x + 1$, it cannot catch the two adjacent isolated errors (see the previous section). For example, $x^4 + x^2 + x + 1$ can catch all odd-numbered errors since it can be written as a product of the two polynomials $x + 1$ and $x^3 + x^2 + 1$.

A generator that contains a factor of $x + 1$ can detect all odd-numbered errors.

Burst Errors

Now let us extend our analysis to the burst error, which is the most important of all. A burst error is of the form $e(x) = (x^j + \dots + x^i)$. Note the difference between a burst error and two isolated single-bit errors. The first can have two terms or more; the second can only have two terms. We can factor out x^i and write the error as $x^i(x^{j-i} + \dots + 1)$. If our generator can detect a single error (minimum condition for a generator), then it cannot divide x^i . What we should worry about are those generators that divide $x^{j-i} + \dots + 1$. In other words, the remainder of $(x^{j-i} + \dots + 1)/(x^r + \dots + 1)$ must not be zero. Note that the denominator is the generator polynomial. We can have three cases:

1. If $j - i < r$, the remainder can never be zero. We can write $j - i = L - 1$, where L is the length of the error. So $L - 1 < r$ or $L < r + 1$ or $L \leq r$. This means all burst errors with length smaller than or equal to the number of check bits r will be detected.
2. In some rare cases, if $j - i = r$, or $L = r + 1$, the syndrome is 0 and the error is undetected. It can be proved that in these cases, the probability of undetected burst error of length $r + 1$ is $(1/2)^{r-1}$. For example, if our generator is $x^{14} + x^3 + 1$, in which $r = 14$, a burst error of length $L = 15$ can slip by undetected with the probability of $(1/2)^{14-1}$ or almost 1 in 10,000.
3. In some rare cases, if $j - i > r$, or $L > r + 1$, the syndrome is 0 and the error is undetected. It can be proved that in these cases, the probability of undetected burst error of length greater than $r + 1$ is $(1/2)^r$. For example, if our generator is $x^{14} + x^3 + 1$, in which $r = 14$, a burst error of length greater than 15 can slip by undetected with the probability of $(1/2)^{14}$ or almost 1 in 16,000 cases.

-
- All burst errors with $L \leq r$ will be detected.
 - All burst errors with $L = r + 1$ will be detected with probability $1 - (1/2)^{r-1}$.
 - All burst errors with $L > r + 1$ will be detected with probability $1 - (1/2)^r$.
-

Example 10.17

Find the suitability of the following generators in relation to burst errors of different lengths.

- a. $x^6 + 1$
- b. $x^{18} + x^7 + x + 1$
- c. $x^{32} + x^{23} + x^7 + 1$

Solution

- a. This generator can detect all burst errors with a length less than or equal to 6 bits; 3 out of 100 burst errors with length 7 will slip by; 16 out of 1000 burst errors of length 8 or more will slip by.
- b. This generator can detect all burst errors with a length less than or equal to 18 bits; 8 out of 1 million burst errors with length 19 will slip by; 4 out of 1 million burst errors of length 20 or more will slip by.
- c. This generator can detect all burst errors with a length less than or equal to 32 bits; 5 out of 10 billion burst errors with length 33 will slip by; 3 out of 10 billion burst errors of length 34 or more will slip by.

Summary

We can summarize the criteria for a good polynomial generator:

A good polynomial generator needs to have the following characteristics:

1. It should have at least two terms.
 2. The coefficient of the term x^0 should be 1.
 3. It should not divide $x^t + 1$, for t between 2 and $n - 1$.
 4. It should have the factor $x + 1$.
-

Standard Polynomials

Some standard polynomials used by popular protocols for error generation are shown in Table 10.7.

Table 10.7 *Standard polynomials*

Name	Polynomial	Application
CRC-8	$x^8 + x^2 + x + 1$	ATM header
CRC-10	$x^{10} + x^9 + x^5 + x^4 + x^2 + 1$	ATMAAL
CRC-16	$x^{16} + x^{12} + x^5 + 1$	HDLC
CRC-32	$x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x + 1$	LANs

Advantages of Cyclic Codes

We have seen that cyclic codes have a very good performance in detecting single-bit errors, double errors, an odd number of errors, and burst errors. They can easily be implemented in hardware and software. They are especially fast when implemented in hardware. This has made cyclic codes a good candidate for many networks.

Other Cyclic Codes

The cyclic codes we have discussed in this section are very simple. The check bits and syndromes can be calculated by simple algebra. There are, however, more powerful polynomials that are based on abstract algebra involving Galois fields. These are beyond

the scope of this book. One of the most interesting of these codes is the Reed-Solomon code used today for both detection and correction.

10.5 CHECKSUM

The last error detection method we discuss here is called the checksum. The checksum is used in the Internet by several protocols although not at the data link layer. However, we briefly discuss it here to complete our discussion on error checking.

Like linear and cyclic codes, the checksum is based on the concept of redundancy. Several protocols still use the checksum for error detection as we will see in future chapters, although the tendency is to replace it with a CRC. This means that the CRC is also used in layers other than the data link layer.

Idea

The concept of the checksum is not difficult. Let us illustrate it with a few examples.

Example 10.18

Suppose our data is a list of five 4-bit numbers that we want to send to a destination. In addition to sending these numbers, we send the sum of the numbers. For example, if the set of numbers is (7, 11, 12, 0, 6), we send (7, 11, 12, 0, 6, 36), where 36 is the sum of the original numbers. The receiver adds the five numbers and compares the result with the sum. If the two are the same, the receiver assumes no error, accepts the five numbers, and discards the sum. Otherwise, there is an error somewhere and the data are not accepted.

Example 10.19

We can make the job of the receiver easier if we send the negative (complement) of the sum, called the *checksum*. In this case, we send (7, 11, 12, 0, 6, -36). The receiver can add all the numbers received (including the checksum). If the result is 0, it assumes no error; otherwise, there is an error.

One's Complement

The previous example has one major drawback. All of our data can be written as a 4-bit word (they are less than 15) except for the checksum. One solution is to use one's complement arithmetic. In this arithmetic, we can represent unsigned numbers between 0 and $2^n - 1$ using only n bits.^t If the number has more than n bits, the extra leftmost bits need to be added to the n rightmost bits (wrapping). In one's complement arithmetic, a negative number can be represented by inverting all bits (changing a 0 to a 1 and a 1 to a 0). This is the same as subtracting the number from $2^n - 1$.

Example 10.20

How can we represent the number 21 in one's complement arithmetic using only four bits?

^tAlthough one's complement can represent both positive and negative numbers, we are concerned only with unsigned representation here.

Solution

The number 21 in binary is 10101 (it needs five bits). We can wrap the leftmost bit and add it to the four rightmost bits. We have $(0101 + 1) = 0110$ or 6.

Example 10.21

How can we represent the number -6 in one's complement arithmetic using only four bits?

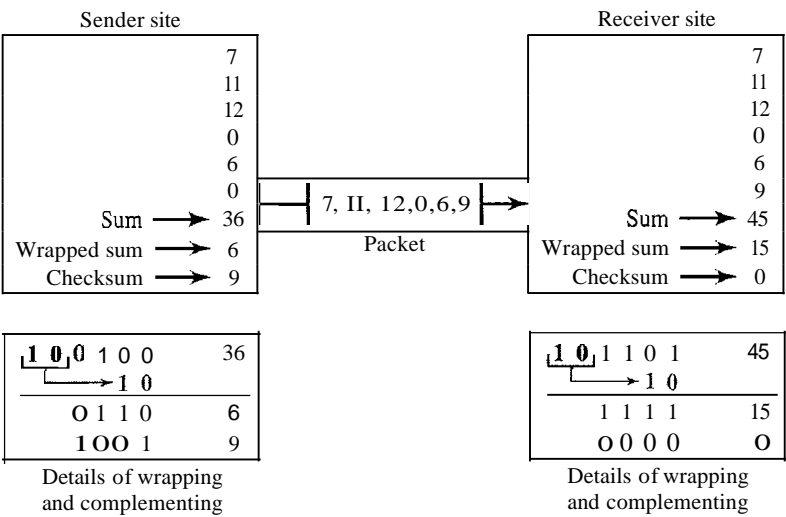
Solution

In one's complement arithmetic, the negative or complement of a number is found by inverting all bits. Positive 6 is 0110; negative 6 is 1001. If we consider only unsigned numbers, this is 9. In other words, the complement of 6 is 9. Another way to find the complement of a number in one's complement arithmetic is to subtract the number from $2^n - 1$ (16 - 1 in this case).

Example 10.22

Let us redo Exercise 10.19 using one's complement arithmetic. Figure 10.24 shows the process at the sender and at the receiver. The sender initializes the checksum to 0 and adds all data items and the checksum (the checksum is considered as one data item and is shown in color). The result is 36. However, 36 cannot be expressed in 4 bits. The extra two bits are wrapped and added with the sum to create the wrapped sum value 6. In the figure, we have shown the details in binary. The sum is then complemented, resulting in the checksum value 9 ($15 - 6 = 9$). The sender now sends six data items to the receiver including the checksum 9. The receiver follows the same procedure as the sender. It adds all data items (including the checksum); the result is 45. The sum is wrapped and becomes 15. The wrapped sum is complemented and becomes 0. Since the value of the checksum is 0, this means that the data is not corrupted. The receiver drops the checksum and keeps the other data items. If the checksum is not zero, the entire packet is dropped.

Figure 10.24



Internet Checksum

Traditionally, the Internet has been using a 16-bit checksum. The sender calculates the checksum by following these steps.

Sender site:

1. The message is divided into 16-bit words.
 2. The value of the checksum word is set to 0.
 3. All words including the checksum are added using one's complement addition.
 4. The sum is complemented and becomes the checksum.
 5. The checksum is sent with the data.
-

The receiver uses the following steps for error detection.

Receiver site:

1. The message (including checksum) is divided into 16-bit words.
 2. All words are added using one's complement addition.
 3. The sum is complemented and becomes the new checksum.
 4. If the value of checksum is 0, the message is accepted; otherwise, it is rejected.
-

The nature of the checksum (treating words as numbers and adding and complementing them) is well-suited for software implementation. Short programs can be written to calculate the checksum at the receiver site or to check the validity of the message at the receiver site.

Example 10.23

Let us calculate the checksum for a text of 8 characters ("Forouzan"). The text needs to be divided into 2-byte (16-bit) words. We use ASCII (see Appendix A) to change each byte to a 2-digit hexadecimal number. For example, F is represented as 0x46 and o is represented as 0x6F. Figure 10.25 shows how the checksum is calculated at the sender and receiver sites. In part a of the figure, the value of partial sum for the first column is 0x36. We keep the rightmost digit (6) and insert the

Figure 10.25

1	0	1	3	Carries
4	6	6	F	(Fo)
7	2	6	F	(ro)
7	5	7	A	luz)
6	1	6	E	(an)
0	0	0	0	Checksum (initial)
8	F	C	6	Sum (partial)
			1	
8	F	C	7	Sum
7	0	3	8	Checksum (to send)

a. Checksum at the sender site

1	0	1	3	Carries
4	6	6	F	IFo)
7	2	6	F	(ro)
7	5	7	A	(uz)
6	1	6	E	(an)
7	0	3	8	Checksum (received)
F	F	F	E	Sum (partial)
			1	
F	F	F	F	Sum
0	0	0	0	Checksum (new)

b. Checksum at the receiver site

leftmost digit (3) as the carry in the second column. The process is repeated for each column. Hexadecimal numbers are reviewed in Appendix B.

Note that if there is any corruption, the checksum recalculated by the receiver is not all as. We leave this an exercise.

Performance

The traditional checksum uses a small number of bits (16) to detect errors in a message of any size (sometimes thousands of bits). However, it is not as strong as the CRC in error-checking capability. For example, if the value of one word is incremented and the value of another word is decremented by the same amount, the two errors cannot be detected because the sum and checksum remain the same. Also if the values of several words are incremented but the total change is a multiple of 65535, the sum and the checksum does not change, which means the errors are not detected. Fletcher and Adler have proposed some weighted checksums, in which each word is multiplied by a number (its weight) that is related to its position in the text. This will eliminate the first problem we mentioned. However, the tendency in the Internet, particularly in designing new protocols, is to replace the checksum with a CRC.

10.6 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [...] refer to the reference list at the end of the text.

Books

Several excellent book are devoted to error coding. Among them we recommend [Ham80], [Zar02], [Ror96], and [SWE04].

RFCs

A discussion of the use of the checksum in the Internet can be found in RFC 1141.

10.7 KEY TERMS

block code	error correction
burst error	error detection
check bit	forward error correction
checksum	generator polynomial
codeword	Hamming code
convolution code	Hamming distance
cyclic code	interference
cyclic redundancy check (CRC)	linear block code
dataword	minimum Hamming distance
error	modular arithmetic

modulus	register
one's complement	retransmission
parity bit	shift register
parity-check code	single-bit error
polynomial	syndrome
redundancy	two-dimensional parity
Reed-Solomon	check

10.8 SUMMARY

- Data can be corrupted during transmission. Some applications require that errors be detected and corrected.
- In a single-bit error, only one bit in the data unit has changed. A burst error means that two or more bits in the data unit have changed.
- To detect or correct errors, we need to send extra (redundant) bits with data.
- There are two main methods of error correction: forward error correction and correction by retransmission.
- We can divide coding schemes into two broad categories: *block coding* and *convolution coding*.
- In coding, we need to use modulo-2 arithmetic. Operations in this arithmetic are very simple; addition and subtraction give the same results. we use the XOR (exclusive OR) operation for both addition and subtraction.
- In block coding, we divide our message into blocks, each of k bits, called datawords. We add r redundant bits to each block to make the length $n :: k + r$. The resulting n -bit blocks are called codewords.
- In block coding, errors be detected by using the following two conditions:
 - a. The receiver has (or can find) a list of valid codewords.
 - b. The original codeword has changed to an invalid one.
- The Hamming distance between two words is the number of differences between corresponding bits. The minimum Hamming distance is the smallest Hamming distance between all possible pairs in a set of words.
- To guarantee the detection of up to s errors in all cases, the minimum Hamming distance in a block code must be $d_{\min} :: s + 1$. To guarantee correction of up to t errors in all cases, the minimum Hamming distance in a block code must be $d_{\min} :: 2t + 1$.
- In a linear block code, the exclusive OR (XOR) of any two valid codewords creates another valid codeword.
- A simple parity-check code is a single-bit error-detecting code in which $n :: k + 1$ with $d_{\min} :: 2$. A simple parity-check code can detect an odd number of errors.
- All Hamming codes discussed in this book have $d_{\min} :: 3$. The relationship between m and n in these codes is $n :: 2m - 1$.
- Cyclic codes are special linear block codes with one extra property. In a cyclic code, if a codeword is cyclically shifted (rotated), the result is another codeword.

- A category of cyclic codes called the cyclic redundancy check (CRC) is used in networks such as LANs and WANs.
- A pattern of 0s and 1s can be represented as a polynomial with coefficients of 0 and 1.
- Traditionally, the Internet has been using a 16-bit checksum, which uses *one's complement* arithmetic. In this arithmetic, we can represent unsigned numbers between 0 and $2^n - 1$ using only n bits.

10.9 PRACTICE SET

Review Questions

1. How does a single-bit error differ from a burst error?
2. Discuss the concept of redundancy in error detection and correction.
3. Distinguish between forward error correction versus error correction by retransmission.
4. What is the definition of a linear block code? What is the definition of a cyclic code?
5. What is the Hamming distance? What is the minimum Hamming distance?
6. How is the simple parity check related to the two-dimensional parity check?
7. In CRC, show the relationship between the following entities (size means the number of bits):
 - a. The size of the dataword and the size of the codeword
 - b. The size of the divisor and the remainder
 - c. The degree of the polynomial generator and the size of the divisor
 - d. The degree of the polynomial generator and the size of the remainder
8. What kind of arithmetic is used to add data items in checksum calculation?
9. What kind of error is undetectable by the checksum?
10. Can the value of a checksum be all 0s (in binary)? Defend your answer. Can the value be all 1s (in binary)? Defend your answer.

Exercises

11. What is the maximum effect of a 2-ms burst of noise on data transmitted at the following rates?
 - a. 1500 bps
 - b. 12 kbps
 - c. 100 kbps
 - d. 100 Mbps
12. Apply the exclusive-or operation on the following pair of patterns (the symbol \oplus means XOR):
 - a. (10001) \oplus (10000)
 - b. (10001) \oplus (10001) (What do you infer from the result?)
 - c. (11100) \oplus (00000) (What do you infer from the result?)
 - d. (10011) \oplus (11111) (What do you infer from the result?)

13. In Table 10.1, the sender sends dataword 10. A 3-bit burst error corrupts the codeword. Can the receiver detect the error? Defend your answer.
14. In Table 10.2, the sender sends dataword 10. If a 3-bit burst error corrupts the first three bits of the codeword, can the receiver detect the error? Defend your answer.
15. What is the Hamming distance for each of the following codewords:
 - a. d (10000, 00000)
 - b. d (10101, 10000)
 - c. d (11111, 11111)
 - d. d (000, 000)
16. Find the minimum Hamming distance for the following cases:
 - a. Detection of two errors.
 - b. Correction of two errors.
 - c. Detection of 3 errors or correction of 2 errors.
 - d. Detection of 6 errors or correction of 2 errors.
17. Using the code in Table 10.2, what is the dataword if one of the following codewords is received?
 - a. 01011
 - b. 11111
 - c. 00000
 - d. 11011
18. Prove that the code represented by Table 10.8 is not a linear code. You need to find only one case that violates the linearity.

Table 10.8 Table for Exercise 18

<i>Dataword</i>	<i>Codeword</i>
00	00000
01	01011
10	10111
11	11111

19. Although it can mathematically be proved that a simple parity check code is a linear code, use manual testing of linearity for five pairs of the codewords in Table 10.3 to partially prove this fact.
20. Show that the Hamming code C(7,4) of Table 10A can detect two-bit errors but not necessarily three-bit error by testing the code in the following cases. The character “V” in the burst error means no error; the character “E” means an error.
 - a. Dataword: 0100 Burst error: VEEVVVV
 - b. Dataword: 0111 Burst error: EVVVVVE
 - c. Dataword: 1111 Burst error: EVEVVVE
 - d. Dataword: 0000 Burst error: EEVEVVV

21. Show that the Hamming code $C(7,4)$ of Table 10A can correct one-bit errors but not more by testing the code in the following cases. The character "V" in the burst error means no error; the character "E" means an error.
- Dataword: 0100 Burst error: **E**VVVVVV
 - Dataword: 0111 Burst error: V**E**VVVVV
 - Dataword: 1111 Burst error: **E**VVVV**E**
 - Dataword: 0000 Burst error: **E**EVVVV**E**
22. Although it can be proved that code in Table 10.6 is both linear and cyclic, use only two tests to partially prove the fact:
- Test the cyclic property on codeword 0101100.
 - Test the linear property on codewords 0010110 and 1111111.
23. We need a dataword of at least 11 bits. Find the values of k and n in the Hamming code $C(n, k)$ with $d_{min} = 3$.
24. Apply the following operations on the corresponding polynomials:
- $(x^3 + xl + x + 1) + (x^4 + xl + x + 1)$
 - $(x^3 + xl + x + 1) - (x^4 + xl + x + 1)$
 - $(x^3 + xl) \times (x^4 + x^2 + x + 1)$
 - $(x^3 + x^2 + x + 1) / (x^2 + 1)$
25. Answer the following questions:
- What is the polynomial representation of 101110?
 - What is the result of shifting 101110 three bits to the left?
 - Repeat part b using polynomials.
 - What is the result of shifting 101110 four bits to the right?
 - Repeat part d using polynomials.
26. Which of the following CRC generators guarantee the detection of a single bit error?
- $x^3 + x + 1$
 - $x^4 + xl$
 - 1
 - $x^2 + 1$
27. Referring to the CRC-8 polynomial in Table 10.7, answer the following questions:
- Does it detect a single error? Defend your answer.
 - Does it detect a burst error of size 6? Defend your answer.
 - What is the probability of detecting a burst error of size 9?
 - What is the probability of detecting a burst error of size 15?
28. Referring to the CRC-32 polynomial in Table 10.7, answer the following questions:
- Does it detect a single error? Defend your answer.
 - Does it detect a burst error of size 16? Defend your answer.
 - What is the probability of detecting a burst error of size 33?
 - What is the probability of detecting a burst error of size 55?

29. Assuming even parity, find the parity bit for each of the following data units.
 - a. 1001011
 - b. 0001100
 - c. 1000000
 - d. 1110111
30. Given the dataword 1010011110 and the divisor 10111,
 - a. Show the generation of the codeword at the sender site (using binary division).
 - h. Show the checking of the codeword at the receiver site (assume no error).
31. Repeat Exercise 30 using polynomials.
32. A sender needs to send the four data items 0x3456, 0xABCC, 0x02BC, and 0xEEEE. Answer the following:
 - a. Find the checksum at the sender site.
 - b. Find the checksum at the receiver site if there is no error.
 - c. Find the checksum at the receiver site if the second data item is changed to 0xABCE.
 - d. Find the checksum at the receiver site if the second data item is changed to 0xABCE and the third data item is changed to 0x02BA.
33. This problem shows a special case in checksum handling. A sender has two data items to send: 0x4567 and 0xBA98. What is the value of the checksum?

CHAPTER 11

Data Link Control

The two main functions of the data link layer are data link control and media access control. The first, data link control, deals with the design and procedures for communication between two adjacent nodes: node-to-node communication. We discuss this functionality in this chapter. The second function of the data link layer is media access control, or how to share the link. We discuss this functionality in Chapter 12.

Data link control functions include framing, flow and error control, and software-implemented protocols that provide smooth and reliable transmission of frames between nodes. In this chapter, we first discuss framing, or how to organize the bits that are carried by the physical layer. We then discuss flow and error control. A subset of this topic, techniques for error detection and correction, was discussed in Chapter 10.

To implement data link control, we need protocols. Each protocol is a set of rules that need to be implemented in software and run by the two nodes involved in data exchange at the data link layer. We discuss five protocols: two for noiseless (ideal) channels and three for noisy (real) channels. Those in the first category are not actually implemented, but provide a foundation for understanding the protocols in the second category.

After discussing the five protocol designs, we show how a bit-oriented protocol is actually implemented by using the High-level Data Link Control (HDLC) Protocol as an example. We also discuss a popular byte-oriented protocol, Point-to-Point Protocol (PPP).

11.1 FRAMING

Data transmission in the physical layer means moving bits in the form of a signal from the source to the destination. The physical layer provides bit synchronization to ensure that the sender and receiver use the same bit durations and timing.

The data link layer, on the other hand, needs to pack bits into frames, so that each frame is distinguishable from another. Our postal system practices a type of framing. The simple act of inserting a letter into an envelope separates one piece of information from another; the envelope serves as the delimiter. In addition, each envelope defines the sender and receiver addresses since the postal system is a many-to-many carrier facility.

Framing in the data link layer separates a message from one source to a destination, or from other messages to other destinations, by adding a sender address and a destination address. The destination address defines where the packet is to go; the sender address helps the recipient acknowledge the receipt.

Although the whole message could be packed in one frame, that is not normally done. One reason is that a frame can be very large, making flow and error control very inefficient. When a message is carried in one very large frame, even a single-bit error would require the retransmission of the whole message. When a message is divided into smaller frames, a single-bit error affects only that small frame.

Fixed-Size Framing

Frames can be of fixed or variable size. In fixed-size framing, there is no need for defining the boundaries of the frames; the size itself can be used as a delimiter. An example of this type of framing is the ATM wide-area network, which uses frames of fixed size called cells. We discuss ATM in Chapter 18.

Variable-Size Framing


Our main discussion in this chapter concerns variable-size framing, prevalent in local-area networks. In variable-size framing, we need a way to define the end of the frame and the beginning of the next. Historically, two approaches were used for this purpose: a character-oriented approach and a bit-oriented approach.

Character-Oriented Protocols

In a character-oriented protocol, data to be carried are 8-bit characters from a coding system such as ASCII (see Appendix A). The header, which normally carries the source and destination addresses and other control information, and the trailer, which carries error detection or error correction redundant bits, are also multiples of 8 bits. To separate one frame from the next, an 8-bit (1-byte) flag is added at the beginning and the end of a frame. The flag, composed of protocol-dependent special characters, signals the start or end of a frame. Figure 11.1 shows the format of a frame in a character-oriented protocol.

Figure 11.1 *A frame in a character-oriented protocol*

Data from upper layer

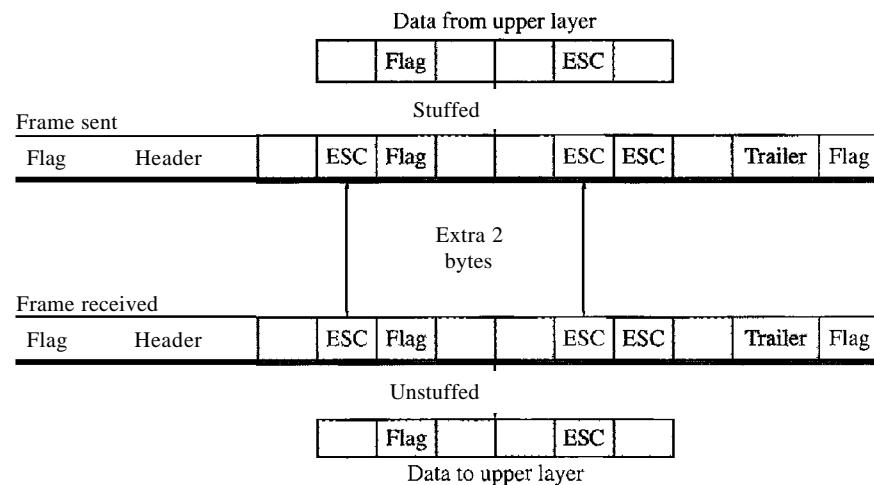


Character-oriented framing was popular when only text was exchanged by the data link layers. The flag could be selected to be any character not used for text communication. Now, however, we send other types of information such as graphs, audio, and video. Any pattern used for the flag could also be part of the information. If this happens, the receiver, when it encounters this pattern in the middle of the data, thinks it has reached the end of the frame. To fix this problem, a byte-stuffing strategy was added to

character-oriented framing. In byte stuffing (or character stuffing), a special byte is added to the data section of the frame when there is a character with the same pattern as the flag. The data section is stuffed with an extra byte. This byte is usually called the escape character (ESC), which has a predefined bit pattern. Whenever the receiver encounters the ESC character, it removes it from the data section and treats the next character as data, not a delimiting flag.

Byte stuffing by the escape character allows the presence of the flag in the data section of the frame, but it creates another problem. What happens if the text contains one or more escape characters followed by a flag? The receiver removes the escape character, but keeps the flag, which is incorrectly interpreted as the end of the frame. To solve this problem, the escape characters that are part of the text must also be marked by another text escape character. In other words, if the escape character is part of the text, an extra one is added to show that the second one is part of the text. Figure 11.2 shows the situation.

Figure 11.2 *Byte stuffing and unstuffing*



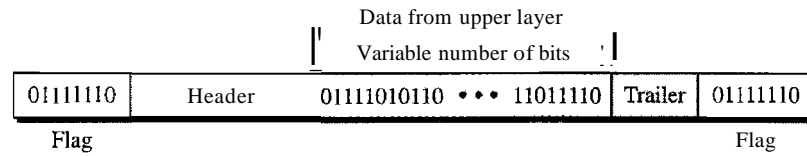
Byte stuffing is the process of adding 1 extra byte whenever there is a flag or escape character in the text.

Character-oriented protocols present another problem in data communications. The universal coding systems in use today, such as Unicode, have 16-bit and 32-bit characters that conflict with 8-bit characters. We can say that in general, the tendency is moving toward the bit-oriented protocols that we discuss next.

Bit-Oriented Protocols

In a bit-oriented protocol, the data section of a frame is a sequence of bits to be interpreted by the upper layer as text, graphic, audio, video, and so on. However, in addition to headers (and possible trailers), we still need a delimiter to separate one frame from the other. Most protocols use a special 8-bit pattern flag 01111110 as the delimiter to define the beginning and the end of the frame, as shown in Figure 11.3.

Figure 11.3 A frame in a bit-oriented protocol

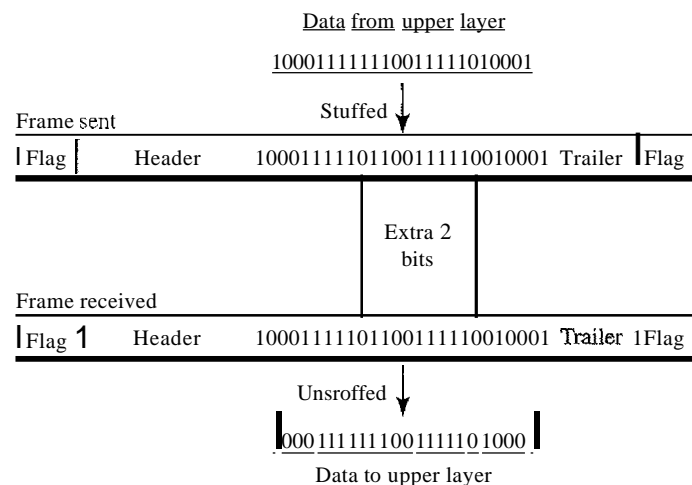


This flag can create the same type of problem we saw in the byte-oriented protocols. That is, if the flag pattern appears in the data, we need to somehow inform the receiver that this is not the end of the frame. We do this by stuffing 1 single bit (instead of 1 byte) to prevent the pattern from looking like a flag. The strategy is called bit stuffing. In bit stuffing, if a 0 and five consecutive 1 bits are encountered, an extra 0 is added. This extra stuffed bit is eventually removed from the data by the receiver. Note that the extra bit is added after one 0 followed by five 1s regardless of the value of the next bit. This guarantees that the flag field sequence does not inadvertently appear in the frame.

Bit stuffing is the process of adding one extra 0 whenever five consecutive 1s follow a 0 in the data, so that the receiver does not mistake the pattern 0111110 for a flag.

Figure 11.4 shows bit stuffing at the sender and bit removal at the receiver. Note that even if we have a 0 after five 1s, we still stuff a 0. The 0 will be removed by the receiver.

Figure 11.4 Bit stuffing and unstuffing



This means that if the flaglike pattern 01111110 appears in the data, it will change to 011111010 (stuffed) and is not mistaken as a flag by the receiver. The real flag 01111110 is not stuffed by the sender and is recognized by the receiver.

11.2 FLOW AND ERROR CONTROL

Data communication requires at least two devices working together, one to send and the other to receive. Even such a basic arrangement requires a great deal of coordination for an intelligible exchange to occur. The most important responsibilities of the data link layer are flow control and error control. Collectively, these functions are known as data link control.

Flow Control

Flow control coordinates the amount of data that can be sent before receiving an acknowledgment and is one of the most important duties of the data link layer. In most protocols, flow control is a set of procedures that tells the sender how much data it can transmit before it must wait for an acknowledgment from the receiver. The flow of data must not be allowed to overwhelm the receiver. Any receiving device has a limited speed at which it can process incoming data and a limited amount of memory in which to store incoming data. The receiving device must be able to inform the sending device before those limits are reached and to request that the transmitting device send fewer frames or stop temporarily. Incoming data must be checked and processed before they can be used. The rate of such processing is often slower than the rate of transmission. For this reason, each receiving device has a block of memory, called a *buffer*, reserved for storing incoming data until they are processed. If the buffer begins to fill up, the receiver must be able to tell the sender to halt transmission until it is once again able to receive.

Flow control refers to a set of procedures used to restrict the amount of data that the sender can send before waiting for acknowledgment.

Error Control

Error control is both error detection and error correction. It allows the receiver to inform the sender of any frames lost or damaged in transmission and coordinates the retransmission of those frames by the sender. In the data link layer, the term *error control* refers primarily to methods of error detection and retransmission. Error control in the data link layer is often implemented simply: Any time an error is detected in an exchange, specified frames are retransmitted. This process is called automatic repeat request (ARQ).

Error control in the data link layer is based on automatic repeat request, which is the retransmission of data.

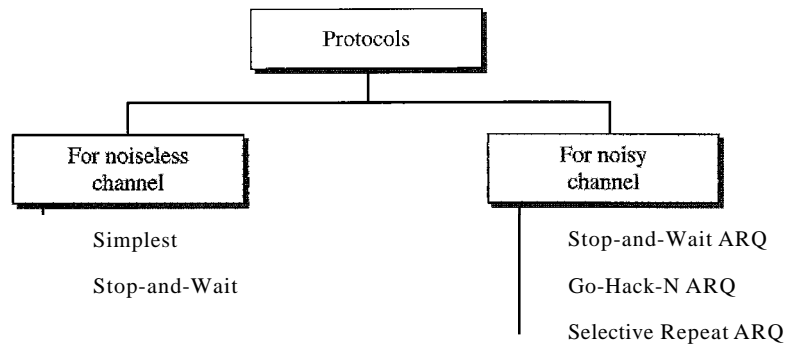
11.3 PROTOCOLS

Now let us see how the data link layer can combine framing, flow control, and error control to achieve the delivery of data from one node to another. The protocols are normally implemented in software by using one of the common programming languages. To make our

discussions language-free, we have written in pseudocode a version of each protocol that concentrates mostly on the procedure instead of delving into the details of language rules.

We divide the discussion of protocols into those that can be used for noiseless (error-free) channels and those that can be used for noisy (error-creating) channels. The protocols in the first category cannot be used in real life, but they serve as a basis for understanding the protocols of noisy channels. Figure 11.5 shows the classifications.

Figure 11.5 *Taxonomy of protocols discussed in this chapter*



There is a difference between the protocols we discuss here and those used in real networks. All the protocols we discuss are unidirectional in the sense that the data frames travel from one node, called the sender, to another node, called the receiver. Although special frames, called acknowledgment (ACK) and negative acknowledgment (NAK) can flow in the opposite direction for flow and error control purposes, data flow in only one direction.

In a real-life network, the data link protocols are implemented as bidirectional; data flow in both directions. In these protocols the flow and error control information such as ACKs and NAKs is included in the data frames in a technique called piggybacking. Because bidirectional protocols are more complex than unidirectional ones, we chose the latter for our discussion. If they are understood, they can be extended to bidirectional protocols. We leave this extension as an exercise.

11.4 NOISELESS CHANNELS

Let us first assume we have an ideal channel in which no frames are lost, duplicated, or corrupted. We introduce two protocols for this type of channel. The first is a protocol that does not use flow control; the second is the one that does. Of course, neither has error control because we have assumed that the channel is a perfect noiseless channel.

Simplest Protocol

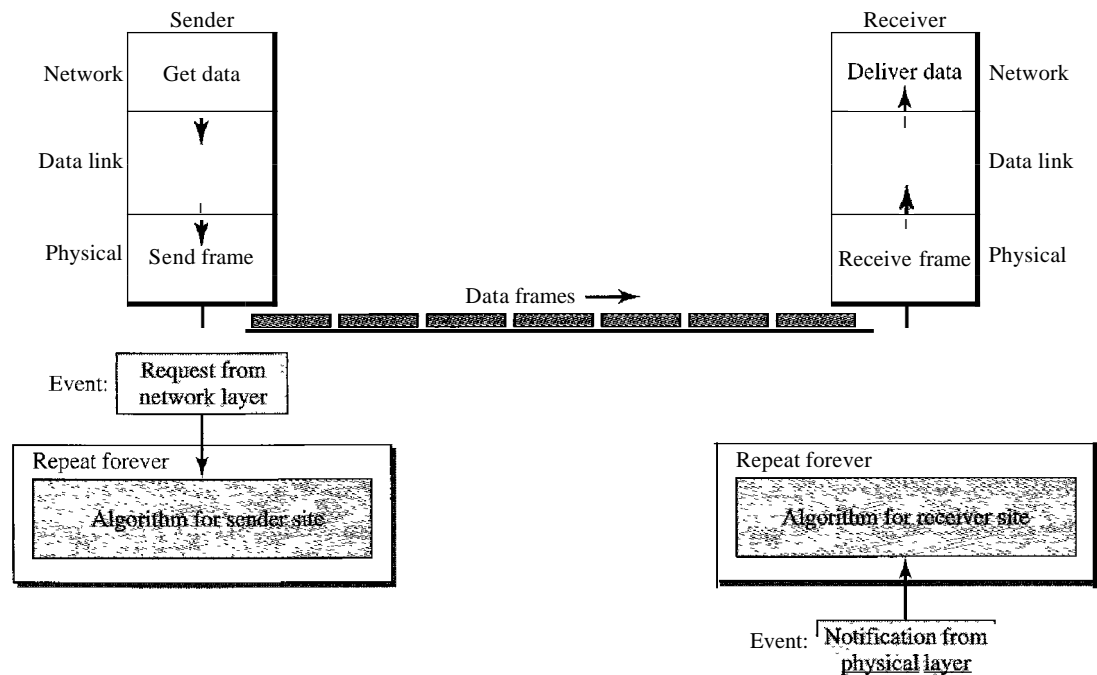
Our first protocol, which we call the Simplest Protocol for lack of any other name, is one that has no flow or error control. Like other protocols we will discuss in this chapter, it is a unidirectional protocol in which data frames are traveling in only one direction—from the

sender to receiver. We assume that the receiver can immediately handle any frame it receives with a processing time that is small enough to be negligible. The data link layer of the receiver immediately removes the header from the frame and hands the data packet to its network layer, which can also accept the packet immediately. In other words, the receiver can never be overwhelmed with incoming frames.

Design

There is no need for flow control in this scheme. The data link layer at the sender site gets data from its network layer, makes a frame out of the data, and sends it. The data link layer at the receiver site receives a frame from its physical layer, extracts data from the frame, and delivers the data to its network layer. The data link layers of the sender and receiver provide transmission services for their network layers. The data link layers use the services provided by their physical layers (such as signaling, multiplexing, and so on) for the physical transmission of bits. Figure 11.6 shows a design.

Figure 11.6 The design of the simplest protocol with no flow or error control



We need to elaborate on the procedure used by both data link layers. The sender site cannot send a frame until its network layer has a data packet to send. The receiver site cannot deliver a data packet to its network layer until a frame arrives. If the protocol is implemented as a procedure, we need to introduce the idea of events in the protocol. The procedure at the sender site is constantly running; there is no action until there is a request from the network layer. The procedure at the receiver site is also constantly running, but there is no action until notification from the physical layer arrives. Both procedures are constantly running because they do not know when the corresponding events will occur.

Algorithms

Algorithm 11.1 shows the procedure at the sender site.

Algorithm 11.1 *Sender-site algorithm for the simplest protocol*

1	while (true)	// Repeat forever
2	{	
3	WaitForEvent()	// Sleep until an event occurs
4	if(Event(RequestToSend))	// There is a packet to send
5	{	
6	GetData()	
7	MakeFrame()	
8	SendFrame()	// Send the frame
9	}	
10	}	

Analysis The algorithm has an infinite loop, which means lines 3 to 9 are repeated forever once the program starts. The algorithm is an event-driven one, which means that it *sleeps* (line 3) until an event *wakes it up* (line 4). This means that there may be an undefined span of time between the execution of line 3 and line 4; there is a gap between these actions. When the event, a request from the network layer, occurs, lines 6 through 8 are executed. The program then repeats the loop and again sleeps at line 3 until the next occurrence of the event. We have written pseudocode for the main process. We do not show any details for the modules GetData, MakeFrame, and SendFrame. GetDataO takes a data packet from the network layer, MakeFrameO adds a header and delimiter flags to the data packet to make a frame, and SendFrameO delivers the frame to the physical layer for transmission.

Algorithm 11.2 shows the procedure at the receiver site.

Algorithm 11.2 *Receiver-site algorithm for the simplest protocol*

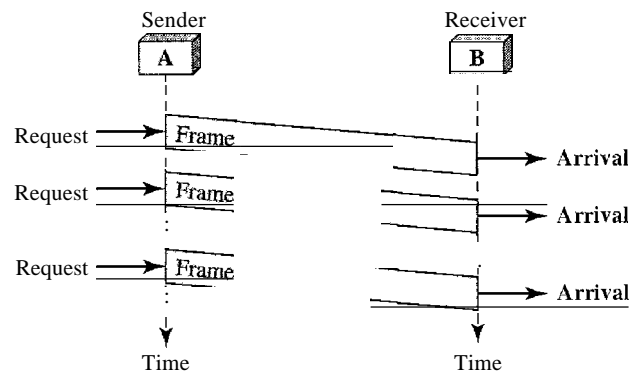
1	while(true)	// Repeat forever
2	{	
3	WaitForEvent()	// Sleep until an event occurs
4	if(Event(ArrivalNotification))	// Data frame arrived
5	{	
6	ReceiveFrame()	
7	ExtractData()	
8	DeliverData()	// Deliver data to network layer
9	}	
10	}	

Analysis This algorithm has the same format as Algorithm 11.1, except that the direction of the frames and data is upward. The event here is the arrival of a data frame. After the event occurs, the data link layer receives the frame from the physical layer using the ReceiveFrameO process, extracts the data from the frame using the ExtractDataO process, and delivers the data to the network layer using the DeliverDataO process. Here, we also have an event-driven algorithm because the algorithm never knows when the data frame will arrive.

Example 11.1

Figure 11.7 shows an example of communication using this protocol. It is very simple. The sender sends a sequence of frames without even thinking about the receiver. To send three frames, three events occur at the sender site and three events at the receiver site. Note that the data frames are shown by tilted boxes; the height of the box defines the transmission time difference between the first bit and the last bit in the frame.

Figure 11.7 Flow diagram for Example 11.1



Stop-and-Wait Protocol

If data frames arrive at the receiver site faster than they can be processed, the frames must be stored until their use. Normally, the receiver does not have enough storage space, especially if it is receiving data from many sources. This may result in either the discarding of frames or denial of service. To prevent the receiver from becoming overwhelmed with frames, we somehow need to tell the sender to slow down. There must be feedback from the receiver to the sender.

The protocol we discuss now is called the Stop-and-Wait Protocol because the sender sends one frame, stops until it receives confirmation from the receiver (okay to go ahead), and then sends the next frame. We still have unidirectional communication for data frames, but auxiliary ACK frames (simple tokens of acknowledgment) travel from the other direction. We add flow control to our previous protocol.

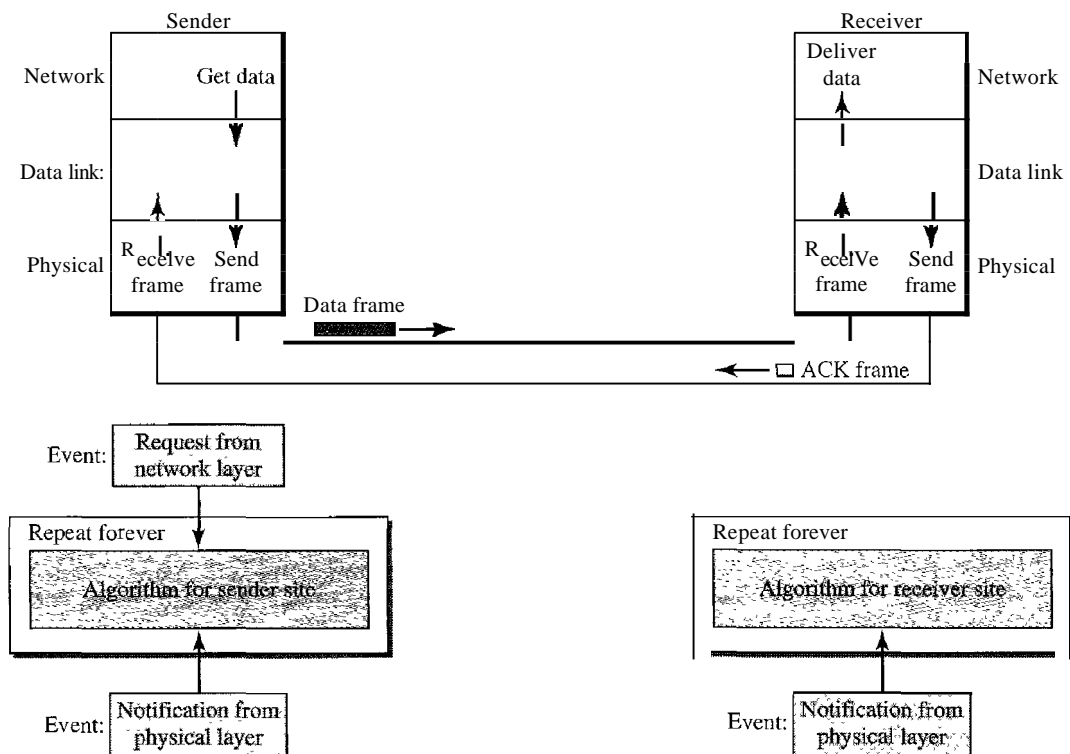
Design

Figure 11.8 illustrates the mechanism. Comparing this figure with Figure 11.6, we can see the traffic on the forward channel (from sender to receiver) and the reverse channel. At any time, there is either one data frame on the forward channel or one ACK frame on the reverse channel. We therefore need a half-duplex link.

Algorithms

Algorithm 11.3 is for the sender site.

Figure 11.8 Design of Stop-and-Wait Protocol



Algorithm 11.3 Sender-site algorithm for Stop-and-Wait Protocol

<pre> 1 while (true) 2 canSend = true 3 { 4 WaitForEvent() 5 if (Event (RequestToSend) AND canSend) 6 { 7 GetData() 8 MakeFrame() 9 SendFrame() 10 canSend = false; 11 } 12 WaitForEvent() 13 if (Event (ArrivalNotification)) 14 { 15 ReceiveFrame(); 16 canSend = true; 17 } 18 } </pre>	<pre> //Repeat forever //Allow the first frame to go // Sleep until an event occurs //Send the data frame //cannot send until ACK arrives // Sleep until an event occurs // An ACK has arrived //Receive the ACK frame </pre>
--	---

Analysis Here two events can occur: a request from the network layer or an arrival notification from the physical layer. The responses to these events must alternate. In other words, after a frame is sent, the algorithm must ignore another network layer request until that frame is

acknowledged. We know that two arrival events cannot happen one after another because the channel is error-free and does not duplicate the frames. The requests from the network layer, however, may happen one after another without an arrival event in between. We need somehow to prevent the immediate sending of the data frame. Although there are several methods, we have used a simple *canSend* variable that can either be true or false. When a frame is sent, the variable is set to false to indicate that a new network request cannot be sent until *canSend* is true. When an ACK is received, *canSend* is set to true to allow the sending of the next frame.

Algorithm 11.4 shows the procedure at the receiver site.

Algorithm 11.4 Receiver-site algorithm for Stop-and-Wait Protocol

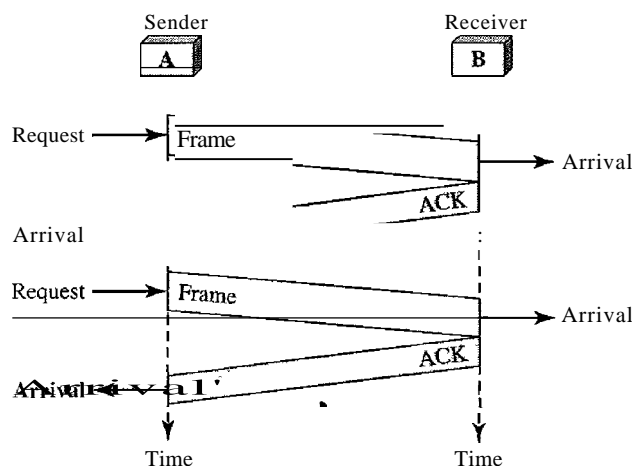
1	while (true)	II Repeat forever
2	{	
3	WaitForEvent();	II Sleep until an event occurs
4	if(Event(ArrivalNotification))	II Data frame arrives
5	{	
6	ReceiveFrame();	
7	ExtractData();	
8	Deliver(data);	II Deliver data to network layer
9	SendFrame();	II Send an ACK frame
10	}	
11	}	

Analysis This is very similar to Algorithm 11.2 with one exception. After the data frame arrives, the receiver sends an ACK frame (line 9) to acknowledge the receipt and allow the sender to send the next frame.

Example 11.2

Figure 11.9 shows an example of communication using this protocol. It is still very simple. The sender sends one frame and waits for feedback from the receiver. When the ACK arrives, the sender sends the next frame. Note that sending two frames in the protocol involves the sender in four events and the receiver in two events.

Figure 11.9 Flow diagram for Example 11.2



11.5 NOISY CHANNELS

Although the Stop-and-Wait Protocol gives us an idea of how to add flow control to its predecessor, noiseless channels are nonexistent. We can ignore the error (as we sometimes do), or we need to add error control to our protocols. We discuss three protocols in this section that use error control.

Stop-and-Wait Automatic Repeat Request

Our first protocol, called the Stop-and-Wait Automatic Repeat Request (Stop-and-Wait ARQ), adds a simple error control mechanism to the Stop-and-Wait Protocol. Let us see how this protocol detects and corrects errors.

To detect and correct corrupted frames, we need to add redundancy bits to our data frame (see Chapter 10). When the frame arrives at the receiver site, it is checked and if it is corrupted, it is silently discarded. The detection of errors in this protocol is manifested by the silence of the receiver.

Lost frames are more difficult to handle than corrupted ones. In our previous protocols, there was no way to identify a frame. The received frame could be the correct one, or a duplicate, or a frame out of order. The solution is to number the frames. When the receiver receives a data frame that is out of order, this means that frames were either lost or duplicated.

The corrupted and lost frames need to be resent in this protocol. If the receiver does not respond when there is an error, how can the sender know which frame to resend? To remedy this problem, the sender keeps a copy of the sent frame. At the same time, it starts a timer. If the timer expires and there is no ACK for the sent frame, the frame is resent, the copy is held, and the timer is restarted. Since the protocol uses the stop-and-wait mechanism, there is only one specific frame that needs an ACK even though several copies of the same frame can be in the network.

Error correction in Stop-and-Wait ARQ is done by keeping a copy of the sent frame and retransmitting of the frame when the timer expires.

Since an ACK frame can also be corrupted and lost, it too needs redundancy bits and a sequence number. The ACK frame for this protocol has a sequence number field. In this protocol, the sender simply discards a corrupted ACK frame or ignores an out-of-order one.

Sequence Numbers

As we discussed, the protocol specifies that frames need to be numbered. This is done by using sequence numbers. A field is added to the data frame to hold the sequence number of that frame.

One important consideration is the range of the sequence numbers. Since we want to minimize the frame size, we look for the smallest range that provides unambiguous

communication. The sequence numbers of course can wrap around. For example, if we decide that the field is m bits long, the sequence numbers start from 0, go to $2^m - 1$, and then are repeated.

Let us reason out the range of sequence numbers we need. Assume we have used x as a sequence number; we only need to use $x + 1$ after that. There is no need for $x + 2$. To show this, assume that the sender has sent the frame numbered x . Three things can happen.

1. The frame arrives safe and sound at the receiver site; the receiver sends an acknowledgment. The acknowledgment arrives at the sender site, causing the sender to send the next frame numbered $x + 1$.
2. The frame arrives safe and sound at the receiver site; the receiver sends an acknowledgment, but the acknowledgment is corrupted or lost. The sender resends the frame (numbered x) after the time-out. Note that the frame here is a duplicate. The receiver can recognize this fact because it expects frame $x + 1$ but frame x was received.
3. The frame is corrupted or never arrives at the receiver site; the sender resends the frame (numbered x) after the time-out.

We can see that there is a need for sequence numbers x and $x + 1$ because the receiver needs to distinguish between case 1 and case 2. But there is no need for a frame to be numbered $x + 2$. In case 1, the frame can be numbered x again because frames x and $x + 1$ are acknowledged and there is no ambiguity at either site. In cases 2 and 3, the new frame is $x + 1$, not $x + 2$. If only x and $x + 1$ are needed, we can let $x = 0$ and $x + 1 = 1$. This means that the sequence is 0, 1, 0, 1, 0, and so on. Is this pattern familiar? This is modulo-2 arithmetic as we saw in Chapter 10.

In Stop-and-Wait **ARQ**, we use sequence numbers to number the frames.
The sequence numbers are based on modulo-2 arithmetic.

Acknowledgment Numbers

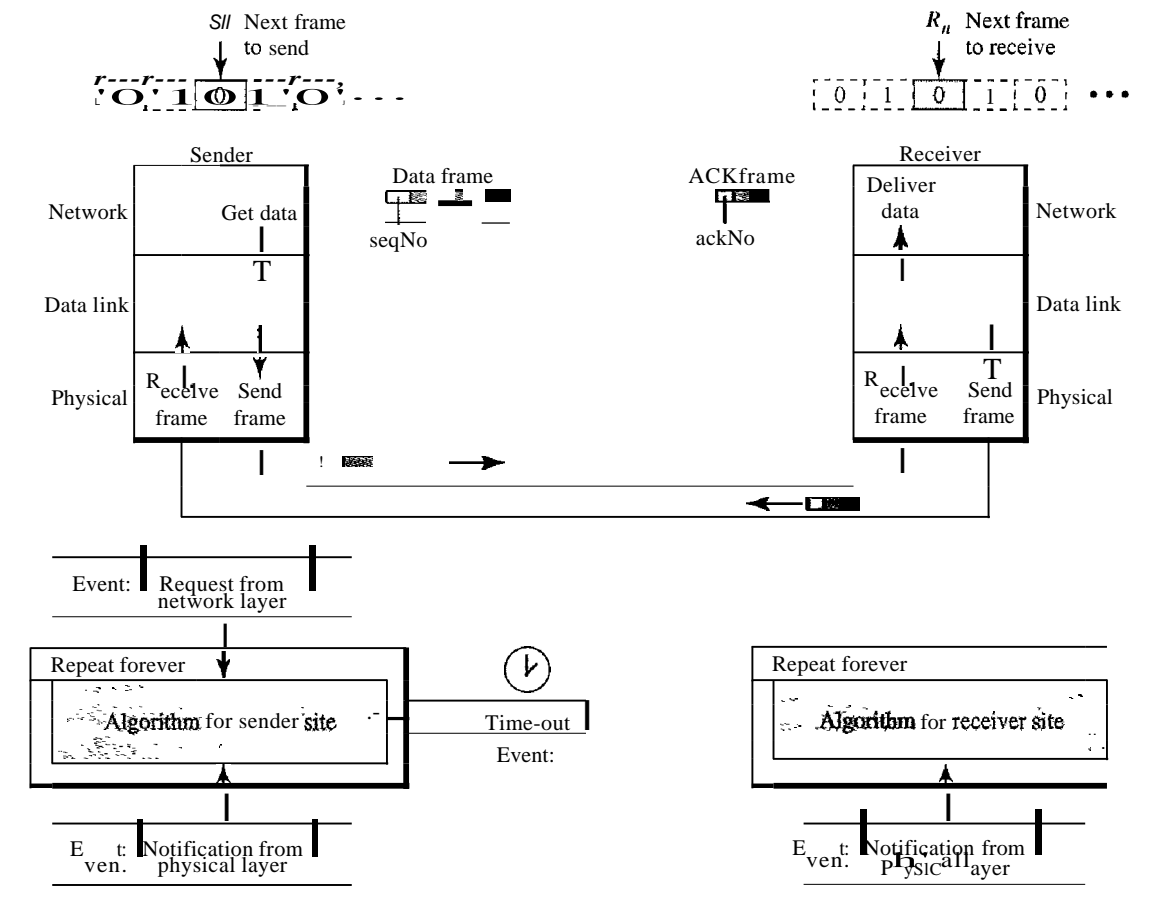
Since the sequence numbers must be suitable for both data frames and ACK frames, we use this convention: The acknowledgment numbers always announce the sequence number of the next frame expected by the receiver. For example, if frame 0 has arrived safe and sound, the receiver sends an ACK frame with acknowledgment 1 (meaning frame 1 is expected next). If frame 1 has arrived safe and sound, the receiver sends an ACK frame with acknowledgment 0 (meaning frame 0 is expected).

In Stop-and-Wait **ARQ**, the acknowledgment number always announces in modulo-2 arithmetic the sequence number of the next frame expected.

Design

Figure 11.10 shows the design of the Stop-and-Wait ARQ Protocol. The sending device keeps a copy of the last frame transmitted until it receives an acknowledgment for that frame. A data frame uses a seqNo (sequence number); an ACK frame uses an ackNo (acknowledgment number). The sender has a control variable, which we call Sn (sender, next frame to send), that holds the sequence number for the next frame to be sent (0 or 1).

Figure 11.10 Design of the Stop-and-Wait ARQ Protocol



The receiver has a control variable, which we call R_n (receiver, next frame expected), that holds the number of the next frame expected. When a frame is sent, the value of S_n is incremented (modulo-2), which means if it is 0, it becomes 1 and vice versa. When a frame is received, the value of R_n is incremented (modulo-2), which means if it is 0, it becomes 1 and vice versa. Three events can happen at the sender site; one event can happen at the receiver site. Variable S_n points to the slot that matches the sequence number of the frame that has been sent, but not acknowledged; R_n points to the slot that matches the sequence number of the expected frame.

Algorithms

Algorithm 11.5 is for the sender site.

Algorithm 11.5 Sender-site algorithm for Stop-and-Wait ARQ

1	$n = 0;$	// Frame 0 should be sent first
2	$\text{anSend} = \text{true};$	// Allow the first request to go
3	while (true)	// Repeat forever
4	{	
5	$\text{WaitForEvent}();$	// Sleep until an event occurs

Algorithm 11.5 Sender-site algorithm for Stop-and-Wait ARQ (continued)

```

6  if (Event (RequestToSend) AND canSend)
7  {
8      GetData() ;
9      MakeFrame(Sn) ;           //The seqNo is Sn
10     StoreFrame(Sn);           //Keep copy
11     SendFrame(Sn) ;
12     StartTimerO;
13     Sn = Sn + 1;
14     canSend = false;
15 }
16 WaitForEvent();               // Sleep
17 if (Event (ArrivalNotification) // An ACK has arrived
18 {
19     ReceiveFrame(ackNo);       //Receive the ACK frame
20     if(not corrupted AND ackNo == Sn) //Valid ACK
21     {
22         Stoptimer{ };
23         PurgeFrame(Sn_1);       //Copy is not needed
24         canSend = true;
25     }
26 }
27
28 if (Event (TimeOut))           // The timer expired
29 {
30     StartTimer();
31     ResendFrame(Sn_1);         //Resend a copy check
32 }
33 }

```

Analysis We first notice the presence of Sn' the sequence number of the next frame to be sent. This variable is initialized once (line 1), but it is incremented every time a frame is sent (line 13) in preparation for the next frame. However, since this is modulo-2 arithmetic, the sequence numbers are 0, 1, 0, 1, and so on. Note that the processes in the first event (SendFrame, StoreFrame, and PurgeFrame) use an Sn defining the frame sent out. We need at least one buffer to hold this frame until we are sure that it is received safe and sound. Line 10 shows that before the frame is sent, it is stored. The copy is used for resending a corrupt or lost frame. We are still using the canSend variable to prevent the network layer from making a request before the previous frame is received safe and sound. If the frame is not corrupted and the ackNo of the ACK frame matches the sequence number of the next frame to send, we stop the timer and purge the copy of the data frame we saved. Otherwise, we just ignore this event and wait for the next event to happen. After each frame is sent, a timer is started. When the timer expires (line 28), the frame is resent and the timer is restarted.

Algorithm 11.6 shows the procedure at the receiver site.

Algorithm 11.6 Receiver-site algorithm for Stop-and-Wait ARQ Protocol

```

1  = 0;                          // Frame 0 expected to arrive first
2  while (true)
3  {
4      WaitForEvent();           // Sleep until an event occurs

```

Algorithm 11.6 Receiver-site algorithm for Stop-and-Wait ARQ Protocol (continued)

5	if(Event(ArrivalNotification»	//Data frame arrives
6	{	
7	ReceiveFrame()	
8	if(corrupted(frame»i	
9	sleep() i	
10	if(seqNo == R _n)	//Valid data frame
11	{	
12	ExtractData();	
13	DeliverData()	//Deliver data
14	R _n = R _n + 1;	
15	}	
16	SendFrame(R _n);	//Send an ACK
17	}	
18	}	

Analysis This is noticeably different from Algorithm 11.4. First, all arrived data frames that are corrupted are ignored. If the seqNo of the frame is the one that is expected (R_n), the frame is accepted, the data are delivered to the network layer, and the value of R_n is incremented. However, there is one subtle point here. Even if the sequence number of the data frame does not match the next frame expected, an ACK is sent to the sender. This ACK, however, just reconfirms the previous ACK instead of confirming the frame received. This is done because the receiver assumes that the previous ACK might have been lost; the receiver is sending a duplicate frame. The resent ACK may solve the problem before the time-out does it.

Example 11.3

Figure 11.11 shows an example of Stop-and-Wait ARQ. Frame 2 is sent and acknowledged. Frame 1 is lost and resent after the time-out. The resent frame 1 is acknowledged and the timer stops. Frame 2 is sent and acknowledged, but the acknowledgment is lost. The sender has no idea if the frame or the acknowledgment is lost, so after the time-out, it resends frame 0, which is acknowledged.

Efficiency

The Stop-and-Wait ARQ discussed in the previous section is very inefficient if our channel is *thick* and *long*. By *thick*, we mean that our channel has a large bandwidth; by *long*, we mean the round-trip delay is long. The product of these two is called the bandwidth-delay product, as we discussed in Chapter 3. We can think of the channel as a pipe. The bandwidth-delay product then is the volume of the pipe in bits. The pipe is always there. If we do not use it, we are inefficient. The bandwidth-delay product is a measure of the number of bits we can send out of our system while waiting for news from the receiver.

Example 11.4

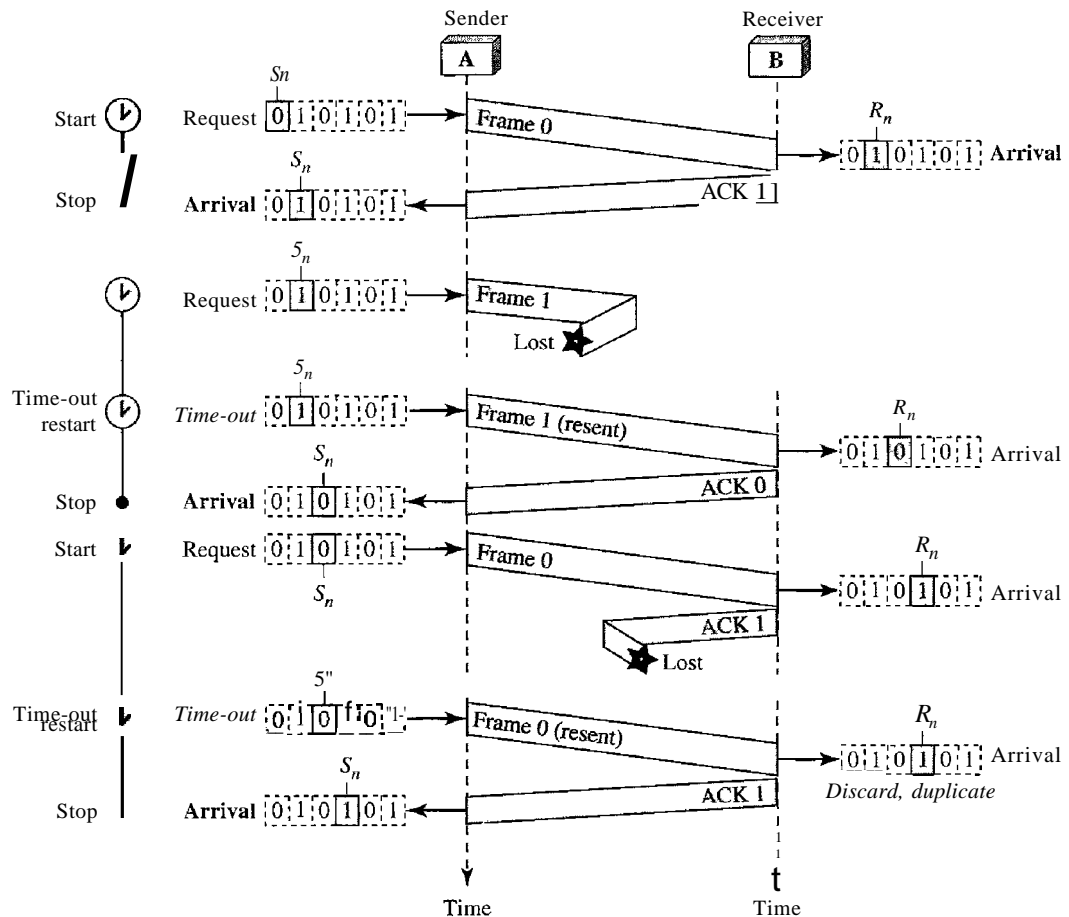
Assume that, in a Stop-and-Wait ARQ system, the bandwidth of the line is 1 Mbps, and 1 bit takes 20 ms to make a round trip. What is the bandwidth-delay product? If the system data frames are 1000 bits in length, what is the utilization percentage of the link?

Solution

The bandwidth-delay product is

$$(1 \times 10^6) \times (20 \times 10^{-3}) = 20,000 \text{ bits}$$

Figure 11.11 Flow diagram for Example 11.3



The system can send 20,000 bits during the time it takes for the data to go from the sender to the receiver and then back again. However, the system sends only 1000 bits. We can say that the link utilization is only $1000/20,000$, or 5 percent. For this reason, for a link with a high bandwidth or long delay, the use of Stop-and-Wait ARQ wastes the capacity of the link.

Example 11.5

What is the utilization percentage of the link in Example 11.4 if we have a protocol that can send up to 15 frames before stopping and worrying about the acknowledgments?

Solution

The bandwidth-delay product is still 20,000 bits. The system can send up to 15 frames or 15,000 bits during a round trip. This means the utilization is $15,000/20,000$, or 75 percent. Of course, if there are damaged frames, the utilization percentage is much less because frames have to be resent.

Pipelining

In networking and in other areas, a task is often begun before the previous task has ended. This is known as pipelining. There is no pipelining in Stop-and-Wait ARQ because we need to wait for a frame to reach the destination and be acknowledged before the next frame can be sent. However, pipelining does apply to our next two protocols because

several frames can be sent before we receive news about the previous frames. Pipelining improves the efficiency of the transmission if the number of bits in transition is large with respect to the bandwidth-delay product.

Go-Back-N Automatic Repeat Request

To improve the efficiency of transmission (filling the pipe), multiple frames must be in transition while waiting for acknowledgment. In other words, we need to let more than one frame be outstanding to keep the channel busy while the sender is waiting for acknowledgment. In this section, we discuss one protocol that can achieve this goal; in the next section, we discuss a second.

The first is called Go-Back-N Automatic Repeat Request (the rationale for the name will become clear later). In this protocol we can send several frames before receiving acknowledgments; we keep a copy of these frames until the acknowledgments arrive.

Sequence Numbers

Frames from a sending station are numbered sequentially. However, because we need to include the sequence number of each frame in the header, we need to set a limit. If the header of the frame allows m bits for the sequence number, the sequence numbers range from 0 to $2^m - 1$. For example, if m is 4, the only sequence numbers are 0 through 15 inclusive. However, we can repeat the sequence. So the sequence numbers are

0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, ...

In other words, the sequence numbers are modulo- 2^m .

In the Go-Back-N Protocol, the sequence numbers are modulo 2^m ,
where m is the size of the sequence number field in bits.

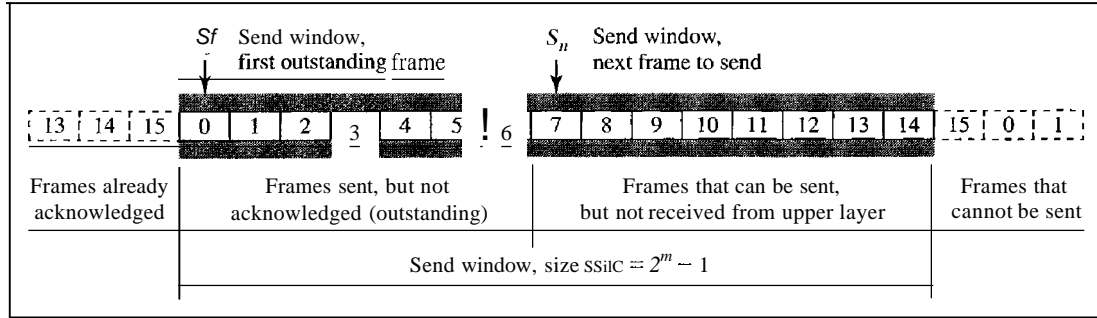
Sliding Window

In this protocol (and the next), the sliding window is an abstract concept that defines the range of sequence numbers that is the concern of the sender and receiver. In other words, the sender and receiver need to deal with only part of the possible sequence numbers. The range which is the concern of the sender is called the send sliding window; the range that is the concern of the receiver is called the receive sliding window. We discuss both here.

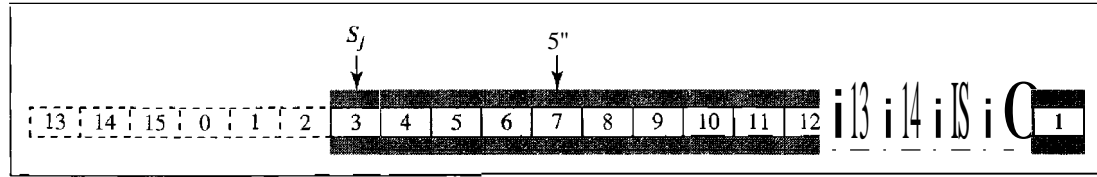
The send window is an imaginary box covering the sequence numbers of the data frames which can be in transit. In each window position, some of these sequence numbers define the frames that have been sent; others define those that can be sent. The maximum size of the window is $2^m - 1$ for reasons that we discuss later. In this chapter, we let the size be fixed and set to the maximum value, but we will see in future chapters that some protocols may have a variable window size. Figure 11.12 shows a sliding window of size 15 ($m=4$).

The window at any time divides the possible sequence numbers into four regions. The first region, from the far left to the left wall of the window, defines the sequence

Figure 11.12 Send window for Go-Back-NARQ



a. Send window before sliding



b. Send window after sliding

numbers belonging to frames that are already acknowledged. The sender does not worry about these frames and keeps no copies of them. The second region, colored in Figure 11.12a, defines the range of sequence numbers belonging to the frames that are sent and have an unknown status. The sender needs to wait to find out if these frames have been received or were lost. We call these outstanding frames. The third range, white in the figure, defines the range of sequence numbers for frames that can be sent; however, the corresponding data packets have not yet been received from the network layer. Finally, the fourth region defines sequence numbers that cannot be used until the window slides, as we see next.

The window itself is an abstraction; three variables define its size and location at any time. We call these variables Sf (send window, the first outstanding frame), Sn (send window, the next frame to be sent), and $Ssize$ (send window, size). The variable Sf defines the sequence number of the first (oldest) outstanding frame. The variable Sn holds the sequence number that will be assigned to the next frame to be sent. Finally, the variable $Ssize$ defines the size of the window, which is fixed in our protocol.

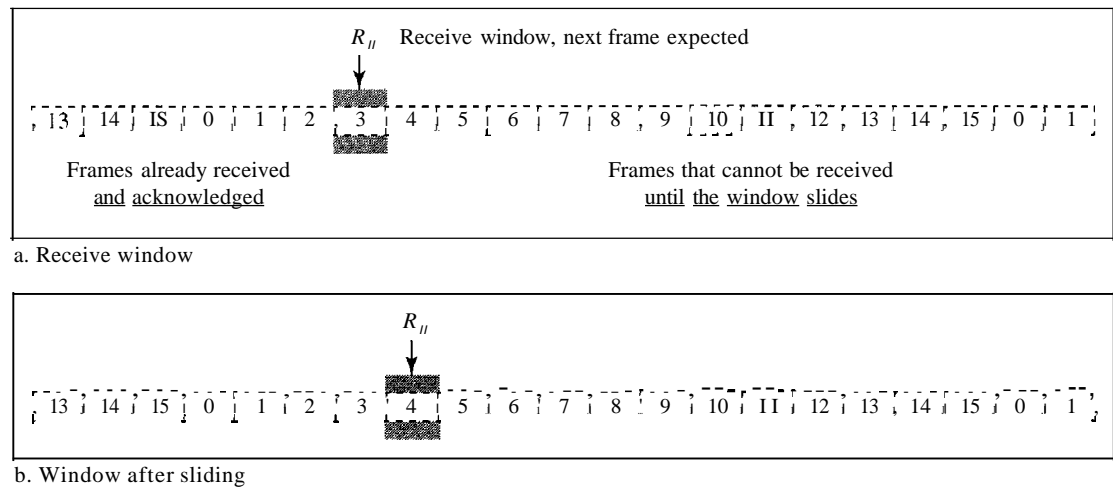
The send window is an abstract concept defining an imaginary
box of size $2^m - 1$ with three variables: Sf , Sn , and $Ssize$

Figure 11.12b shows how a send window can slide one or more slots to the right when an acknowledgment arrives from the other end. As we will see shortly, the acknowledgments in this protocol are cumulative, meaning that more than one frame can be acknowledged by an ACK frame. In Figure 11.12b, frames 0, 1, and 2 are acknowledged, so the window has slid to the right three slots. Note that the value of Sf is 3 because frame 3 is now the first outstanding frame.

The send window can slide one or more slots when a valid acknowledgment arrives.

The receive window makes sure that the correct data frames are received and that the correct acknowledgments are sent. The size of the receive window is always 1. The receiver is always looking for the arrival of a specific frame. Any frame arriving out of order is discarded and needs to be resent. Figure 11.13 shows the receive window.

Figure 11.13 Receive window for Go-Back-NARQ



The receive window is an abstract concept defining an imaginary box of size 1 with one single variable R_n . The window slides when a correct frame has arrived; sliding occurs one slot at a time.

Note that we need only one variable R_n (receive window, next frame expected) to define this abstraction. The sequence numbers to the left of the window belong to the frames already received and acknowledged; the sequence numbers to the right of this window define the frames that cannot be received. Any received frame with a sequence number in these two regions is discarded. Only a frame with a sequence number matching the value of R_n is accepted and acknowledged.

The receive window also slides, but only one slot at a time. When a correct frame is received (and a frame is received only one at a time), the window slides.

Timers

Although there can be a timer for each frame that is sent, in our protocol we use only one. The reason is that the timer for the first outstanding frame always expires first; we send all outstanding frames when this timer expires.

Acknowledgment

The receiver sends a positive acknowledgment if a frame has arrived safe and sound and in order. If a frame is damaged or is received out of order, the receiver is silent and will discard all subsequent frames until it receives the one it is expecting. The silence of

the receiver causes the timer of the unacknowledged frame at the sender site to expire. This, in turn, causes the sender to go back and resend all frames, beginning with the one with the expired timer. The receiver does not have to acknowledge each frame received. It can send one cumulative acknowledgment for several frames.

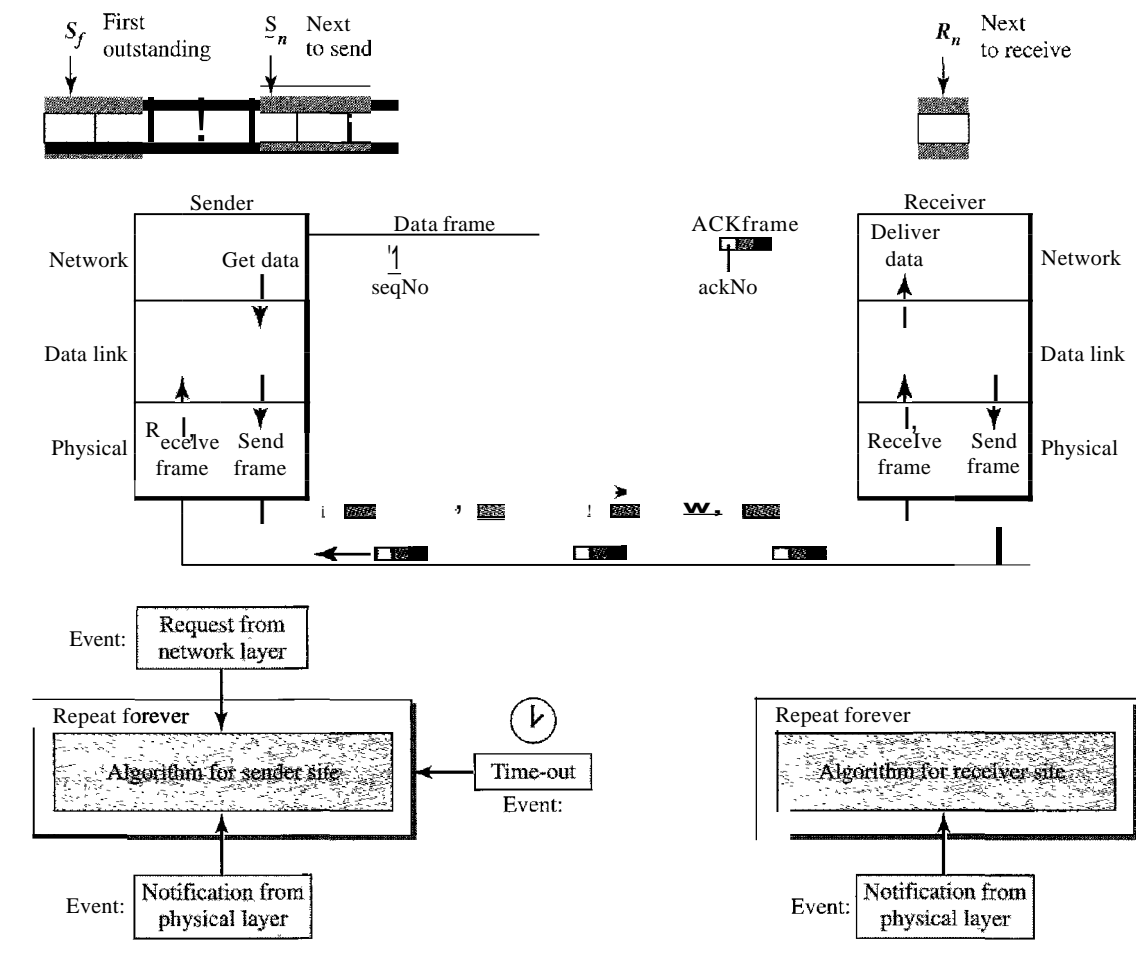
Resending a Frame

When the timer expires, the sender resends all outstanding frames. For example, suppose the sender has already sent frame 6, but the timer for frame 3 expires. This means that frame 3 has not been acknowledged; the sender goes back and sends frames 3, 4, 5, and 6 again. That is why the protocol is called *Go-Back-NARQ*.

Design

Figure 11.14 shows the design for this protocol. As we can see, multiple frames can be in transit in the forward direction, and multiple acknowledgments in the reverse direction. The idea is similar to Stop-and-Wait ARQ; the difference is that the send

Figure 11.14 Design of *Go-Back-NARQ*

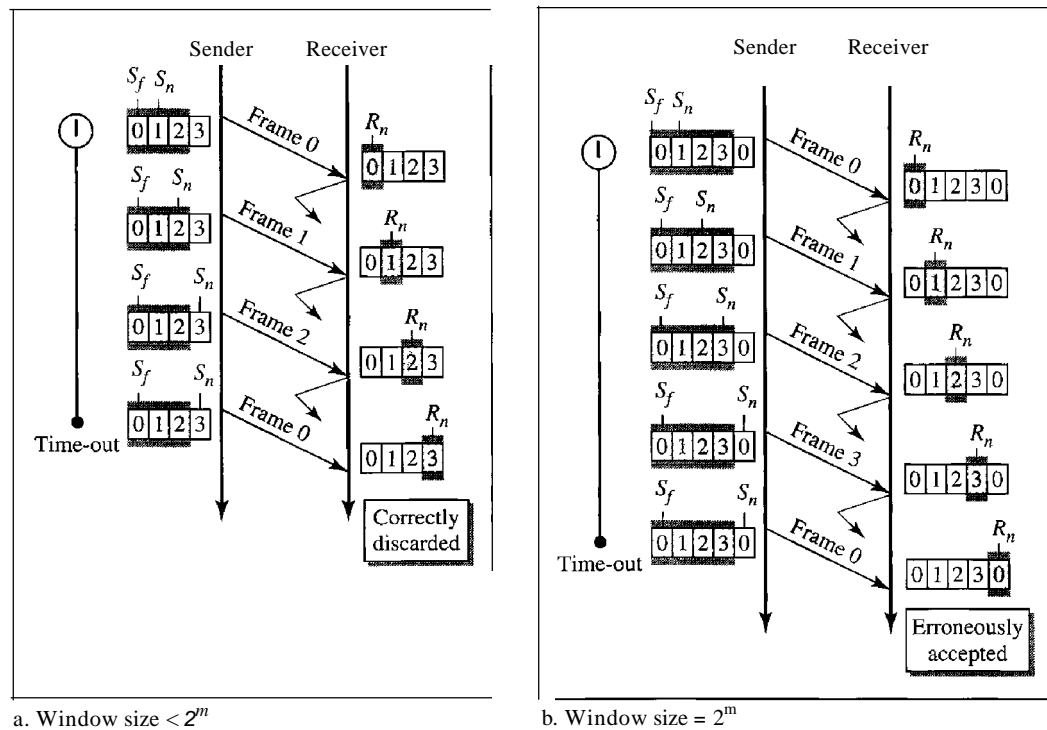


window allows us to have as many frames in transition as there are slots in the send window.

Send Window Size

We can now show why the size of the send window must be less than 2^m . As an example, we choose $m = 2$, which means the size of the window can be $2^m - 1$, or 3. Figure 11.15 compares a window size of 3 against a window size of 4. If the size of the window is 3 (less than 2^2) and all three acknowledgments are lost, the frame timer expires and all three frames are resent. The receiver is now expecting frame 3, not frame 0, so the duplicate frame is correctly discarded. On the other hand, if the size of the window is 4 (equal to 2^2) and all acknowledgments are lost, the sender will send a duplicate of frame 0. However, this time the window of the receiver expects to receive frame 0, so it accepts frame 0, not as a duplicate, but as the first frame in the next cycle. This is an error.

Figure 11.15 Window size for Go-Back-NARQ



In Go-Back-NARQ, the size of the send window must be less than 2^m ; the size of the receiver window is always 1.

Algorithms

Algorithm 11.7 shows the procedure for the sender in this protocol.

Algorithm 11.7 Go-Back-N sender algorithm

```

1 Sw =  $2^m - 1$ ;
2 Sf = 0;
3 Sn = 0;
4
5 while (true)                                //Repeat forever
6 {
7   WaitForEvent();
8   if(Event(RequestToSend»                //A packet to send
9   {
10      if(Sn-Sf >= Sw)                       IIIf window is full
11         Sleep();
12      GetData();
13      MakeFrame(Sn);
14      StoreFrame(Sn);
15      SendFrame(Sn);
16      Sn = Sn + 1;
17      if(timer not running)
18         StartTimer();
19   }
20
21   if{Event{ArrivalNotification»           IIACK arrives
22   {
23      Receive(ACK);
24      if{corrupted{ACK»
25         Sleep();
26      if{{ackNo>sf}&&{ackNO<=Sn»           IIIf a valid ACK
27      While(Sf <= ackNo)
28      {
29         PurgeFrame{Sf);
30         Sf = Sf + 1;
31      }
32      StopTimer();
33   }
34
35   if{Event{TimeOut»                        liThe timer expires
36   {
37      StartTimer();
38      Temp = Sf;
39      while{Temp < Sn);
40      {
41         SendFrame(Sf);
42         Sf = Sf + 1;
43      }
44   }
45 }

```

Analysis This algorithm first initializes three variables. Unlike Stop-and-Wait ARQ, this protocol allows several requests from the network layer without the need for other events to occur; we just need to be sure that the window is not full (line 12). In our approach, if the window is full,

the request is just ignored and the network layer needs to try again. Some implementations use other methods such as enabling or disabling the network layer. The handling of the arrival event is more complex than in the previous protocol. If we receive a corrupted ACK, we ignore it. If the ackNa belongs to one of the outstanding frames, we use a loop to purge the buffers and move the left wall to the right. The time-out event is also more complex. We first start a new timer. We then resend all outstanding frames.

Algorithm 11.8 is the procedure at the receiver site.

Algorithm 11.8 *Go-Back-N receiver algorithm*

```

1   $R_n = 0$ ;
2
3  while (true)                                II Repeat forever
4  {
5      WaitForEvent();
6
7      if(Event{ArrivalNotification}» /Data frame arrives
8      (
9          Receive(Frame);
10         if(corrupted(Frame)»
11             Sleep();
12         if(seqNo =  $R_n$ )                        III If expected frame
13         {
14             DeliverData()i                     IID Deliver data
15              $R_n = R_n + 1$ ;                     IISlide window
16             SendACK( $R_n$ );
17         }
18     }
19 }
```

Analysis This algorithm is simple. We ignore a corrupt or out-of-order frame. If a frame arrives with an expected sequence number, we deliver the data, update the value of R_n , and send an ACK with the ackNa showing the next frame expected.

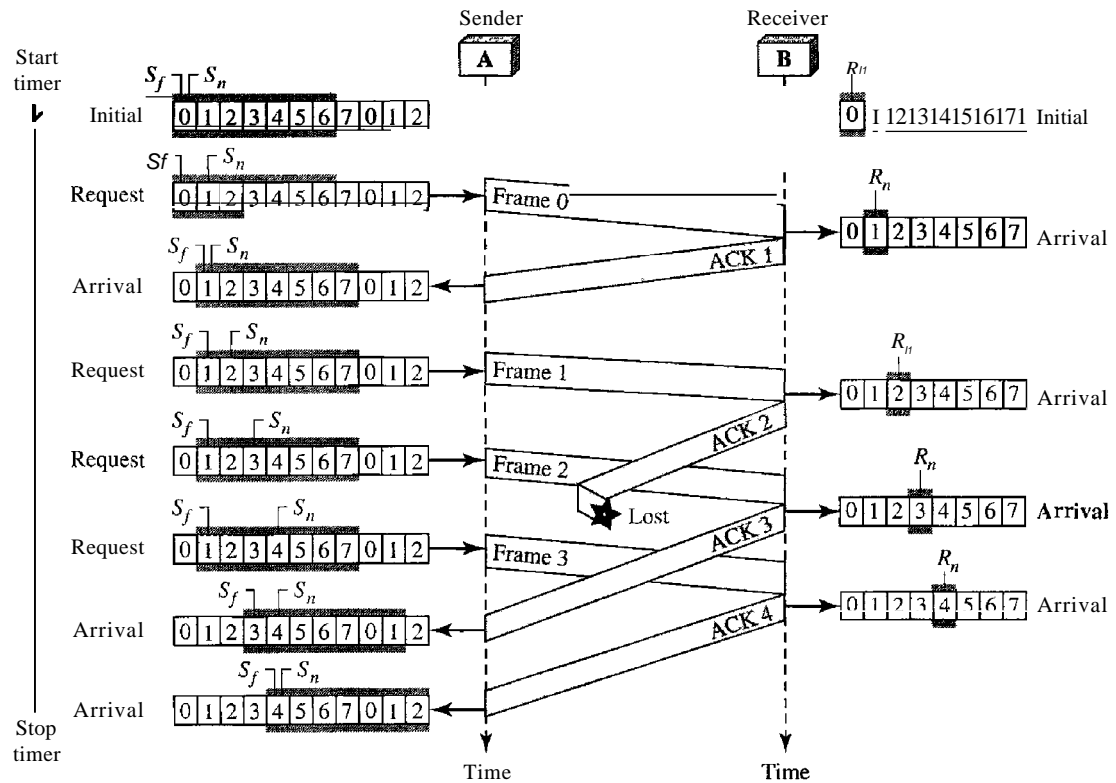
Example 11.6

Figure 11.16 shows an example of Go-Back-N. This is an example of a case where the forward channel is reliable, but the reverse is not. No data frames are lost, but some ACKs are delayed and one is lost. The example also shows how cumulative acknowledgments can help if acknowledgments are delayed or lost.

After initialization, there are seven sender events. Request events are triggered by data from the network layer; arrival events are triggered by acknowledgments from the physical layer. There is no time-out event here because all outstanding frames are acknowledged before the timer expires. Note that although ACK 2 is lost, ACK 3 serves as both ACK 2 and ACK3.

There are four receiver events, all triggered by the arrival of frames from the physical layer.

Figure 11.16 Flow diagram for Example 11.6



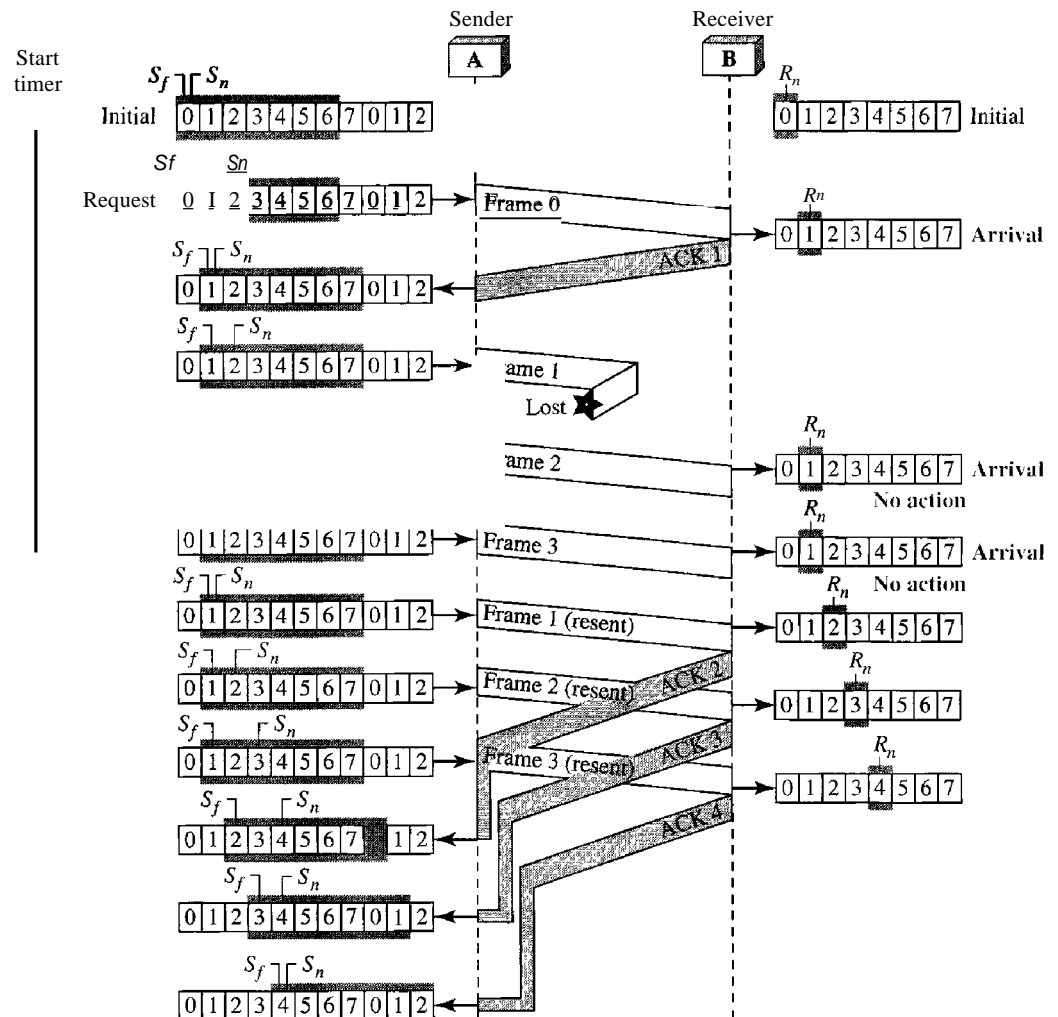
Example 11.7

Figure 11.17 shows what happens when a frame is lost. Frames 0, 1, 2, and 3 are sent. However, frame 1 is lost. The receiver receives frames 2 and 3, but they are discarded because they are received out of order (frame 1 is expected). The sender receives no acknowledgment about frames 1, 2, or 3. Its timer finally expires. The sender sends all outstanding frames (1, 2, and 3) because it does not know what is wrong. Note that the resending of frames 1, 2, and 3 is the response to one single event. When the sender is responding to this event, it cannot accept the triggering of other events. This means that when ACK 2 arrives, the sender is still busy with sending frame 3. The physical layer must wait until this event is completed and the data link layer goes back to its sleeping state. We have shown a vertical line to indicate the delay. It is the same story with ACK 3; but when ACK 3 arrives, the sender is busy responding to ACK 2. It happens again when ACK 4 arrives. Note that before the second timer expires, all outstanding frames have been sent and the timer is stopped.

Go-Back-NARQ Versus Stop-and-Wait ARQ

The reader may find that there is a similarity between *Go-Back-NARQ* and *Stop-and-Wait ARQ*. We can say that the *Stop-and-Wait ARQ* Protocol is actually a *Go-Back-NARQ* in which there are only two sequence numbers and the send window size is 1. In other words, $m = 1$, $2^m - 1 = 1$. In *Go-Back-NARQ*, we said that the addition is modulo- 2^m ; in *Stop-and-Wait ARQ* it is 2, which is the same as 2^m when $m = 1$.

Figure 11.17 Flow diagram for Example 11.7



Stop-and-WaitARQ is a special case of Go-Back-NARQ
in which the size of the send window is 1.

Selective Repeat Automatic Repeat Request

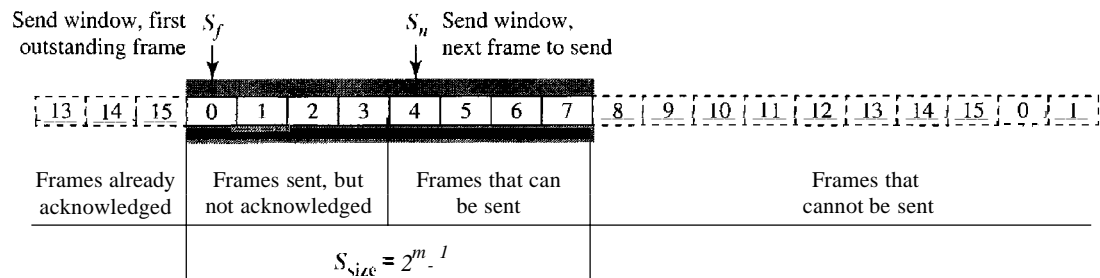
Go-Back-NARQ simplifies the process at the receiver site. The receiver keeps track of only one variable, and there is no need to buffer out-of-order frames; they are simply discarded. However, this protocol is very inefficient for a noisy link. In a noisy link a frame has a higher probability of damage, which means the resending of multiple frames. This resending uses up the bandwidth and slows down the transmission. For noisy links, there is another mechanism that does not resend N frames when just one frame is damaged; only the damaged frame is resent. This mechanism is called Selective RepeatARQ. It is more efficient for noisy links, but the processing at the receiver is more complex.

Windows

The Selective Repeat Protocol also uses two windows: a send window and a receive window. However, there are differences between the windows in this protocol and the ones in Go-Back-N. First, the size of the send window is much smaller; it is 2^{m-1} . The reason for this will be discussed later. Second, the receive window is the same size as the send window.

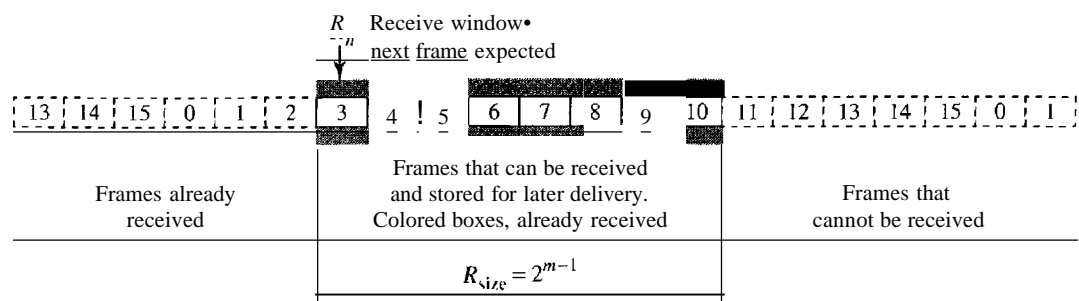
The send window maximum size can be 2^{m-1} . For example, if $m = 4$, the sequence numbers go from 0 to 15, but the size of the window is just 8 (it is 15 in the *Go-Back-N* Protocol). The smaller window size means less efficiency in filling the pipe, but the fact that there are fewer duplicate frames can compensate for this. The protocol uses the same variables as we discussed for Go-Back-N. We show the Selective Repeat send window in Figure 11.18 to emphasize the size. Compare it with Figure 11.12.

Figure 11.18 Send window for Selective Repeat ARQ



The receive window in Selective Repeat is totally different from the one in Go-Back-N. First, the size of the receive window is the same as the size of the send window (2^{m-1}). The Selective Repeat Protocol allows as many frames as the size of the receive window to arrive out of order and be kept until there is a set of in-order frames to be delivered to the network layer. Because the sizes of the send window and receive window are the same, all the frames in the send frame can arrive out of order and be stored until they can be delivered. We need, however, to mention that the receiver never delivers packets out of order to the network layer. Figure 11.19 shows the receive window in this

Figure 11.19 Receive window for Selective Repeat ARQ

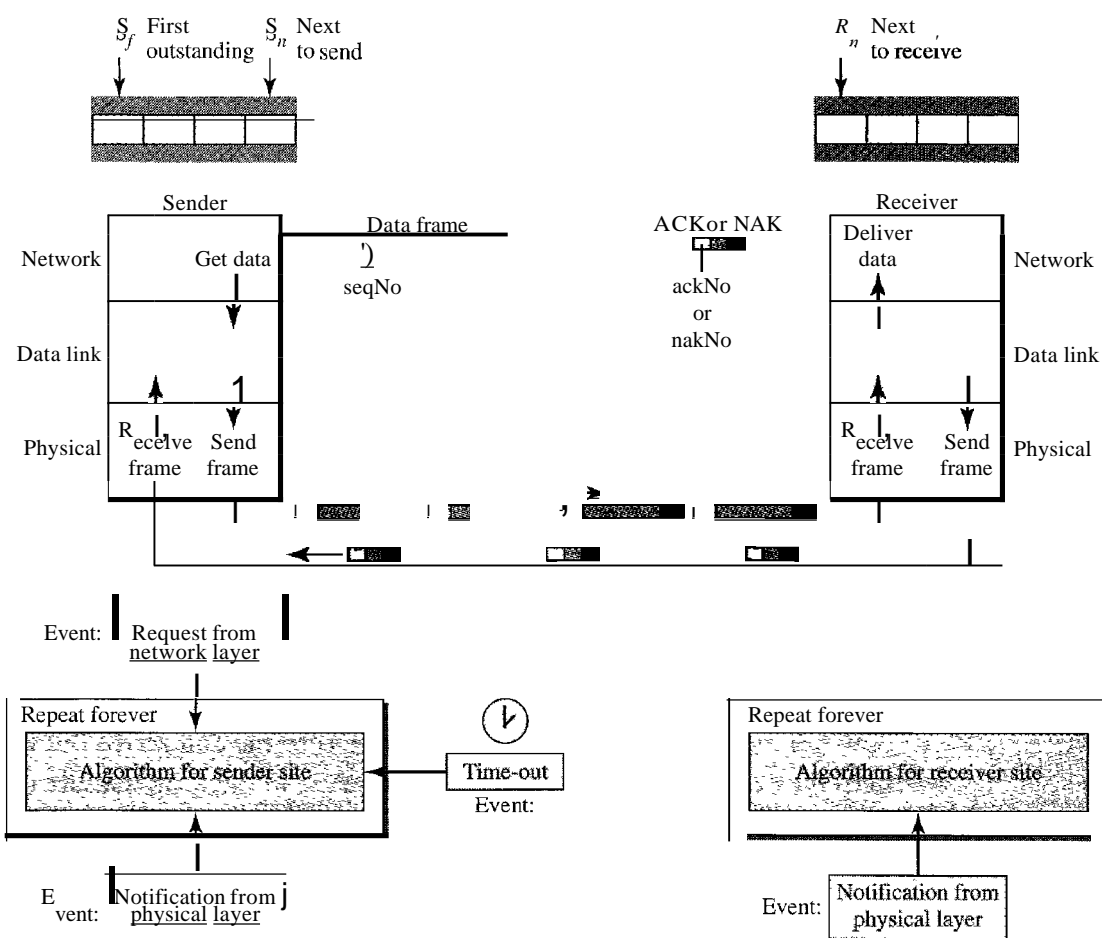


protocol. Those slots inside the window that are colored define frames that have arrived out of order and are waiting for their neighbors to arrive before delivery to the network layer.

Design

The design in this case is to some extent similar to the one we described for the 00-Back-N, but more complicated, as shown in Figure 11.20.

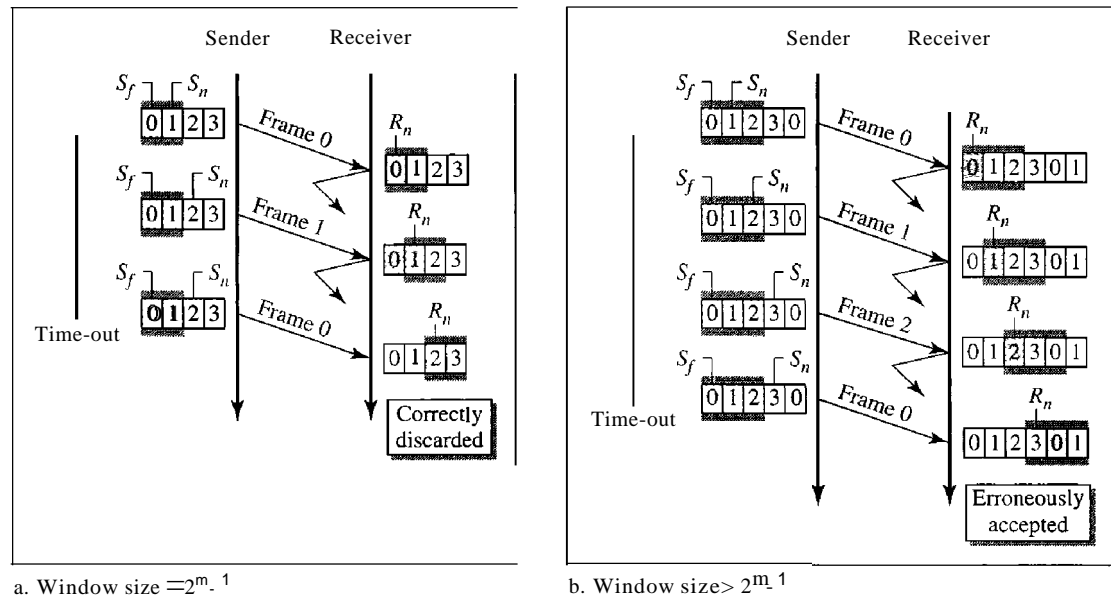
Figure 11.20 Design of Selective Repeat ARQ



Window Sizes

We can now show why the size of the sender and receiver windows must be at most one-half of 2^m . For an example, we choose $m = 2$, which means the size of the window is $2^m/2$, or 2. Figure 11.21 compares a window size of 2 with a window size of 3.

If the size of the window is 2 and all acknowledgments are lost, the timer for frame 0 expires and frame 0 is resent. However, the window of the receiver is now expecting

Figure 11.21 *Selective Repeat ARQ, window size*

frame 2, not frame 0, so this duplicate frame is correctly discarded. When the size of the window is 3 and all acknowledgments are lost, the sender sends a duplicate of frame 0. However, this time, the window of the receiver expects to receive frame 0 (0 is part of the window), so it accepts frame 0, not as a duplicate, but as the first frame in the next cycle. This is clearly an error.

In Selective Repeat ARQ, the size of the sender and receiver window must be at most one-half of 2^m .

Algorithms

Algorithm 11.9 shows the procedure for the sender.

Algorithm 11.9 *Sender-site Selective Repeat algorithm*

```

1    $W = 2^{m-1}$ 
2    $Si = 0$ 
3    $Ri = 0$ 
4
5   hile (true)                                //Repeat forever
6   {
7       WaitForEvent()
8       if(Event(RequestToSend))                //There is a packet to sen
9       {

```

Algorithm 11.9 Sender-site Selective Repeat algorithm (continued)

```

10      if{Sn-S;E >= Sw}                                I/If window is full
11          Sleep{};
12      GetData{};
13      MakeFrame(Sn);
14      StoreFrame{Sn};
15      SendFrame(Sn);
16      Sn = Sn + 1;
17      StartTimer{Sn};
18  }
19
20  if(Event{ArrivalNotification}» IIACK arrives
21  {
22      Receive{frame};                                I/Receive ACK or NAK
23      if{corrupted{frame}»
24          Sleep{};
25      if (FrameType == NAK)
26          if (nakNo between Sf and So)
27          {
28              resend{nakNo};
29              StartTimer{nakNo};
30          }
31      if (FrameType == ACK)
32          if (ackNo between Sf and So)
33          {
34              while{sf < ackNo)
35              {
36                  Purge(sf);
37                  stopTimer(Sf);
38                  Sf = Sf + 1;
39              }
40          }
41      }
42
43  if(Event{TimeOut{t}»}                                liThe timer expires
44  {
45      StartTimer{t};
46      SendFrame{t};
47  }
48  }
```

Analysis The handling of the request event is similar to that of the previous protocol except that one timer is started for each frame sent. The arrival event is more complicated here. An ACK or a NAK frame may arrive. If a valid NAK frame arrives, we just resend the corresponding frame. If a valid ACK arrives, we use a loop to purge the buffers, stop the corresponding timer, and move the left wall of the window. The time-out event is simpler here; only the frame which times out is resent.

Algorithm 11.10 shows the procedure for the receiver.

Algorithm 11.10 *Receiver-site Selective Repeat algorithm*

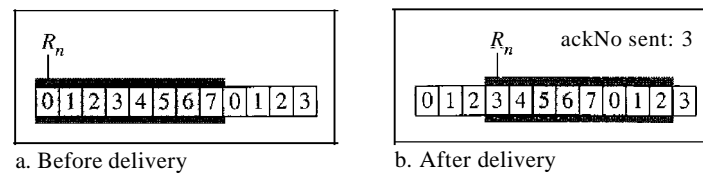
```

1   $R_n = 0$ ;
2  NakSent = false;
3  AckNeeded = false;
4  Repeat(for all slots)
5      Marked(slot) = false;
6
7  !while (true)                                IIRepeat forever
8  {
9      WaitForEvent(i)
10
11     if{Event{ArrivalNotification»           jData frame arrives
12     {
13         Receive(Frame);
14         if(corrupted(Frame)&& (NOT NakSent)
15         {
16             SendNAK( $R_n$ );
17             NakSent = true;
18             Sleep{};
19         }
20         if(seqNo <>  $R_n$ )&& (NOT NakSent)
21         {
22             SendNAK( $R_n$ );
23             NakSent = true;
24             if {(seqNo in window)&&(IMarked(seqNo»
25             {
26                 StoreFrame{seqNo)
27                 Marked(seqNo)= true;
28                 while{Marked( $R_n$ )
29                 {
30                     DeliverData( $R_n$ );
31                     Purge( $R_n$ );
32                      $R_n = R_n + 1$ ;
33                     AckNeeded = true;
34                 }
35                 if(AckNeeded);
36                 {
37                     SendAck( $R_n$ );
38                     AckNeeded = false;
39                     NakSent = false;
40                 }
41             }
42         }
43     }
44 }
```

Analysis Here we need more initialization. In order not to overwhelm the other side with NAKs, we use a variable called NakSent. To know when we need to send an ACK, we use a variable called AckNeeded. Both of these are initialized to false. We also use a set of variables to

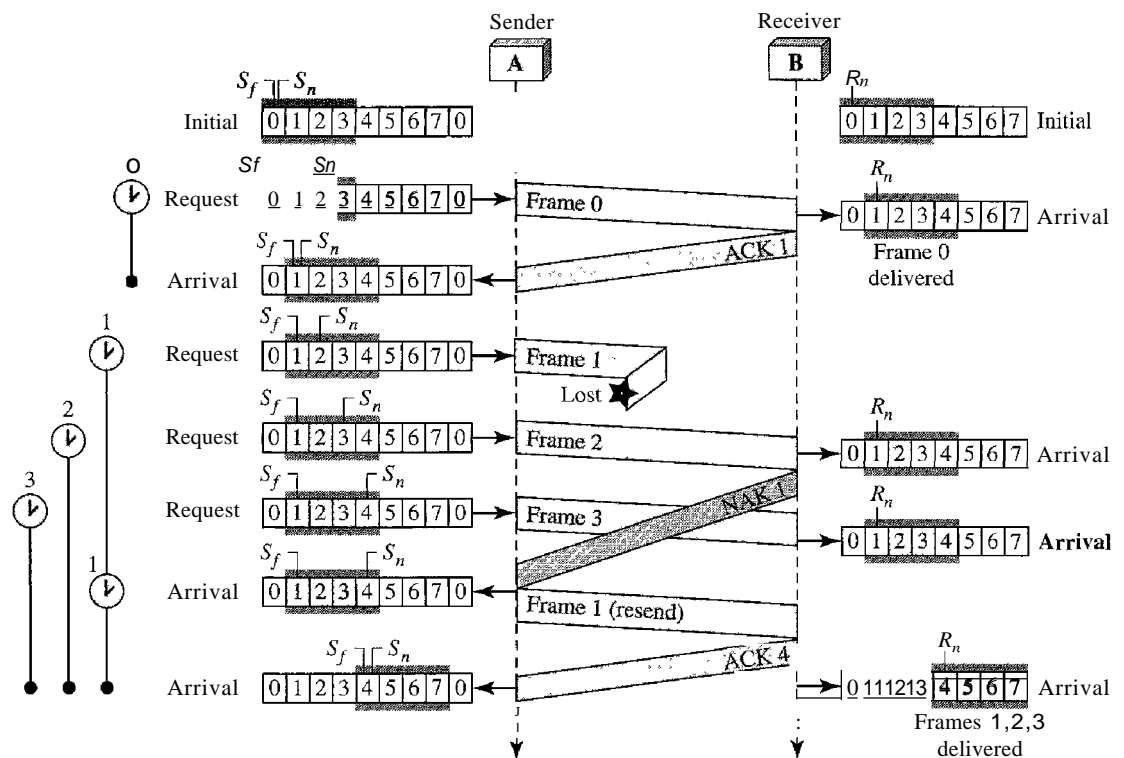
mark the slots in the receive window once the corresponding frame has arrived and is stored. If we receive a corrupted frame and a NAK has not yet been sent, we send a NAK to tell the other site that we have not received the frame we expected. **If** the frame is not corrupted and the sequence number is in the window, we store the frame and mark the slot. If contiguous frames, starting from R_n have been marked, we deliver their data to the network layer and slide the window. Figure 11.22 shows this situation.

Figure 11.22 *Delivery of data in Selective Repeat ARQ*



Example 11.8

This example is similar to Example 11.3 in which frame 1 is lost. We show how Selective Repeat behaves in this case. Figure 11.23 shows the situation.

Figure 11.23 *Flow diagram for Example 11.8*

One main difference is the number of timers. Here, each frame sent or resent needs a timer, which means that the timers need to be numbered (0, 1, 2, and 3). The timer for frame 0 starts at the first request, but stops when the ACK for this frame arrives. The timer for frame 1 starts at the second request, restarts when a NAK arrives, and finally stops when the last ACK arrives. The other two timers start when the corresponding frames are sent and stop at the last arrival event.

At the receiver site we need to distinguish between the acceptance of a frame and its delivery to the network layer. At the second arrival, frame 2 arrives and is stored and marked (colored slot), but it cannot be delivered because frame 1 is missing. At the next arrival, frame 3 arrives and is marked and stored, but still none of the frames can be delivered. Only at the last arrival, when finally a copy of frame 1 arrives, can frames 1, 2, and 3 be delivered to the network layer. There are two conditions for the delivery of frames to the network layer: First, a set of consecutive frames must have arrived. Second, the set starts from the beginning of the window. After the first arrival, there was only one frame and it started from the beginning of the window. After the last arrival, there are three frames and the first one starts from the beginning of the window.

Another important point is that a NAK is sent after the second arrival, but not after the third, although both situations look the same. The reason is that the protocol does not want to crowd the network with unnecessary NAKs and unnecessary resent frames. The second NAK would still be a NAK to inform the sender to resend frame 1 again; this has already been done. The first NAK sent is remembered (using the `nakSent` variable) and is not sent again until the frame slides. A NAK is sent once for each window position and defines the first slot in the window.

The next point is about the ACKs. Notice that only two ACKs are sent here. The first one acknowledges only the first frame; the second one acknowledges three frames. In Selective Repeat, ACKs are sent when data are delivered to the network layer. If the data belonging to n frames are delivered in one shot, only one ACK is sent for all of them.

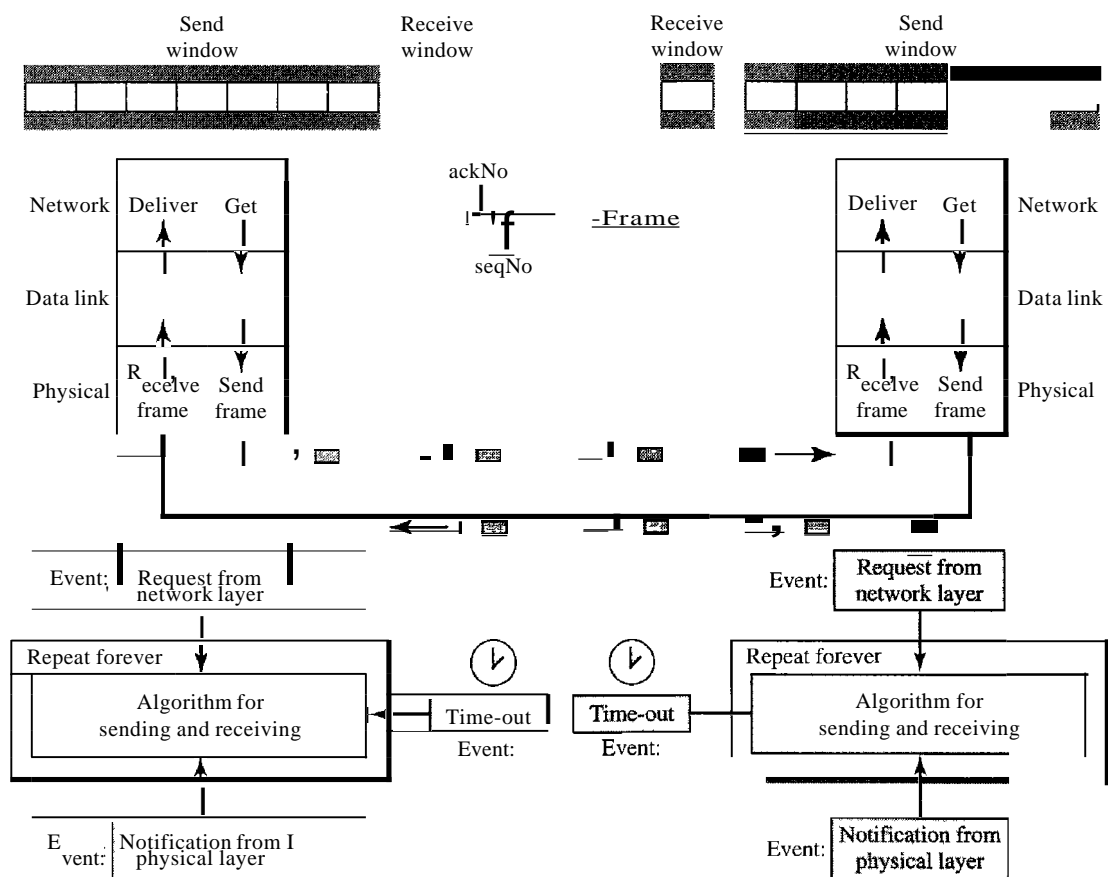
Piggybacking

The three protocols we discussed in this section are all unidirectional: data frames flow in only one direction although control information such as ACK and NAK frames can travel in the other direction. In real life, data frames are normally flowing in both directions: from node A to node B and from node B to node A. This means that the control information also needs to flow in both directions. A technique called **piggybacking** is used to improve the efficiency of the bidirectional protocols. When a frame is carrying data from A to B, it can also carry control information about arrived (or lost) frames from B; when a frame is carrying data from B to A, it can also carry control information about the arrived (or lost) frames from A.

We show the design for a Go-Back-N ARQ using piggybacking in Figure 11.24. Note that each node now has two windows: one send window and one receive window. Both also need to use a timer. Both are involved in three types of events: request, arrival, and time-out. However, the arrival event here is complicated; when a frame arrives, the site needs to handle control information as well as the frame itself. Both of these concerns must be taken care of in one event, the arrival event. The request event uses only the send window at each site; the arrival event needs to use both windows.

An important point about piggybacking is that both sites must use the same algorithm. This algorithm is complicated because it needs to combine two arrival events into one. We leave this task as an exercise.

Figure 11.24 Design of piggybacking in Go-Back-NARQ



11.6 HDLC

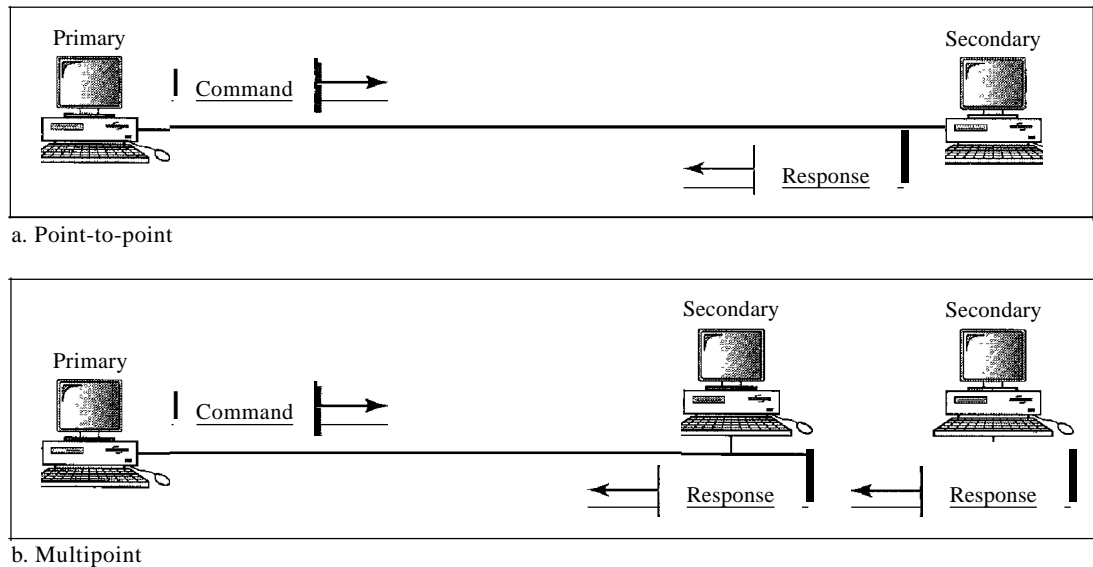
High-level Data Link Control (HDLC) is a bit-oriented protocol for communication over point-to-point and multipoint links. It implements the ARQ mechanisms we discussed in this chapter.

Configurations and Transfer Modes

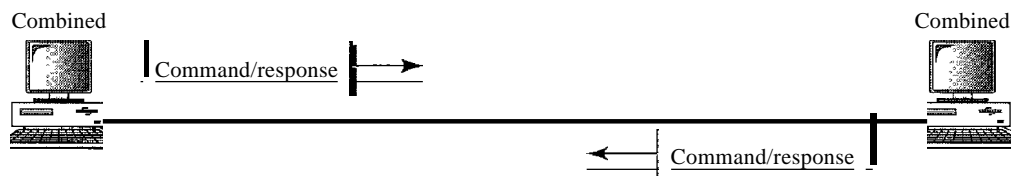
HDLC provides two common transfer modes that can be used in different configurations: normal response mode (NRM) and asynchronous balanced mode (ABM).

Normal Response Mode

In normal response mode (NRM), the station configuration is unbalanced. We have one primary station and multiple secondary stations. A primary station can send commands; a secondary station can only respond. The NRM is used for both point-to-point and multiple-point links, as shown in Figure 11.25.

Figure 11.25 *Normal response mode**Asynchronous Balanced Mode*

In asynchronous balanced mode (ABM), the configuration is balanced. The link is point-to-point, and each station can function as a primary and a secondary (acting as peers), as shown in Figure 11.26. This is the common mode today.

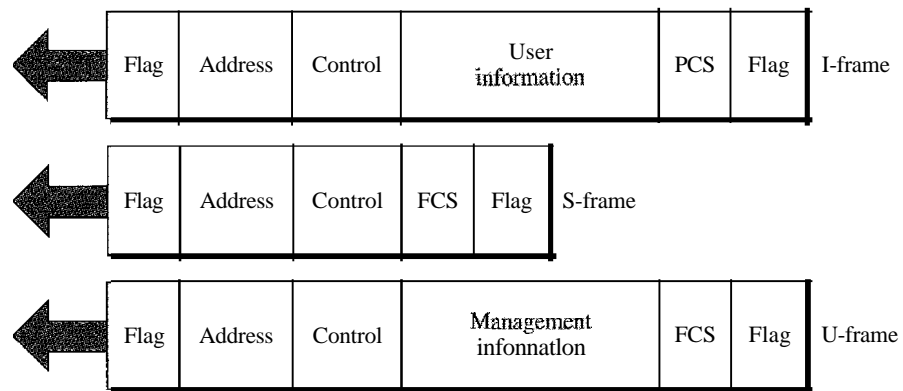
Figure 11.26 *Asynchronous balanced mode***Frames**

To provide the flexibility necessary to support all the options possible in the modes and configurations just described, HDLC defines three types of frames: information frames (I-frames), supervisory frames (S-frames), and unnumbered frames (V-frames). Each type of frame serves as an envelope for the transmission of a different type of message. I-frames are used to transport user data and control information relating to user data (piggybacking). S-frames are used only to transport control information. V-frames are reserved for system management. Information carried by V-frames is intended for managing the link itself.

Frame Format

Each frame in HDLC may contain up to six fields, as shown in Figure 11.27: a beginning flag field, an address field, a control field, an information field, a frame check sequence (FCS) field, and an ending flag field. In multiple-frame transmissions, the ending flag of one frame can serve as the beginning flag of the next frame.

Figure 11.27 HDLC frames



Fields

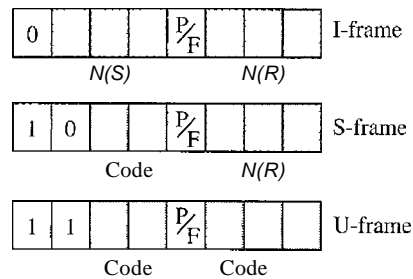
Let us now discuss the fields and their use in different frame types.

- **Flag field.** The flag field of an HDLC frame is an 8-bit sequence with the bit pattern 01111110 that identifies both the beginning and the end of a frame and serves as a synchronization pattern for the receiver.
- **Address field.** The second field of an HDLC frame contains the address of the secondary station. If a primary station created the frame, it contains a *to* address. If a secondary creates the frame, it contains a *from* address. An address field can be 1 byte or several bytes long, depending on the needs of the network. One byte can identify up to 128 stations (1 bit is used for another purpose). Larger networks require multiple-byte address fields. If the address field is only 1 byte, the last bit is always a 1. If the address is more than 1 byte, all bytes but the last one will end with 0; only the last will end with 1. Ending each intermediate byte with 0 indicates to the receiver that there are more address bytes to come.
- **Control field.** The control field is a 1- or 2-byte segment of the frame used for flow and error control. The interpretation of bits in this field depends on the frame type. We discuss this field later and describe its format for each frame type.
- **Information field.** The information field contains the user's data from the network layer or management information. Its length can vary from one network to another.
- **FCS field.** The frame check sequence (FCS) is the HDLC error detection field. It can contain either a 2- or 4-byte ITU-T CRC.

Control Field

The control field determines the type of frame and defines its functionality. So let us discuss the format of this field in greater detail. The format is specific for the type of frame, as shown in Figure 11.28.

Figure 11.28 Control field format for the different frame types



Control Field for I-Frames

I-frames are designed to carry user data from the network layer. In addition, they can include flow and error control information (piggybacking). The subfields in the control field are used to define these functions. The first bit defines the type. If the first bit of the control field is 0, this means the frame is an I-frame. The next 3 bits, called $N(S)$, define the sequence number of the frame. Note that with 3 bits, we can define a sequence number between 0 and 7; but in the extension format, in which the control field is 2 bytes, this field is larger. The last 3 bits, called $N(R)$, correspond to the acknowledgment number when piggybacking is used. The single bit between $N(S)$ and $N(R)$ is called the *PIF* bit. The *PIF* field is a single bit with a dual purpose. It has meaning only when it is set (bit = 1) and can mean *poll* or *final*. It means *poll* when the frame is sent by a primary station to a secondary (when the address field contains the address of the receiver). It means *final* when the frame is sent by a secondary to a primary (when the address field contains the address of the sender).

Control Field for S-Frames

Supervisory frames are used for flow and error control whenever piggybacking is either impossible or inappropriate (e.g., when the station either has no data of its own to send or needs to send a command or response other than an acknowledgment). S-frames do not have information fields. If the first 2 bits of the control field is 10, this means the frame is an S-frame. The last 3 bits, called $N(R)$, corresponds to the acknowledgment number (ACK) or negative acknowledgment number (NAK) depending on the type of S-frame. The 2 bits called code is used to define the type of S-frame itself. With 2 bits, we can have four types of S-frames, as described below:

- Receive ready (RR). If the value of the code subfield is 00, it is an RR S-frame. This kind of frame acknowledges the receipt of a safe and sound frame or group of frames. In this case, the value $N(R)$ field defines the acknowledgment number.

- Receive not ready (RNR). If the value of the code subfield is 10, it is an RNR S-frame. This kind of frame is an RR frame with additional functions. It acknowledges the receipt of a frame or group of frames, and it announces that the receiver is busy and cannot receive more frames. It acts as a kind of congestion control mechanism by asking the sender to slow down. The value of NCR is the acknowledgment number.
- Reject (REJ). If the value of the code subfield is 01, it is a REJ S-frame. This is a NAK frame, but not like the one used for Selective Repeat ARQ. It is a NAK that can be used in *Go-Back-N* ARQ to improve the efficiency of the process by informing the sender, before the sender time expires, that the last frame is lost or damaged. The value of NCR is the negative acknowledgment number.
- Selective reject (SREJ). If the value of the code subfield is 11, it is an SREJ S-frame. This is a NAK frame used in Selective Repeat ARQ. Note that the HDLC Protocol uses the term *selective reject* instead of *selective repeat*. The value of $N(R)$ is the negative acknowledgment number.

Control Field for V-Frames

Unnumbered frames are used to exchange session management and control information between connected devices. Unlike S-frames, U-frames contain an information field, but one used for system management information, not user data. As with S-frames, however, much of the information carried by U-frames is contained in codes included in the control field. U-frame codes are divided into two sections: a 2-bit prefix before the PtF bit and a 3-bit suffix after the PtF bit. Together, these two segments (5 bits) can be used to create up to 32 different types of U-frames. Some of the more common types are shown in Table 11.1.

Table 11.1 *U-frame control command and response*

<i>Code</i>	<i>Command</i>	<i>Response</i>	<i>Meaning</i>
00 001	SNRM		Set normal response mode
11 011	SNRME		Set normal response mode, extended
11 100	SABM	DM	Set asynchronous balanced mode or disconnect mode
11 110	SABME		Set asynchronous balanced mode, extended
00 000	UI	UI	Unnumbered information
00 110		UA	Unnumbered acknowledgment
00 010	DISC	RD	Disconnect or request disconnect
10 000	SIM	RIM	Set initialization mode or request information mode
00 100	UP		Unnumbered poll
11 001	RSET		Reset
11 101	XID	XID	Exchange ID
10 001	FRMR	FRMR	Frame reject

Example 11.9: Connection/Disconnection

Figure 11.29 shows how V-frames can be used for connection establishment and connection release. Node A asks for a connection with a set asynchronous balanced mode (SABM) frame; node B gives a positive response with an unnumbered acknowledgment (VA) frame. After these two exchanges, data can be transferred between the two nodes (not shown in the figure). After data transfer, node A sends a DISC (disconnect) frame to release the connection; it is confirmed by node B responding with a VA (unnumbered acknowledgment).

Figure 11.29 Example of connection and disconnection

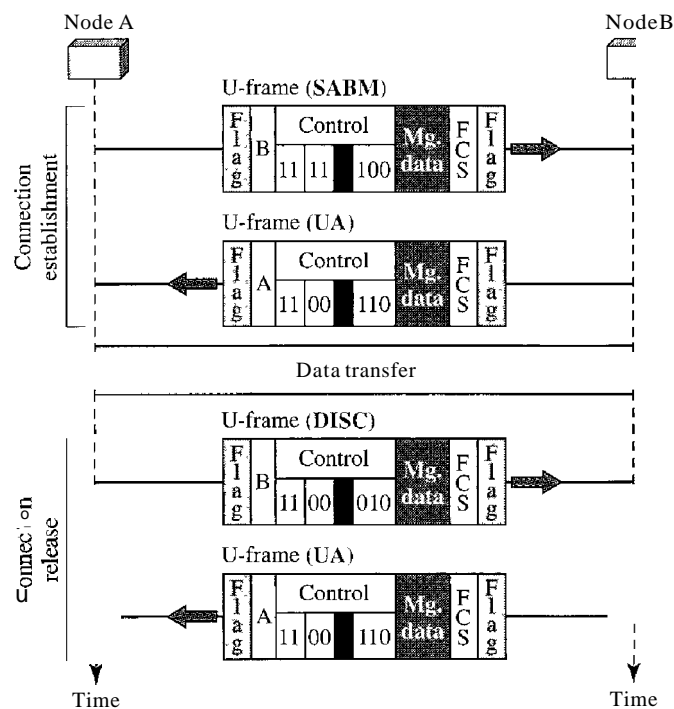
*Example 11.10: Piggybacking without Error*

Figure 11.30 shows an exchange using piggybacking. Node A begins the exchange of information with an I-frame numbered 0 followed by another I-frame numbered 1. Node B piggybacks its acknowledgment of both frames onto an I-frame of its own. Node B's first I-frame is also numbered 0 [$N(S)$ field] and contains a 2 in its $N(R)$ field, acknowledging the receipt of A's frames 1 and 0 and indicating that it expects frame 2 to arrive next. Node B transmits its second and third I-frames (numbered 1 and 2) before accepting further frames from node A. Its $N(R)$ information, therefore, has not changed: B frames 1 and 2 indicate that node B is still expecting A's frame 2 to arrive next. Node A has sent all its data. Therefore, it cannot piggyback an acknowledgment onto an I-frame and sends an S-frame instead. The RR code indicates that A is still ready to receive. The number 3 in the $N(R)$ field tells B that frames 0, 1, and 2 have all been accepted and that A is now expecting frame number 3.

Figure 11.30 Example of piggybacking without error

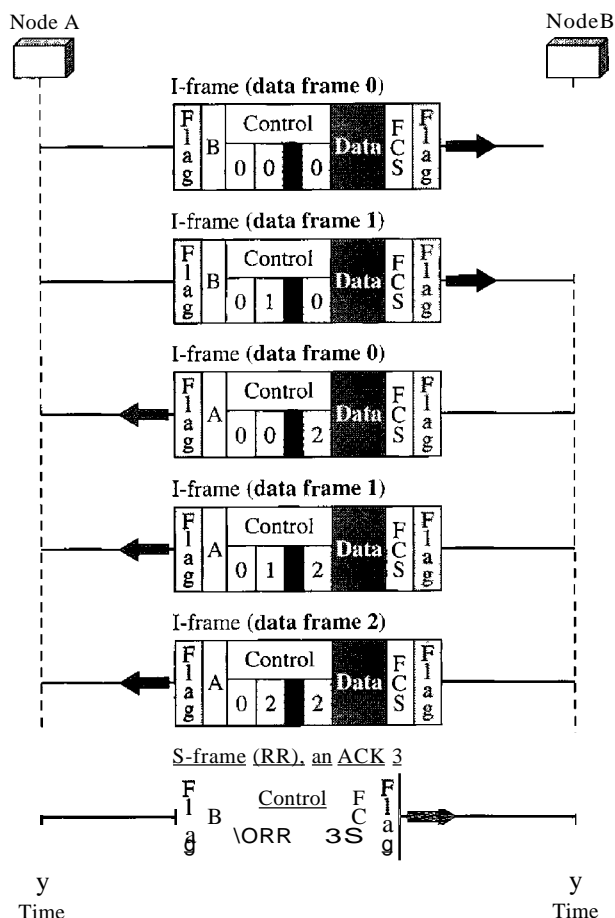
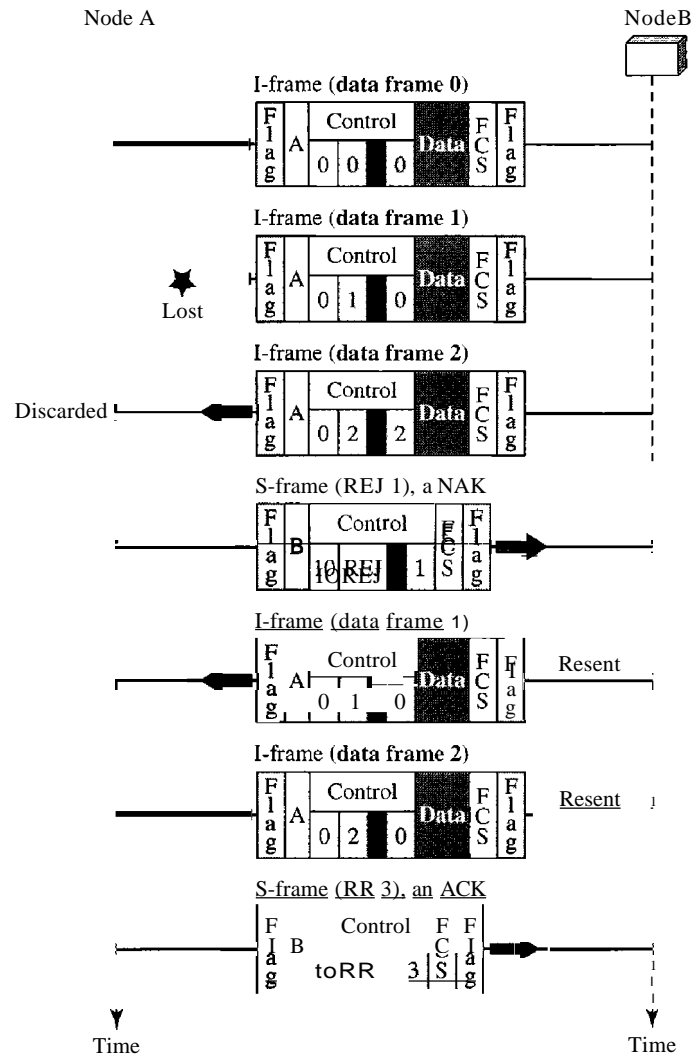
**Example 11.11: Piggybacking with Error**

Figure 11.31 shows an exchange in which a frame is lost. Node B sends three data frames (0, 1, and 2), but frame 1 is lost. When node A receives frame 2, it discards it and sends a *REI* frame for frame 1. Note that the protocol being used is *Go-Back-N* with the special use of an REI frame as a NAK frame. The NAK frame does two things here: It confirms the receipt of frame 0 and declares that frame 1 and any following frames must be resent. Node B, after receiving the REI frame, resends frames 1 and 2. Node A acknowledges the receipt by sending an RR frame (ACK) with acknowledgment number 3.

11.7 POINT-TO-POINT PROTOCOL

Although HDLC is a general protocol that can be used for both point-to-point and multi-point configurations, one of the most common protocols for point-to-point access is the Point-to-Point Protocol (PPP). Today, millions of Internet users who need to connect their home computers to the server of an Internet service provider use PPP. The majority of these users have a traditional modem; they are connected to the Internet through a telephone line, which provides the services of the physical layer. But to control and

Figure 11.31 Example of piggybacking with error



manage the transfer of data, there is a need for a point-to-point protocol at the data link layer. PPP is by far the most common.

PPP provides several services:

1. PPP defines the format of the frame to be exchanged between devices.
2. PPP defines how two devices can negotiate the establishment of the link and the exchange of data.
3. PPP defines how network layer data are encapsulated in the data link frame.
4. PPP defines how two devices can authenticate each other.
5. PPP provides multiple network layer services supporting a variety of network layer protocols.
6. PPP provides connections over multiple links.
7. PPP provides network address configuration. This is particularly useful when a home user needs a temporary network address to connect to the Internet.

On the other hand, to keep PPP simple, several services are missing:

1. PPP does not provide flow control. A sender can send several frames one after another with no concern about overwhelming the receiver.
2. PPP has a very simple mechanism for error control. A CRC field is used to detect errors. If the frame is corrupted, it is silently discarded; the upper-layer protocol needs to take care of the problem. Lack of error control and sequence numbering may cause a packet to be received out of order.
3. PPP does not provide a sophisticated addressing mechanism to handle frames in a multipoint configuration.

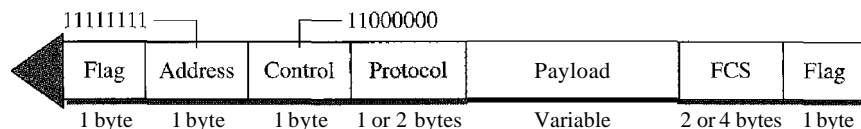
Framing

PPP is a byte-oriented protocol. Framing is done according to the discussion of byte-oriented protocols at the beginning of this chapter.

Frame Format

Figure 11.32 shows the format of a PPP frame. The description of each field follows:

Figure 11.32 PPP frame format



- **Flag.** A PPP frame starts and ends with a 1-byte flag with the bit pattern 01111110. Although this pattern is the same as that used in HDLC, there is a big difference. PPP is a byte-oriented protocol; HDLC is a bit-oriented protocol. The flag is treated as a byte, as we will explain later.
- **Address.** The address field in this protocol is a constant value and set to 11111111 (broadcast address). During negotiation (discussed later), the two parties may agree to omit this byte.
- **Control.** This field is set to the constant value 11000000 (imitating unnumbered frames in HDLC). As we will discuss later, PPP does not provide any flow control. Error control is also limited to error detection. This means that this field is not needed at all, and again, the two parties can agree, during negotiation, to omit this byte.
- **Protocol.** The protocol field defines what is being carried in the data field: either user data or other information. We discuss this field in detail shortly. This field is by default 2 bytes long, but the two parties can agree to use only 1 byte.
- **Payload field.** This field carries either the user data or other information that we will discuss shortly. The data field is a sequence of bytes with the default of a maximum of 1500 bytes; but this can be changed during negotiation. The data field is byte-stuffed if the flag byte pattern appears in this field. Because there is no field defining the size of the data field, padding is needed if the size is less than the maximum default value or the maximum negotiated value.
- **FCS.** The frame check sequence (FCS) is simply a 2-byte or 4-byte standard CRC.

Byte Stuffing

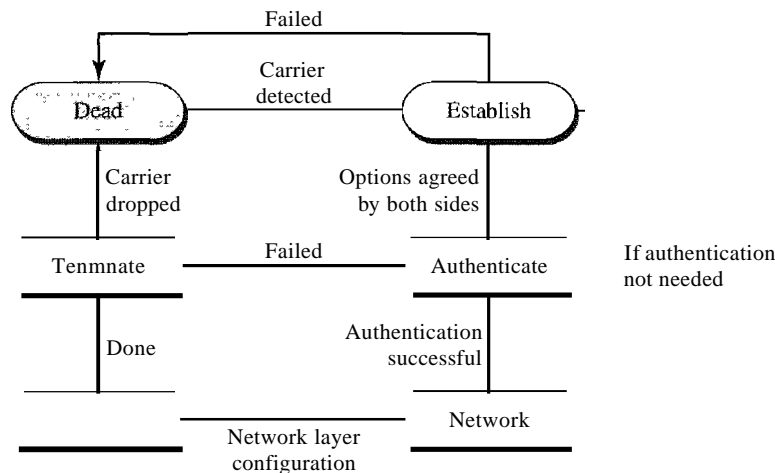
The similarity between PPP and HDLe ends at the frame format. PPP, as we discussed before, is a byte-oriented protocol totally different from HDLC. As a byte-oriented protocol, the flag in PPP is a byte and needs to be escaped whenever it appears in the data section of the frame. The escape byte is 01111101, which means that every time the flaglike pattern appears in the data, this extra byte is stuffed to tell the receiver that the next byte is not a flag.

PPP is a byte-oriented protocol using byte stuffing with the escape byte 01111101.

Transition Phases

A PPP connection goes through phases which can be shown in a transition phase diagram (see Figure 11.33).

Figure 11.33 *Transition phases*



- D **Dead.** In the dead phase the link is not being used. There is no active carrier (at the physical layer) and the line is quiet.
- D **Establish.** When one of the nodes starts the communication, the connection goes into this phase. In this phase, options are negotiated between the two parties. If the negotiation is successful, the system goes to the authentication phase (if authentication is required) or directly to the networking phase. The link control protocol packets, discussed shortly, are used for this purpose. Several packets may be exchanged here.
- D **Authenticate.** The authentication phase is optional; the two nodes may decide, during the establishment phase, not to skip this phase. However, if they decide to proceed with authentication, they send several authentication packets, discussed later. If the result is successful, the connection goes to the networking phase; otherwise, it goes to the termination phase.
- D **Network.** In the network phase, negotiation for the network layer protocols takes place. PPP specifies that two nodes establish a network layer agreement before data at

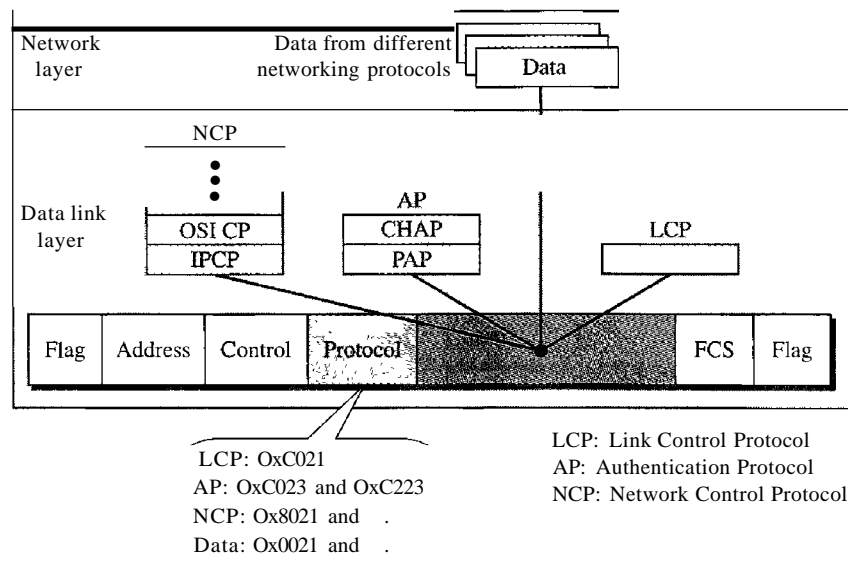
the network layer can be exchanged. The reason is that PPP supports multiple protocols at the network layer. If a node is running multiple protocols simultaneously at the network layer, the receiving node needs to know which protocol will receive the data.

- O Open.** In the open phase, data transfer takes place. When a connection reaches this phase, the exchange of data packets can be started. The connection remains in this phase until one of the endpoints wants to terminate the connection.
- O Terminate.** In the termination phase the connection is terminated. Several packets are exchanged between the two ends for house cleaning and closing the link.

Multiplexing

Although PPP is a data link layer protocol, PPP uses another set of other protocols to establish the link, authenticate the parties involved, and carry the network layer data. Three sets of protocols are defined to make PPP powerful: the Link Control Protocol (LCP), two Authentication Protocols (APs), and several Network Control Protocols (NCPs). At any moment, a PPP packet can carry data from one of these protocols in its data field, as shown in Figure 11.34. Note that there is one LCP, two APs, and several NCPs. Data may also come from several different network layers.

Figure 11.34 Multiplexing in PPP



Link Control Protocol

The **Link Control Protocol (LCP)** is responsible for establishing, maintaining, configuring, and terminating links. It also provides negotiation mechanisms to set options between the two endpoints. Both endpoints of the link must reach an agreement about the options before the link can be established. See Figure 11.35.

All LCP packets are carried in the payload field of the PPP frame with the protocol field set to C021 in hexadecimal.

The code field defines the type of LCP packet. There are 11 types of packets as shown in Table 11.2.

Figure 11.35 LCP packet encapsulated in a frame

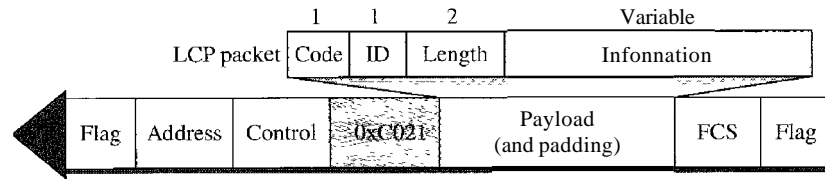


Table 11.2 LCP packets

<i>Code</i>	<i>Packet Type</i>	<i>Description</i>
0x01	Configure-request	Contains the list of proposed options and their values
0x02	Configure-ack	Accepts all options proposed
0x03	Configure-nak	Announces that some options are not acceptable
0x04	Configure-reject	Announces that some options are not recognized
0x05	Terminate-request	Request to shut down the line
0x06	Terminate-ack	Accept the shutdown request
0x07	Code-reject	Announces an unknown code
0x08	Protocol-reject	Announces an unknown protocol
0x09	Echo-request	A type of hello message to check if the other end is alive
0x0A	Echo-reply	The response to the echo-request message
0x0B	Discard-request	A request to discard the packet

There are three categories of packets. The first category, comprising the first four packet types, is used for link configuration during the establish phase. The second category, comprising packet types 5 and 6, is used for link termination during the termination phase. The last five packets are used for link monitoring and debugging.

The ID field holds a value that matches a request with a reply. One endpoint inserts a value in this field, which will be copied into the reply packet. The length field defines the length of the entire LCP packet. The information field contains information, such as options, needed for some LCP packets.

There are many options that can be negotiated between the two endpoints. Options are inserted in the information field of the configuration packets. In this case, the information field is divided into three fields: option type, option length, and option data. We list some of the most common options in Table 11.3.

Table 11.3 Common options

<i>Option</i>	<i>Default</i>
Maximum receive unit (payload field size)	1500
Authentication protocol	None
Protocol field compression	Off
Address and control field compression	Off

Authentication Protocols

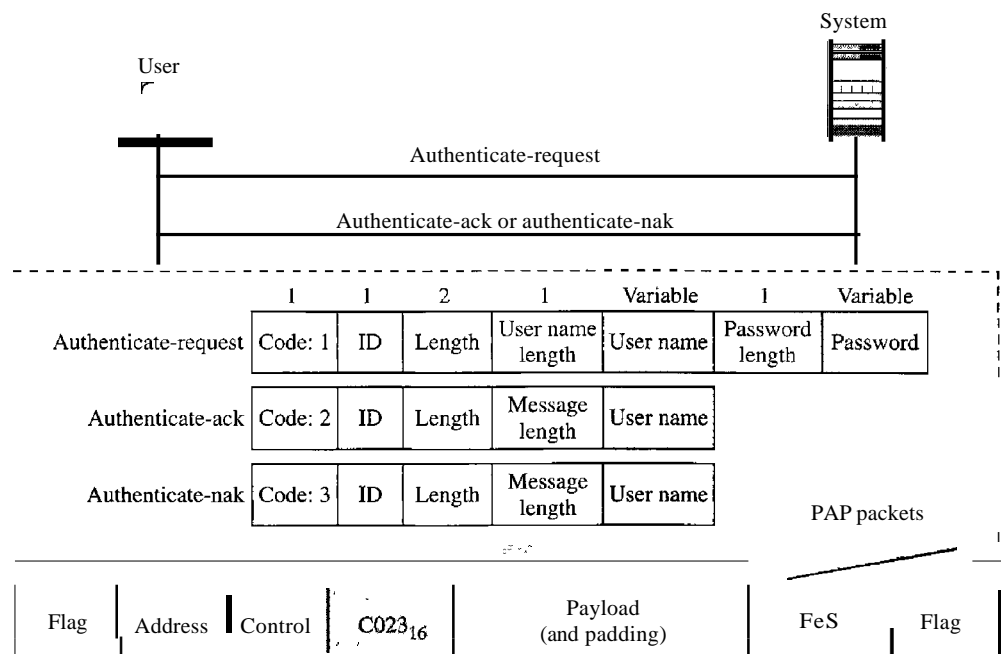
Authentication plays a very important role in PPP because PPP is designed for use over dial-up links where verification of user identity is necessary. **Authentication** means validating the identity of a user who needs to access a set of resources. PPP has created two protocols for authentication: Password Authentication Protocol and Challenge Handshake Authentication Protocol. Note that these protocols are used during the authentication phase.

PAP The **Password Authentication Protocol (PAP)** is a simple authentication procedure with a two-step process:

1. The user who wants to access a system sends an authentication identification (usually the user name) and a password.
2. The system checks the validity of the identification and password and either accepts or denies connection.

Figure 11.36 shows the three types of packets used by PAP and how they are actually exchanged. When a PPP frame is carrying any PAP packets, the value of the protocol field is OxCO23. The three PAP packets are authenticate-request, authenticate-ack, and authenticate-nak. The first packet is used by the user to send the user name and password. The second is used by the system to allow access. The third is used by the system to deny access.

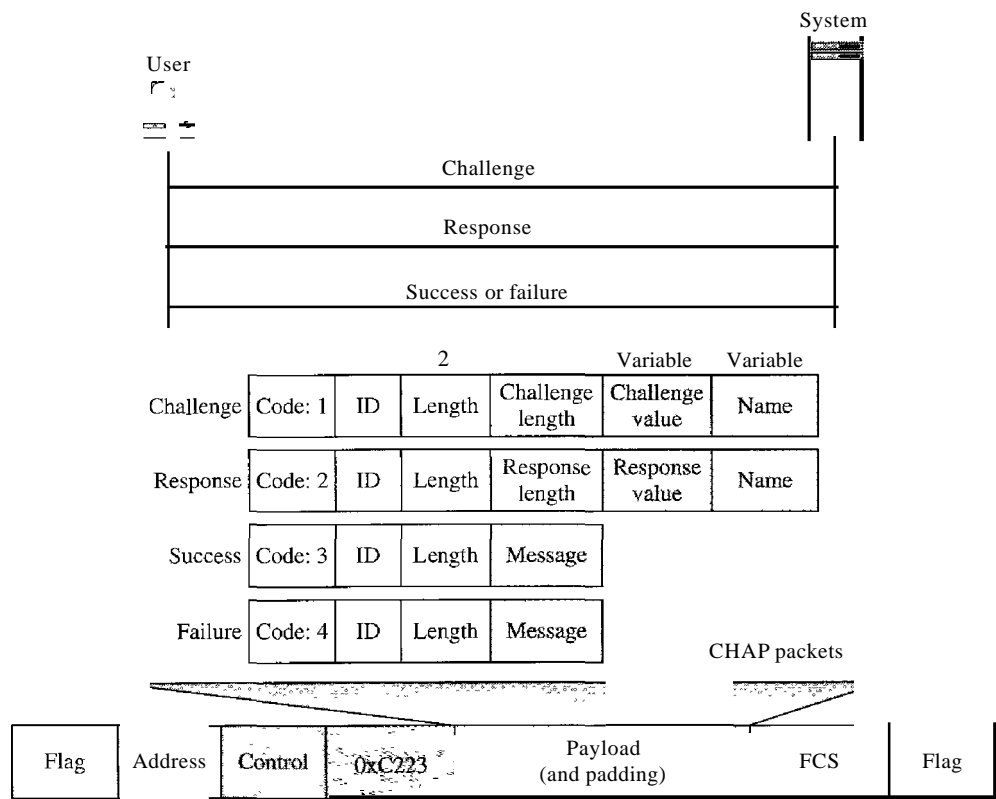
Figure 11.36 PAP packets encapsulated in a PPP frame



CHAP The **Challenge Handshake Authentication Protocol (CHAP)** is a three-way hand-shaking authentication protocol that provides greater security than PAP. In this method, the password is kept secret; it is never sent online.

1. The system sends the user a challenge packet containing a challenge value, usually a few bytes.
2. The user applies a predefined function that takes the challenge value and the user's own password and creates a result. The user sends the result in the response packet to the system.
3. The system does the same. It applies the same function to the password of the user (known to the system) and the challenge value to create a result. If the result created is the same as the result sent in the response packet, access is granted; otherwise, it is denied. CHAP is more secure than PAP, especially if the system continuously changes the challenge value. Even if the intruder learns the challenge value and the result, the password is still secret. Figure 11.37 shows the packets and how they are used.

Figure 11.37 CHAP packets encapsulated in a PPP frame



CHAP packets are encapsulated in the PPP frame with the protocol value C223 in hexadecimal. There are four CHAP packets: challenge, response, success, and failure. The first packet is used by the system to send the challenge value. The second is used by the user to return the result of the calculation. The third is used by the system to allow access to the system. The fourth is used by the system to deny access to the system.

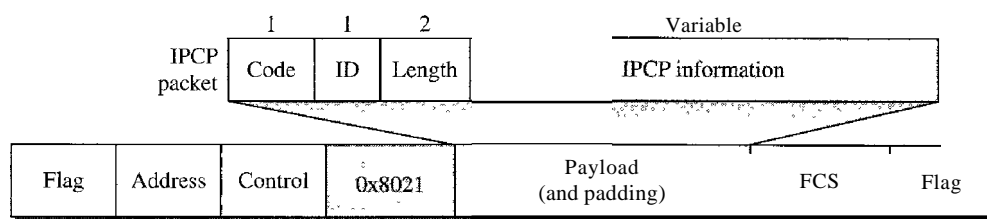
Network Control Protocols

PPP is a multiple-network layer protocol. It can carry a network layer data packet from protocols defined by the Internet, OSI, Xerox, DECnet, AppleTalk, Novel, and so on.

To do this, PPP has defined a specific Network Control Protocol for each network protocol. For example, IPCP (Internet Protocol Control Protocol) configures the link for carrying IP data packets. Xerox CP does the same for the Xerox protocol data packets, and so on. Note that none of the NCP packets carry network layer data; they just configure the link at the network layer for the incoming data.

IPCP One NCP protocol is the Internet Protocol Control Protocol (IPCP). This protocol configures the link used to carry IP packets in the Internet. IPCP is especially of interest to us. The format of an IPCP packet is shown in Figure 11.38. Note that the value of the protocol field in hexadecimal is 8021.

Figure 11.38 *fPCP packet encapsulated in PPP frame*



IPCP defines seven packets, distinguished by their code values, as shown in Table 11.4.

Table 11.4 *Code value for IPCP packets*

<i>Code</i>	<i>IPCP Packet</i>
0x01	Configure-request
0x02	Configure-ack
0x03	Configure-nak
0x04	Configure-reject
0x05	Terminate-request
0x06	Terminate-ack
0x07	Code-reject

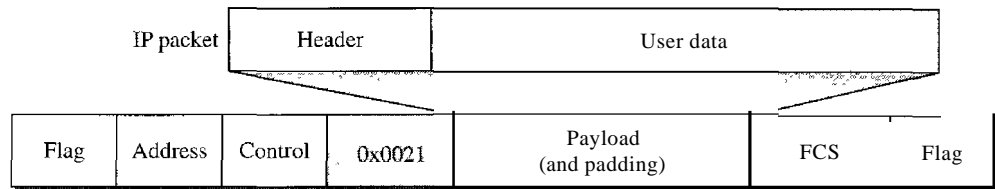
Other Protocols There are other NCP protocols for other network layer protocols. The OSI Network Layer Control Protocol has a protocol field value of 8023; the Xerox NS IDP Control Protocol has a protocol field value of 8025; and so on. The value of the code and the format of the packets for these other protocols are the same as shown in Table 11.4.

Data from the Network Layer

After the network layer configuration is completed by one of the NCP protocols, the users can exchange data packets from the network layer. Here again, there are different

protocol fields for different network layers. For example, if PPP is carrying data from the IP network layer, the field value is 0021 (note that the three rightmost digits are the same as for IPCP). If PPP is carrying data from the OSI network layer, the value of the protocol field is 0023, and so on. Figure 11.39 shows the frame for IP.

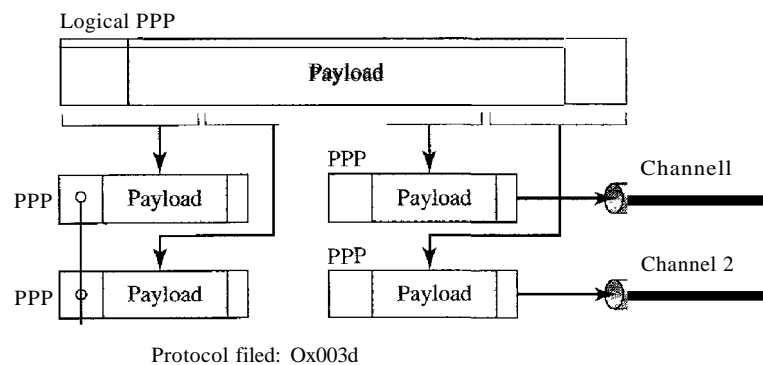
Figure 11.39 IP datagram encapsulated in a PPP frame



Multilink PPP

PPP was originally designed for a single-channel point-to-point physical link. The availability of multiple channels in a single point-to-point link motivated the development of Multilink PPP. In this case, a logical PPP frame is divided into several actual PPP frames. A segment of the logical frame is carried in the payload of an actual PPP frame, as shown in Figure 11.40. To show that the actual PPP frame is carrying a fragment of a

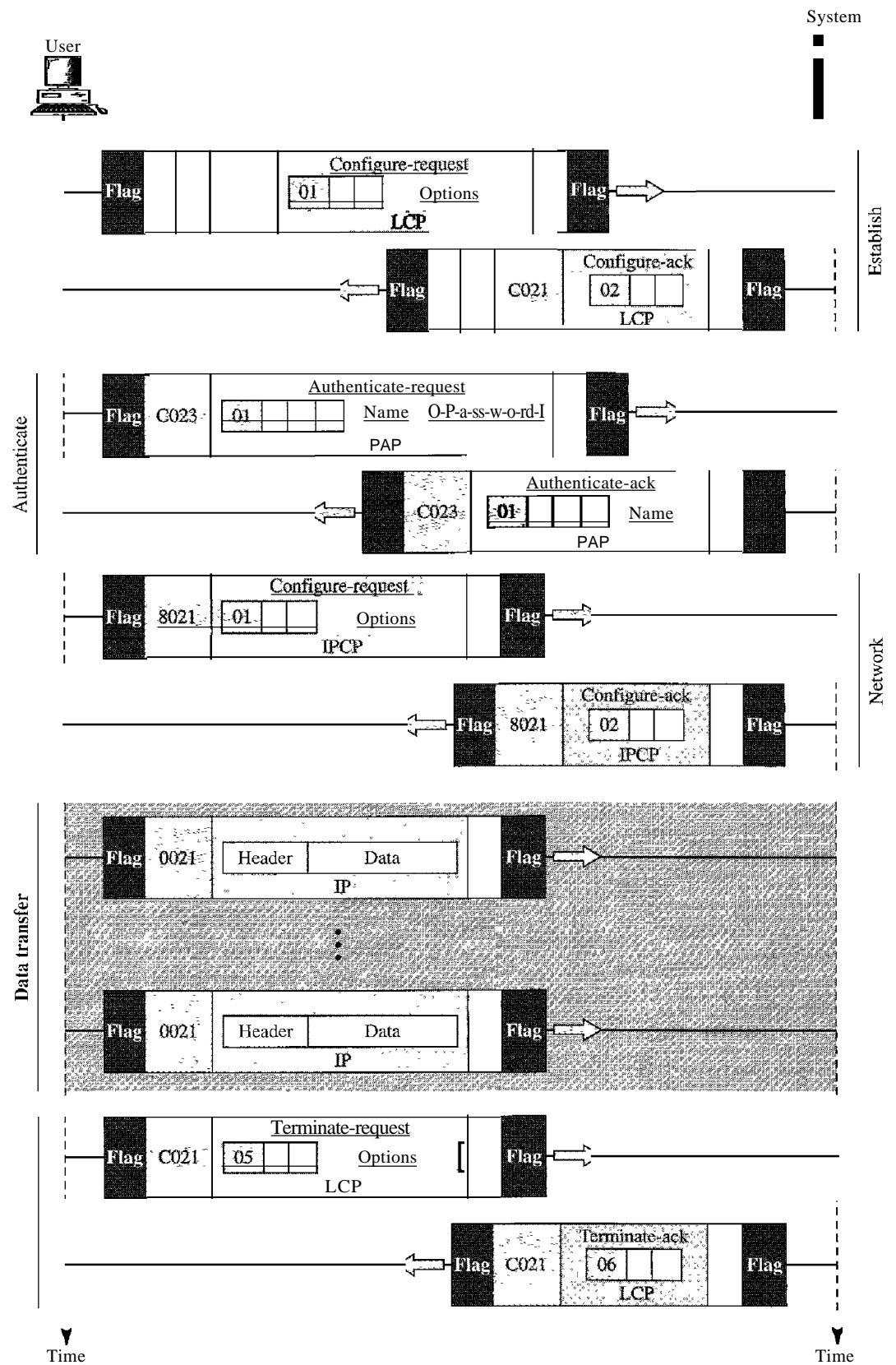
Figure 11.40 Multilink PPP



logical PPP frame, the protocol field is set to 0x003d. This new development adds complexity. For example, a sequence number needs to be added to the actual PPP frame to show a fragment's position in the logical frame.

Example 11.12

Let us go through the phases followed by a network layer packet as it is transmitted through a PPP connection. Figure 11.41 shows the steps. For simplicity, we assume unidirectional movement of data from the user site to the system site (such as sending an e-mail through an ISP).

Figure 11.41 *An example*

The first two frames show link establishment. We have chosen two options (not shown in the figure): using PAP for authentication and suppressing the address control fields. Frames 3 and 4 are for authentication. Frames 5 and 6 establish the network layer connection using IPCP.

The next several frames show that some IP packets are encapsulated in the PPP frame. The system (receiver) may have been running several network layer protocols, but it knows that the incoming data must be delivered to the IP protocol because the NCP protocol used before the data transfer was IPCP.

After data transfer, the user then terminates the data link connection, which is acknowledged by the system. Of COUrse the user or the system could have chosen to terminate the network layer IPCP and keep the data link layer running if it wanted to run another NCP protocol.

The example is trivial, but it points out the similarities of the packets in LCP, AP, and NCP. It also shows the protocol field values and code numbers for particular protocols.

11.8 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [...] refer to the reference list at the end of the text.

Books

A discussion of data link control can be found in [GW04], Chapter 3 of [Tan03], Chapter 7 of [Sta04], Chapter 12 of [Kes97], and Chapter 2 of [PD03]. More advanced materials can be found in [KMK04].

11.9 KEY TERMS

acknowledgment (ACK)	flow control
asynchronous balanced mode (ABM)	framing
automatic repeat request (ARQ)	<i>Go-Back-N</i> ARQ Protocol
bandwidth-delay product	High-level Data Link Control (HDLC)
bit-oriented protocol	information frame (I-frame)
bit stuffing	Internet Protocol Control Protocol (IPCP)
byte stuffing	Link Control Protocol (LCP)
Challenge Handshake Authentication Protocol (CHAP)	negative acknowledgment (NAK)
character-oriented protocol	noiseless channel
data link control	noisy channel
error control	normal response mode (NRM)
escape character (ESC)	Password Authentication Protocol (PAP)
event	piggybacking
fixed-size framing	pipelining
flag	Point-to-Point Protocol (PPP)
	primary station

receive sliding window	sliding window
secondary station	Stop-and-Wait ARQ Protocol
Selective Repeat ARQ Protocol	Stop-and-Wait Protocol
send sliding window	supervisory frame (S-frame)
sequence number	transition phase
Simplest Protocol	unnumbered frame (D-frame)
	variable-size framing

11.10 SUMMARY

- O Data link control deals with the design and procedures for communication between two adjacent nodes: node-to-node communication.
- O Framing in the data link layer separates a message from one source to a destination, or from other messages going from other sources to other destinations,
- O Frames can be of fixed or variable size. In fixed-size framing, there is no need for defining the boundaries of frames; in variable-size framing, we need a delimiter (flag) to define the boundary of two frames.
- O Variable-size framing uses two categories of protocols: byte-oriented (or character-oriented) and bit-oriented. In a byte-oriented protocol, the data section of a frame is a sequence of bytes; in a bit-oriented protocol, the data section of a frame is a sequence of bits.
- O In byte-oriented (or character-oriented) protocols, we use byte stuffing; a special byte added to the data section of the frame when there is a character with the same pattern as the flag.
- O In bit-oriented protocols, we use bit stuffing; an extra 0 is added to the data section of the frame when there is a sequence of bits with the same pattern as the flag.
- O Flow control refers to a set of procedures used to restrict the amount of data that the sender can send before waiting for acknowledgment. Error control refers to methods of error detection and correction.
- O For the noiseless channel, we discussed two protocols: the Simplest Protocol and the Stop-and-Wait Protocol. The first protocol has neither flow nor error control; the second has no error control. In the Simplest Protocol, the sender sends its frames one after another with no regards to the receiver. In the Stop-and-Wait Protocol, the sender sends one frame, stops until it receives confirmation from the receiver, and then sends the next frame.
- D For the noisy channel, we discussed three protocols: Stop-and-Wait ARQ, 00-Back-N, and Selective Repeat ARQ. The Stop-and-Wait ARQ Protocol, adds a simple error control mechanism to the Stop-and-Wait Protocol. In the 00-Back-N ARQ Protocol, we can send several frames before receiving acknowledgments, improving the efficiency of transmission. In the Selective Repeat ARQ protocol we avoid unnecessary transmission by sending only frames that are corrupted.
- D Both 00-Back-N and Selective-Repeat Protocols use a sliding window. In 00-Back-N ARQ, if m is the number of bits for the sequence number, then the size of

the send window must be less than 2^m ; the size of the receiver window is always 1. In Selective Repeat ARQ, the size of the sender and receiver window must be at most one-half of 2^m .

- O A technique called piggybacking is used to improve the efficiency of the bidirectional protocols. When a frame is carrying data from A to B, it can also carry control information about frames from B; when a frame is carrying data from B to A, it can also carry control information about frames from A.
- O High-level Data Link Control (HDLC) is a bit-oriented protocol for communication over point-to-point and multipoint links. However, the most common protocols for point-to-point access is the Point-to-Point Protocol (PPP), which is a byte-oriented protocol.

11.11 PRACTICE SET

Review Questions

1. Briefly describe the services provided by the data link layer.
2. Define framing and the reason for its need.
3. Compare and contrast byte-oriented and bit-oriented protocols. Which category has been popular in the past (explain the reason)? Which category is popular now (explain the reason)?
4. Compare and contrast byte-stuffing and bit-stuffing. Which technique is used in byte-oriented protocols? Which technique is used in bit-oriented protocols?
5. Compare and contrast flow control and error control.
6. What are the two protocols we discussed for noiseless channels in this chapter?
7. What are the three protocols we discussed for noisy channels in this chapter?
8. Explain the reason for moving from the Stop-and-Wait ARQ Protocol to the 00-Back-NARQ Protocol.
9. Compare and contrast the Go-Back-NARQ Protocol with Selective-RepeatARQ.
10. Compare and contrast HDLC with PPP. Which one is byte-oriented; which one is bit-oriented?
11. Define piggybacking and its usefulness.
12. Which of the protocols described in this chapter utilize pipelining?

Exercises

13. Byte-stuff the data in Figure 11.42.

Figure 11.42 Exercise 13

ESC			Flag			ESC	ESC	ESC		Flag	
-----	--	--	------	--	--	-----	-----	-----	--	------	--

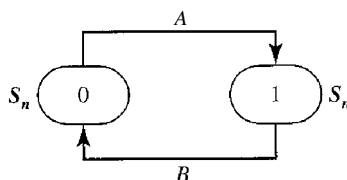
14. Bit-stuff the data in Figure 11.43.

Figure 11.43 Exercise 14

1000111111100111110100011111111111000011111 |

15. Design two simple algorithms for byte-stuffing. The first adds bytes at the sender; the second removes bytes at the receiver.
16. Design two simple algorithms for bit-stuffing. The first adds bits at the sender; the second removes bits at the receiver.
17. A sender sends a series of packets to the same destination using 5-bit sequence numbers. If the sequence number starts with 0, what is the sequence number after sending 100 packets?
18. Using 5-bit sequence numbers, what is the maximum size of the send and receive windows for each of the following protocols?
 - a. Stop-and-Wait ARQ
 - b. Go-Back-N ARQ
 - c. Selective-Repeat ARQ
19. Design a bidirectional algorithm for the Simplest Protocol using piggybacking. Note that the both parties need to use the same algorithm.
20. Design a bidirectional algorithm for the Stop-and-Wait Protocol using piggybacking. Note that both parties need to use the same algorithm.
21. Design a bidirectional algorithm for the Stop-and-Wait ARQ Protocol using piggybacking. Note that both parties need to use the same algorithm.
22. Design a bidirectional algorithm for the Go-Back-N ARQ Protocol using piggybacking. Note that both parties need to use the same algorithm.
23. Design a bidirectional algorithm for the Selective-Repeat ARQ Protocol using piggybacking. Note that both parties need to use the same algorithm.
24. Figure 11.44 shows a state diagram to simulate the behavior of Stop-and-Wait ARQ at the sender site.

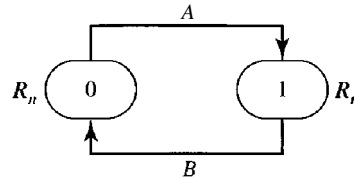
Figure 11.44 Exercise 24



The states have a value of S_n (0 or 1). The arrows show the transitions. Explain the events that cause the two transitions labeled A and B.

25. Figure 11.45 shows a state diagram to simulate the behavior of Stop-and-Wait ARQ at the receiver site.

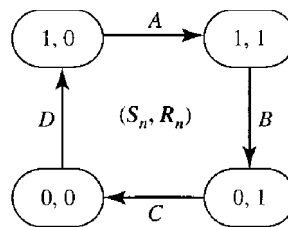
Figure 11.45 Exercise 25



The states have a value of R_n (0 or 1). The arrows show the transitions. Explain the events that cause the two transitions labeled A and B.

26. In Stop-and-Wait ARQ, we can combine the state diagrams of the sender and receiver in Exercises 24 and 25. One state defines the combined values of R_n and S_n . This means that we can have four states, each defined by (x, y) , where x defines the value of S_n and y defines the value of R_n . In other words, we can have the four states shown in Figure 11.46. Explain the events that cause the four transitions labeled A, B, C, and D.

Figure 11.46 Exercise 26



27. The timer of a system using the Stop-and-Wait ARQ Protocol has a time-out of 6 ms. Draw the flow diagram similar to Figure 11.11 for four frames if the round trip delay is 4 ms. Assume no data frame or control frame is lost or damaged.
28. Repeat Exercise 27 if the time-out is 4 ms and the round trip delay is 6.
29. Repeat Exercise 27 if the first frame (frame 0) is lost.
30. A system uses the Stop-and-Wait ARQ Protocol. If each packet carries 1000 bits of data, how long does it take to send 1 million bits of data if the distance between the sender and receiver is 5000 Km and the propagation speed is 2×10^8 m? Ignore transmission, waiting, and processing delays. We assume no data or control frame is lost or damaged.
31. Repeat Exercise 30 using the Go-back-N ARQ Protocol with a window size of 7. Ignore the overhead due to the header and trailer.
32. Repeat Exercise 30 using the Selective-Repeat ARQ Protocol with a window size of 4. Ignore the overhead due to the header and the trailer.

CHAPTER 12

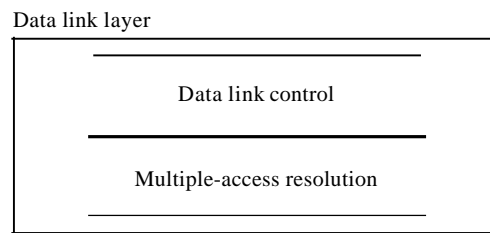
Multiple Access

In Chapter 11 we discussed data link control, a mechanism which provides a link with reliable communication. In the protocols we described, we assumed that there is an available dedicated link (or channel) between the sender and the receiver. This assumption may or may not be true. If, indeed, we have a dedicated link, as when we connect to the Internet using PPP as the data link control protocol, then the assumption is true and we do not need anything else.

On the other hand, if we use our cellular phone to connect to another cellular phone, the channel (the band allocated to the vendor company) is not dedicated. A person a few feet away from us may be using the same channel to talk to her friend.

We can consider the data link layer as two sublayers. The upper sublayer is responsible for data link control, and the lower sublayer is responsible for resolving access to the shared media. If the channel is dedicated, we do not need the lower sublayer. Figure 12.1 shows these two sublayers in the data link layer.

Figure 12.1 *Data link layer divided into two functionality-oriented sublayers*



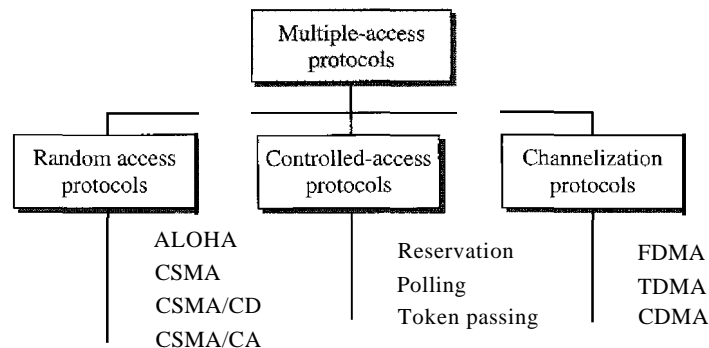
We will see in Chapter 13 that the IEEE has actually made this division for LANs. The upper sublayer that is responsible for flow and error control is called the logical link control (LLC) layer; the lower sublayer that is mostly responsible for multiple-access resolution is called the media access control (MAC) layer.

When nodes or stations are connected and use a common link, called a multipoint or broadcast link, we need a multiple-access protocol to coordinate access to the link. The problem of controlling the access to the medium is similar to the rules of speaking

in an assembly. The procedures guarantee that the right to speak is upheld and ensure that two people do not speak at the same time, do not interrupt each other, do not monopolize the discussion, and so on.

The situation is similar for multipoint networks. Many formal protocols have been devised to handle access to a shared link. We categorize them into three groups. Protocols belonging to each group are shown in Figure 12.2.

Figure 12.2 *Taxonomy of multiple-access protocols discussed in this chapter*



12.1 RANDOM ACCESS

In random access or contention methods, no station is superior to another station and none is assigned the control over another. No station permits, or does not permit, another station to send. At each instance, a station that has data to send uses a procedure defined by the protocol to make a decision on whether or not to send. This decision depends on the state of the medium (idle or busy). In other words, each station can transmit when it desires on the condition that it follows the predefined procedure, including the testing of the state of the medium.

Two features give this method its name. First, there is no scheduled time for a station to transmit. Transmission is random among the stations. That is why these methods are called *random access*. Second, no rules specify which station should send next. Stations compete with one another to access the medium. That is why these methods are also called *contention* methods.

In a random access method, each station has the right to the medium without being controlled by any other station. However, if more than one station tries to send, there is an access conflict-collision-and the frames will be either destroyed or modified. To avoid access conflict or to resolve it when it happens, each station follows a procedure that answers the following questions:

- When can the station access the medium?
- What can the station do if the medium is busy?
- How can the station determine the success or failure of the transmission?
- What can the station do if there is an access conflict?

The random access methods we study in this chapter have evolved from a very interesting protocol known as ALOHA, which used a very simple procedure called multiple access (MA). The method was improved with the addition of a procedure that forces the station to sense the medium before transmitting. This was called carrier sense multiple access. This method later evolved into two parallel methods: carrier sense multiple access with collision detection (CSMA/CD) and carrier sense multiple access with collision avoidance (CSMA/CA). CSMA/CD tells the station what to do when a collision is detected. CSMA/CA tries to avoid the collision.

ALOHA

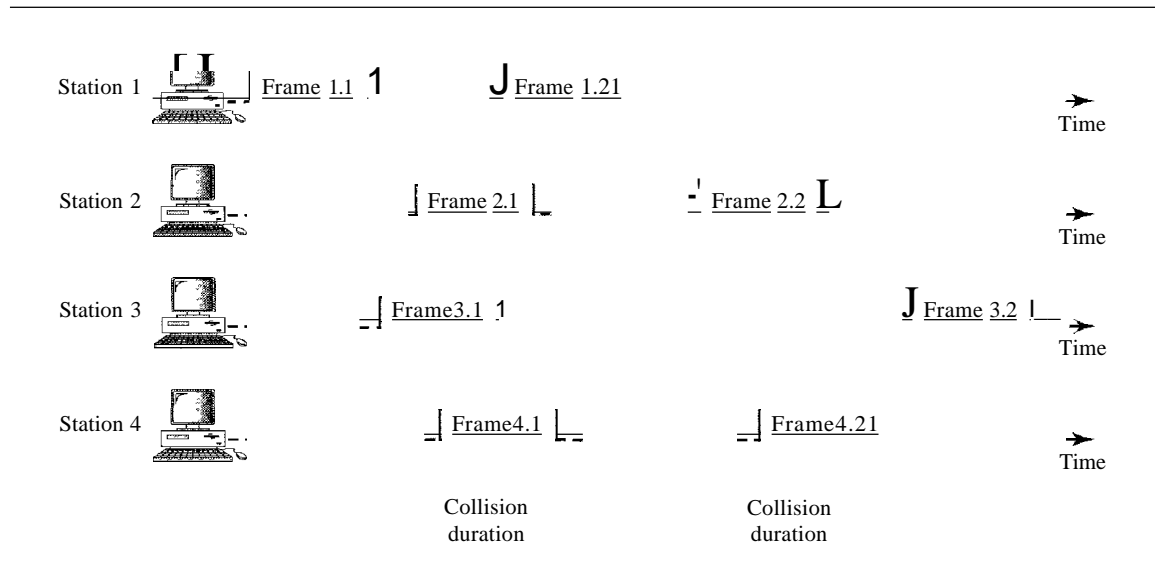
ALOHA, the earliest random access method, was developed at the University of Hawaii in early 1970. It was designed for a radio (wireless) LAN, but it can be used on any shared medium.

It is obvious that there are potential collisions in this arrangement. The medium is shared between the stations. When a station sends data, another station may attempt to do so at the same time. The data from the two stations collide and become garbled.

Pure ALOHA

The original ALOHA protocol is called pure ALOHA. This is a simple, but elegant protocol. The idea is that each station sends a frame whenever it has a frame to send. However, since there is only one channel to share, there is the possibility of collision between frames from different stations. Figure 12.3 shows an example of frame collisions in pure ALOHA.

Figure 12.3 Frames in a pure ALOHA network



There are four stations (unrealistic assumption) that contend with one another for access to the shared channel. The figure shows that each station sends two frames; there are a total of eight frames on the shared medium. Some of these frames collide because multiple frames are in contention for the shared channel. Figure 12.3 shows that only

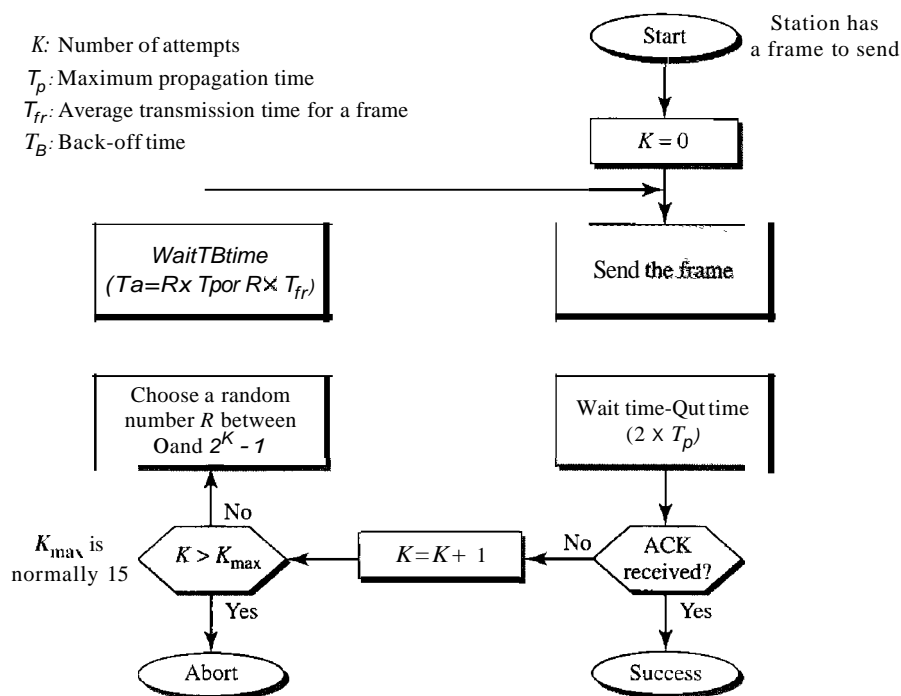
two frames survive: frame 1.1 from station 1 and frame 3.2 from station 3. We need to mention that even if one bit of a frame coexists on the channel with one bit from another frame, there is a collision and both will be destroyed.

It is obvious that we need to resend the frames that have been destroyed during transmission. The pure ALOHA protocol relies on acknowledgments from the receiver. When a station sends a frame, it expects the receiver to send an acknowledgment. If the acknowledgment does not arrive after a time-out period, the station assumes that the frame (or the acknowledgment) has been destroyed and resends the frame.

A collision involves two or more stations. If all these stations try to resend their frames after the time-out, the frames will collide again. Pure ALOHA dictates that when the time-out period passes, each station waits a random amount of time before resending its frame. The randomness will help avoid more collisions. We call this time the back-off time T_B .

Pure ALOHA has a second method to prevent congesting the channel with retransmitted frames. After a maximum number of retransmission attempts K_{\max} , a station must give up and try later. Figure 12.4 shows the procedure for pure ALOHA based on the above strategy.

Figure 12.4 Procedure for pure ALOHA protocol



The time-out period is equal to the maximum possible round-trip propagation delay, which is twice the amount of time required to send a frame between the two most widely separated stations ($2 \times T_p$). The back-off time T_B is a random value that normally depends on K (the number of attempted unsuccessful transmissions). The formula for T_B depends on the implementation. One common formula is the **binary exponential back-off**. In this

method, for each retransmission, a multiplier in the range 0 to $2^K - 1$ is randomly chosen and multiplied by T_p (maximum propagation time) or T_{fr} (the average time required to send out a frame) to find T_B . Note that in this procedure, the range of the random numbers increases after each collision. The value of K_{max} is usually chosen as 15.

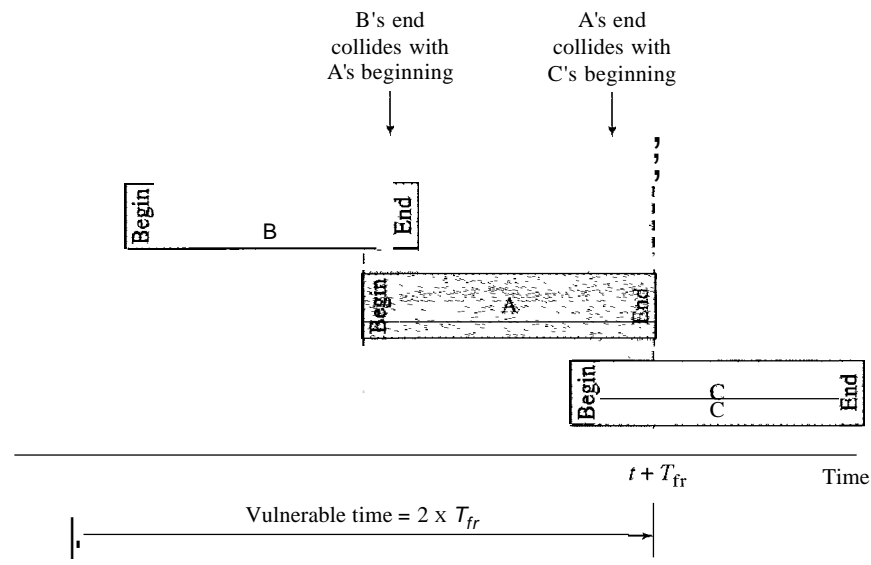
Example 12.1

The stations on a wireless ALOHA network are a maximum of 600 km apart. If we assume that signals propagate at 3×10^8 m/s, we find $T_p = (600 \times 10^3) / (3 \times 10^8) = 2$ ms. Now we can find the value of T_B for different values of K .

- For $K = 1$, the range is $\{0, 1\}$. The station needs to generate a random number with a value of 0 or 1. This means that T_B is either 0 ms (0×2) or 2 ms (1×2), based on the outcome of the random variable.
- For $K = 2$, the range is $\{0, 1, 2, 3\}$. This means that T_B can be 0, 2, 4, or 6 ms, based on the outcome of the random variable.
- For $K = 3$, the range is $\{0, 1, 2, 3, 4, 5, 6, 7\}$. This means that T_B can be 0, 2, 4, ..., 14 ms, based on the outcome of the random variable.
- We need to mention that if $K > 10$, it is normally set to 10.

Vulnerable time Let us find the length of time, the **vulnerable time**, in which there is a possibility of collision. We assume that the stations send fixed-length frames with each frame taking T_{fr} s to send. Figure 12.5 shows the vulnerable time for station A.

Figure 12.5 Vulnerable time for pure ALOHA protocol



Station A sends a frame at time t . Now imagine station B has already sent a frame between $t - T_{fr}$ and t . This leads to a collision between the frames from station A and station B. The end of B's frame collides with the beginning of A's frame. On the other hand, suppose that station C sends a frame between t and $t + T_{fr}$. Here, there is a collision between frames from station A and station C. The beginning of C's frame collides with the end of A's frame.

Looking at Figure 12.5, we see that the vulnerable time, during which a collision may occur in pure ALOHA, is 2 times the frame transmission time.

$$\text{Pure ALOHA vulnerable time} = 2 \times T_{fr}$$

Example 12.2

A pure ALOHA network transmits 200-bit frames on a shared channel of 200 kbps. What is the requirement to make this frame collision-free?

Solution

Average frame transmission time T_{fr} is 200 bits/200 kbps or 1 ms. The vulnerable time is $2 \times 1 \text{ ms} = 2 \text{ ms}$. This means no station should send later than 1 ms before this station starts transmission and no station should start sending during the one 1-ms period that this station is sending.

Throughput Let us call G the average number of frames generated by the system during one frame transmission time. Then it can be proved that the average number of successful transmissions for pure ALOHA is $S = G \times e^{-2G}$. The maximum throughput S_{\max} is 0.184, for $G = \frac{1}{2}$. In other words, if one-half a frame is generated during one frame transmission time (in other words, one frame during two frame transmission times), then 18.4 percent of these frames reach their destination successfully. This is an expected result because the vulnerable time is 2 times the frame transmission time. Therefore, if a station generates only one frame in this vulnerable time (and no other stations generate a frame during this time), the frame will reach its destination successfully.

The throughput for pure ALOHA is $S = G \times e^{-2G}$.

The maximum throughput $S_{\max} = 0.184$ when $G = (1/2)$.

Example 12.3

A pure ALOHA network transmits 200-bit frames on a shared channel of 200 kbps. What is the throughput if the system (all stations together) produces

- a. 1000 frames per second
- b. 500 frames per second
- c. 250 frames per second

Solution

The frame transmission time is $200/200$ kbps or 1 ms.

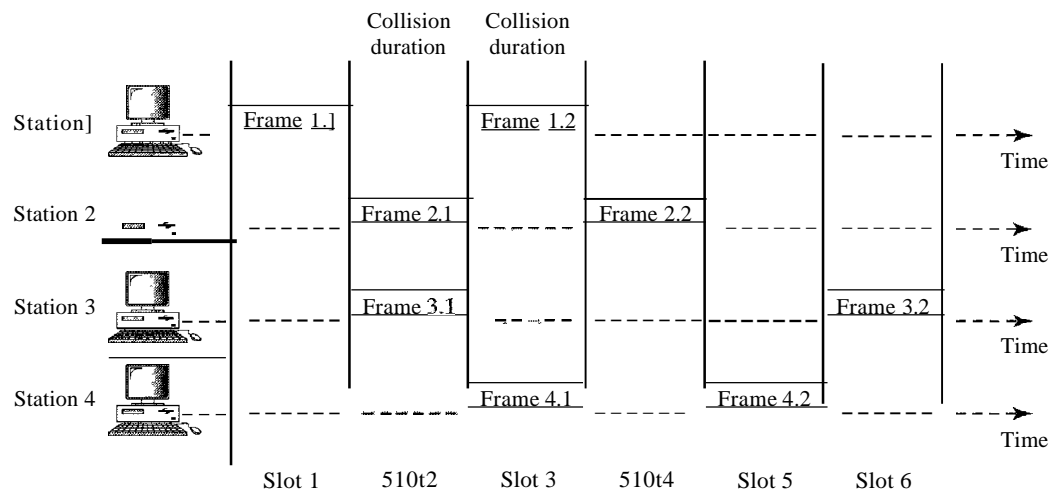
- a. If the system creates 1000 frames per second, this is 1 frame per millisecond. The load is 1. In this case $S = G \times e^{-2G}$ or $S = 0.135$ (13.5 percent). This means that the throughput is $1000 \times 0.135 = 135$ frames. Only 135 frames out of 1000 will probably survive.
- b. If the system creates 500 frames per second, this is $(1/2)$ frame per millisecond. The load is $(1/2)$. In this case $S = G \times e^{-2G}$ or $S = 0.184$ (18.4 percent). This means that the throughput is $500 \times 0.184 = 92$ and that only 92 frames out of 500 will probably survive. Note that this is the maximum throughput case, percentagewise.
- c. If the system creates 250 frames per second, this is $(1/4)$ frame per millisecond. The load is $(1/4)$. In this case $S = G \times e^{-2G}$ or $S = 0.152$ (15.2 percent). This means that the throughput is $250 \times 0.152 = 38$. Only 38 frames out of 250 will probably survive.

Slotted ALOHA

Pure ALOHA has a vulnerable time of $2 \times T_{fr}$. This is so because there is no rule that defines when the station can send. A station may send soon after another station has started or soon before another station has finished. Slotted ALOHA was invented to improve the efficiency of pure ALOHA.

In slotted ALOHA we divide the time into slots of T_{fr} s and force the station to send only at the beginning of the time slot. Figure 12.6 shows an example of frame collisions in slotted ALOHA.

Figure 12.6 Frames in a slotted ALOHA network



Because a station is allowed to send only at the beginning of the synchronized time slot, if a station misses this moment, it must wait until the beginning of the next time slot. This means that the station which started at the beginning of this slot has already finished sending its frame. Of course, there is still the possibility of collision if two stations try to send at the beginning of the same time slot. However, the vulnerable time is now reduced to one-half, equal to T_{fr} . Figure 12.7 shows the situation.

Figure 12.7 shows that the vulnerable time for slotted ALOHA is one-half that of pure ALOHA.

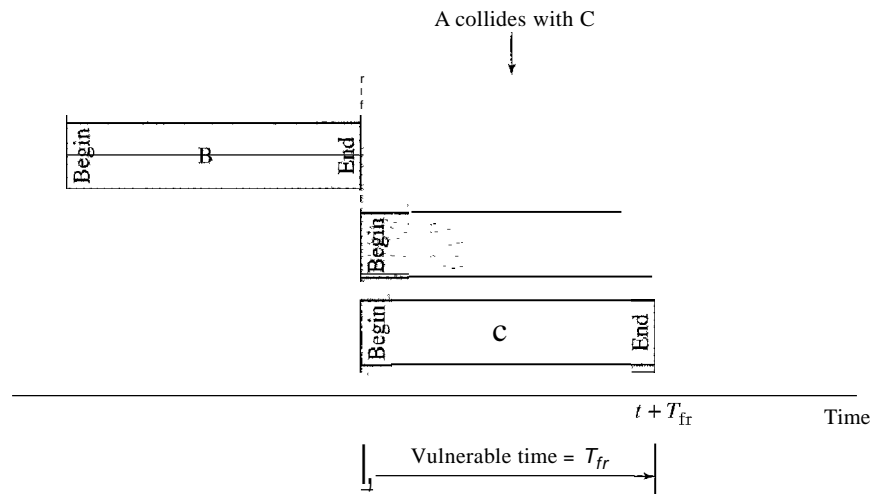
$$\text{Slotted ALOHA vulnerable time} = T_{fr}$$

Throughput It can be proved that the average number of successful transmissions for slotted ALOHA is $S = G \times e^{-G}$. The maximum throughput S_{max} is 0.368, when $G = 1$. In other words, if a frame is generated during one frame transmission time, then 36.8 percent of these frames reach their destination successfully. This result can be expected because the vulnerable time is equal to the frame transmission time. Therefore, if a station generates only one frame in this vulnerable time (and no other station generates a frame during this time), the frame will reach its destination successfully.

The throughput for slotted ALOHA is $S = G \times e^{-G}$.

The maximum throughput $S_{max} = 0.368$ when $G = 1$.

Figure 12.7 Vulnerable time for slotted ALOHA protocol

**Example 12.4**

A slotted ALOHA network transmits 200-bit frames using a shared channel with a 200-kbps bandwidth. Find the throughput if the system (all stations together) produces

- 1000 frames per second
- 500 frames per second
- 250 frames per second

Solution

This situation is similar to the previous exercise except that the network is using slotted ALOHA instead of pure ALOHA. The frame transmission time is $200/200$ kbps or 1 ms.

- In this case G is 1. So $S = G \times e^{-G}$ or $S = 0.368$ (36.8 percent). This means that the throughput is $1000 \times 0.368 = 368$ frames. Only 368 out of 1000 frames will probably survive. Note that this is the maximum throughput case, percentage-wise.
- Here G is $\frac{1}{2}$. In this case $S = G \times e^{-G}$ or $S = 0.303$ (30.3 percent). This means that the throughput is $500 \times 0.303 = 151$. Only 151 frames out of 500 will probably survive.
- Now G is $\frac{1}{4}$. In this case $S = G \times e^{-G}$ or $S = 0.195$ (19.5 percent). This means that the throughput is $250 \times 0.195 = 49$. Only 49 frames out of 250 will probably survive.

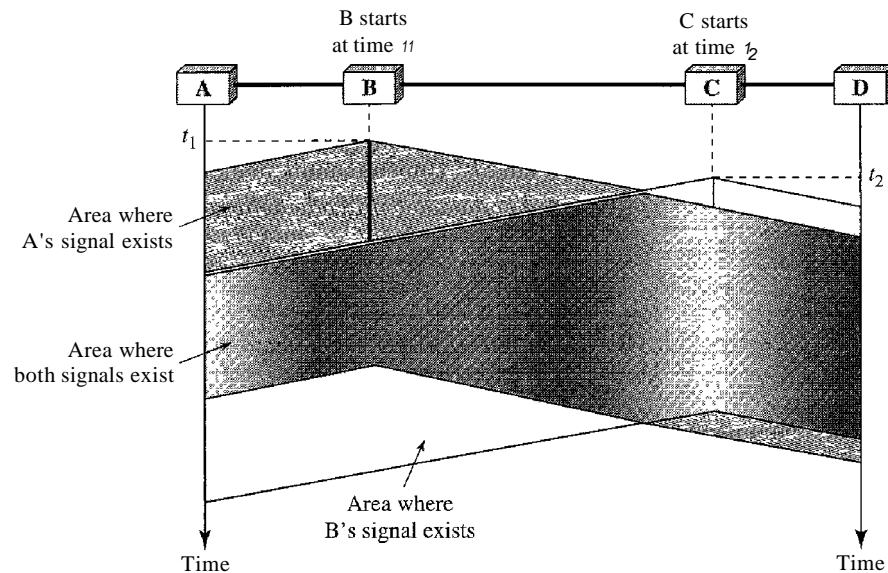
Carrier Sense Multiple Access (CSMA)

To minimize the chance of collision and, therefore, increase the performance, the CSMA method was developed. The chance of collision can be reduced if a station senses the medium before trying to use it. Carrier sense multiple access (CSMA) requires that each station first listen to the medium (or check the state of the medium) before sending. In other words, CSMA is based on the principle "sense before transmit" or "listen before talk."

CSMA can reduce the possibility of collision, but it cannot eliminate it. The reason for this is shown in Figure 12.8, a space and time model of a CSMA network. Stations are connected to a shared channel (usually a dedicated medium).

The possibility of collision still exists because of propagation delay; when a station sends a frame, it still takes time (although very short) for the first bit to reach every station

Figure 12.8 Space/time model of the collision in CSMA



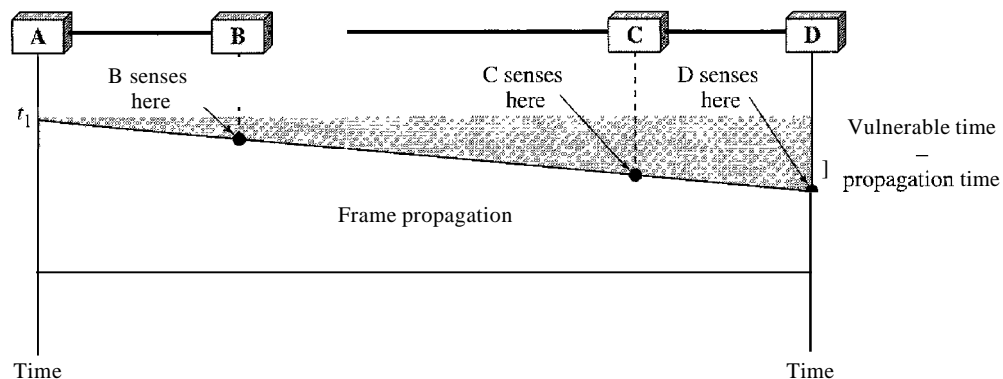
and for every station to sense it. In other words, a station may sense the medium and find it idle, only because the first bit sent by another station has not yet been received.

At time t_1' station B senses the medium and finds it idle, so it sends a frame. At time t_2 ($t_2 > t_1'$) station C senses the medium and finds it idle because, at this time, the first bits from station B have not reached station C. Station C also sends a frame. The two signals collide and both frames are destroyed.

Vulnerable Time

The vulnerable time for CSMA is the propagation time T_p . This is the time needed for a signal to propagate from one end of the medium to the other. When a station sends a frame, and any other station tries to send a frame during this time, a collision will result. But if the first bit of the frame reaches the end of the medium, every station will already have heard the bit and will refrain from sending. Figure 12.9 shows the worst

Figure 12.9 Vulnerable time in CSMA



case. The leftmost station A sends a frame at time tI' which reaches the rightmost station D at time $tI + T_p$. The gray area shows the vulnerable area in time and space.

Persistence Methods

What should a station do if the channel is busy? What should a station do if the channel is idle? Three methods have been devised to answer these questions: the I-persistent method, the nonpersistent method, and the p-persistent method. Figure 12.10 shows the behavior of three persistence methods when a station finds a channel busy.

Figure 12.10 Behavior of three persistence methods

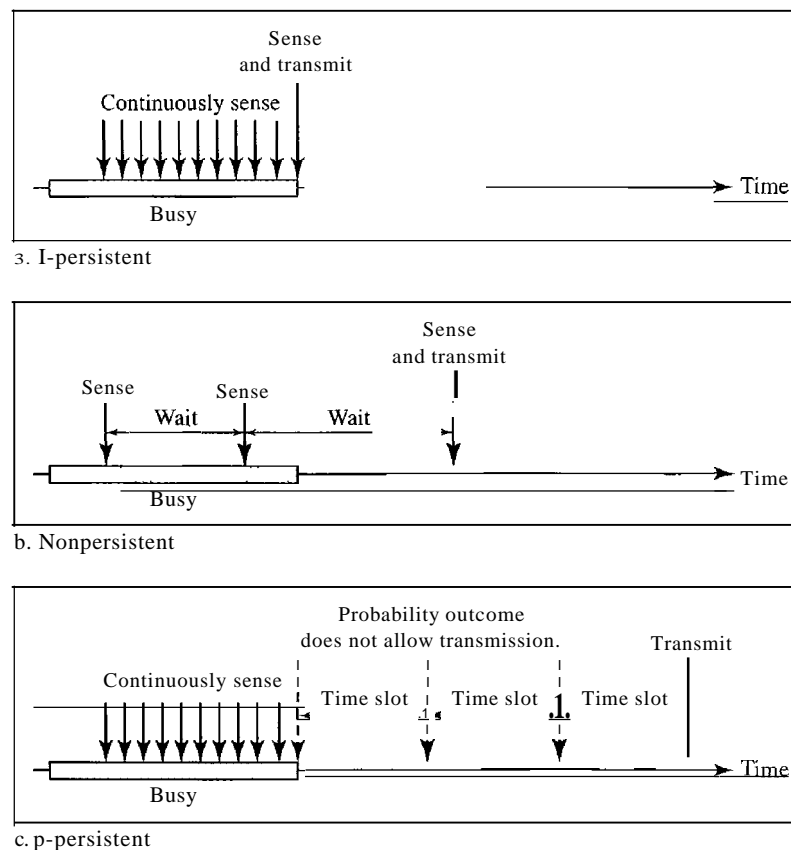
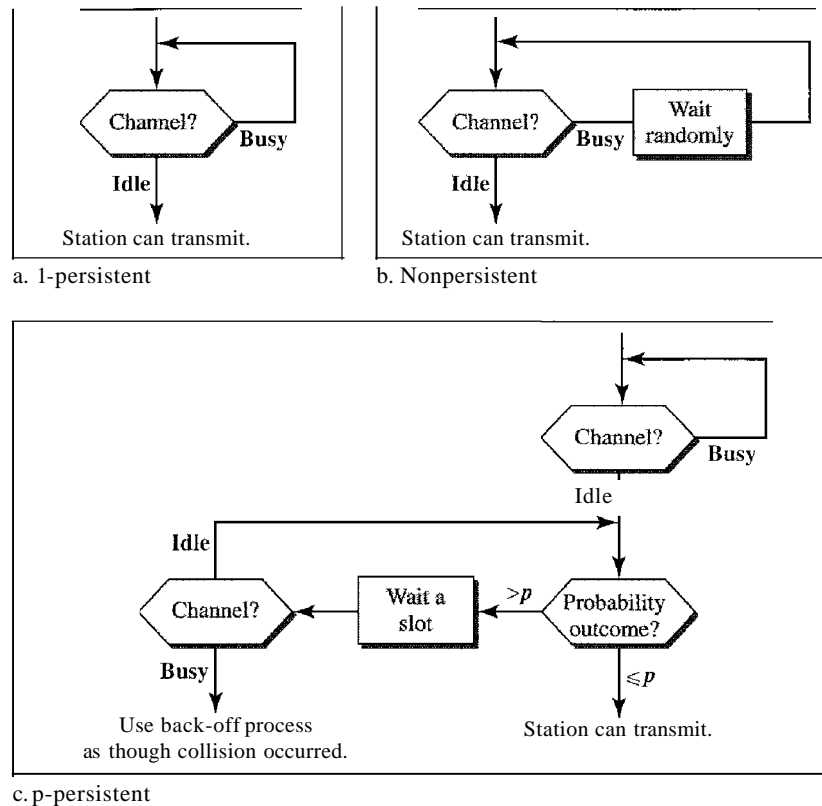


Figure 12.11 shows the flow diagrams for these methods.

I-Persistent The **I-persistent method** is simple and straightforward. In this method, after the station finds the line idle, it sends its frame immediately (with probability I). This method has the highest chance of collision because two or more stations may find the line idle and send their frames immediately. We will see in Chapter 13 that Ethernet uses this method.

Nonpersistent In the **nonpersistent method**, a station that has a frame to send senses the line. If the line is idle, it sends immediately. If the line is not idle, it waits a

Figure 12.11 Flow diagram for three persistence methods

random amount of time and then senses the line again. The nonpersistent approach reduces the chance of collision because it is unlikely that two or more stations will wait the same amount of time and retry to send simultaneously. However, this method reduces the efficiency of the network because the medium remains idle when there may be stations with frames to send.

p-Persistent The **p-persistent method** is used if the channel has time slots with a slot duration equal to or greater than the maximum propagation time. The p-persistent approach combines the advantages of the other two strategies. It reduces the chance of collision and improves efficiency. In this method, after the station finds the line idle it follows these steps:

1. With probability p , the station sends its frame.
2. With probability $q = 1 - p$, the station waits for the beginning of the next time slot and checks the line again.
 - a. If the line is idle, it goes to step 1.
 - b. If the line is busy, it acts as though a collision has occurred and uses the back-off procedure.

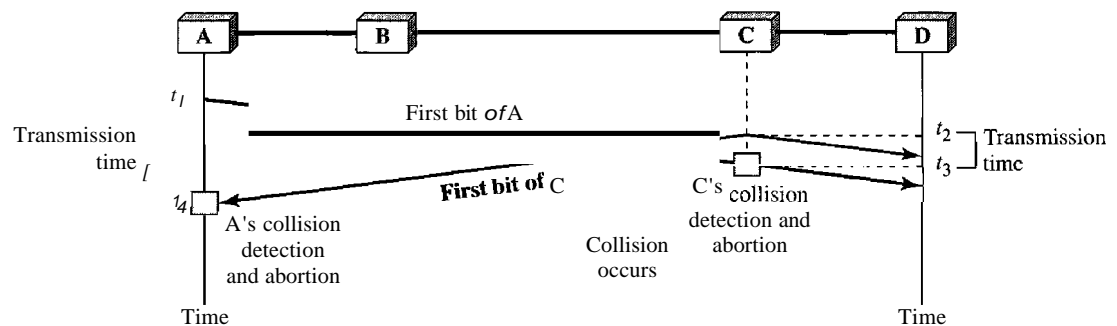
Carrier Sense Multiple Access with Collision Detection (CSMA/CD)

The CSMA method does not specify the procedure following a collision. Carrier sense multiple access with collision detection (CSMA/CD) augments the algorithm to handle the collision.

In this method, a station monitors the medium after it sends a frame to see if the transmission was successful. If so, the station is finished. If, however, there is a collision, the frame is sent again.

To better understand CSMA/CD, let us look at the first bits transmitted by the two stations involved in the collision. Although each station continues to send bits in the frame until it detects the collision, we show what happens as the first bits collide. In Figure 12.12, stations A and C are involved in the collision.

Figure 12.12 Collision of the first bit in CSMA/CD

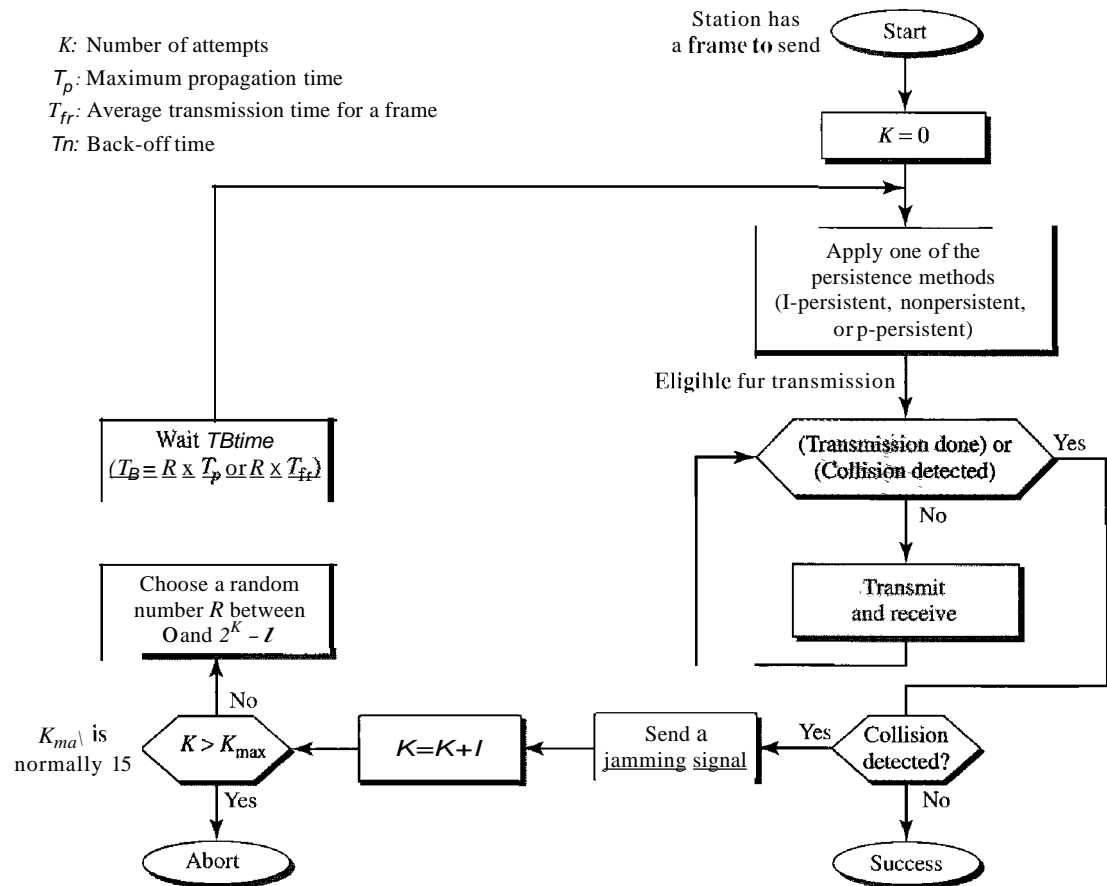


At time t_1 , station A has executed its persistence procedure and starts sending the bits of its frame. At time t_2 , station C has not yet sensed the first bit sent by A. Station C executes its persistence procedure and starts sending the bits in its frame, which propagate both to the left and to the right. The collision occurs sometime after time t_2 . Station C detects a collision at time t_3 when it receives the first bit of A's frame. Station C immediately (or after a short time, but we assume immediately) aborts transmission. Station A detects collision at time t_4 when it receives the first bit of C's frame; it also immediately aborts transmission. Looking at the figure, we see that A transmits for the duration $t_4 - t_1$; C transmits for the duration $t_3 - t_2$. Later we show that, for the protocol to work, the length of any frame divided by the bit rate in this protocol must be more than either of these durations. At time t_4 , the transmission of A's frame, though incomplete, is aborted; at time t_3 , the transmission of B's frame, though incomplete, is aborted.

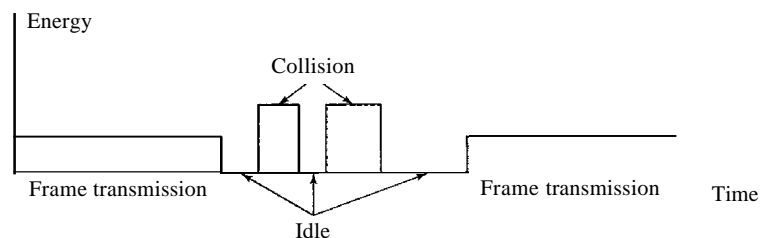
Now that we know the time durations for the two transmissions, we can show a more complete graph in Figure 12.13.

Minimum Frame Size

For CSMA/CD to work, we need a restriction on the frame size. Before sending the last bit of the frame, the sending station must detect a collision, if any, and abort the transmission. This is so because the station, once the entire frame is sent, does not keep a copy of the frame and does not monitor the line for collision detection. Therefore, the frame transmission time T_{fr} must be at least two times the maximum propagation time T_p . To understand the reason, let us think about the worst-case scenario. If the two stations involved in a collision are the maximum distance apart, the signal from the first takes time T_p to reach the second, and the effect of the collision takes another time T_p to reach the first. So the requirement is that the first station must still be transmitting after $2T_p$.

Figure 12.14 Flow diagram for the CSMA/CD

successfully captured the channel and is sending its frame. At the abnormal level, there is a collision and the level of the energy is twice the normal level. A station that has a frame to send or is sending a frame needs to monitor the energy level to determine if the channel is idle, busy, or in collision mode. Figure 12.15 shows the situation.

Figure 12.15 Energy level during transmission, idleness, or collision

Throughput

The throughput of CSMA/CD is greater than that of pure or slotted ALOHA. The maximum throughput occurs at a different value of G and is based on the persistence method

and the value of p in the p-persistent approach. For I-persistent method the maximum throughput is around 50 percent when $G = 1$. For nonpersistent method, the maximum throughput can go up to 90 percent when G is between 3 and 8.

Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA)

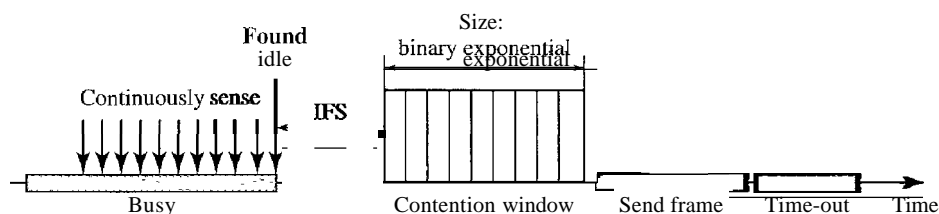
The basic idea behind *CSMA/CD* is that a station needs to be able to receive while transmitting to detect a collision. When there is no collision, the station receives one signal: its own signal. When there is a collision, the station receives two signals: its own signal and the signal transmitted by a second station. To distinguish between these two cases, the received signals in these two cases must be significantly different. In other words, the signal from the second station needs to add a significant amount of energy to the one created by the first station.

In a wired network, the received signal has almost the same energy as the sent signal because either the length of the cable is short or there are repeaters that amplify the energy between the sender and the receiver. This means that in a collision, the detected energy almost doubles.

However, in a wireless network, much of the sent energy is lost in transmission. The received signal has very little energy. Therefore, a collision may add only 5 to 10 percent additional energy. This is not useful for effective collision detection.

We need to avoid collisions on wireless networks because they cannot be detected. Carrier sense multiple access with collision avoidance (*CSMA/CA*) was invented for this network. Collisions are avoided through the use of CSMA/CA's three strategies: the inter-frame space, the contention window, and acknowledgments, as shown in Figure 12.16.

Figure 12.16 Timing in CSMA/CA



Interframe Space (IFS)

First, collisions are avoided by deferring transmission even if the channel is found idle. When an idle channel is found, the station does not send immediately. It waits for a period of time called the interframe space or IFS. Even though the channel may appear idle when it is sensed, a distant station may have already started transmitting. The distant station's signal has not yet reached this station. The IFS time allows the front of the transmitted signal by the distant station to reach this station. If after the IFS time the channel is still idle, the station can send, but it still needs to wait a time equal to the contention time (described next). The IFS variable can also be used to prioritize

stations or frame types. For example, a station that is assigned a shorter IFS has a higher priority.

In CSMA/CA, the IFS can also be used to define
the priority of a station or a frame.

Contention Window

The contention window is an amount of time divided into slots. A station that is ready to send chooses a random number of slots as its wait time. The number of slots in the window changes according to the binary exponential back-off strategy. This means that it is set to one slot the first time and then doubles each time the station cannot detect an idle channel after the IFS time. This is very similar to the p-persistent method except that a random outcome defines the number of slots taken by the waiting station. One interesting point about the contention window is that the station needs to sense the channel after each time slot. However, if the station finds the channel busy, it does not restart the process; it just stops the timer and restarts it when the channel is sensed as idle. This gives priority to the station with the longest waiting time.

In CSMA/CA, if the station finds the channel busy,
it does not restart the timer of the contention window;
it stops the timer and restarts it when the channel becomes idle.

Acknowledgment

With all these precautions, there still may be a collision resulting in destroyed data. In addition, the data may be corrupted during the transmission. The positive acknowledgment and the time-out timer can help guarantee that the receiver has received the frame.

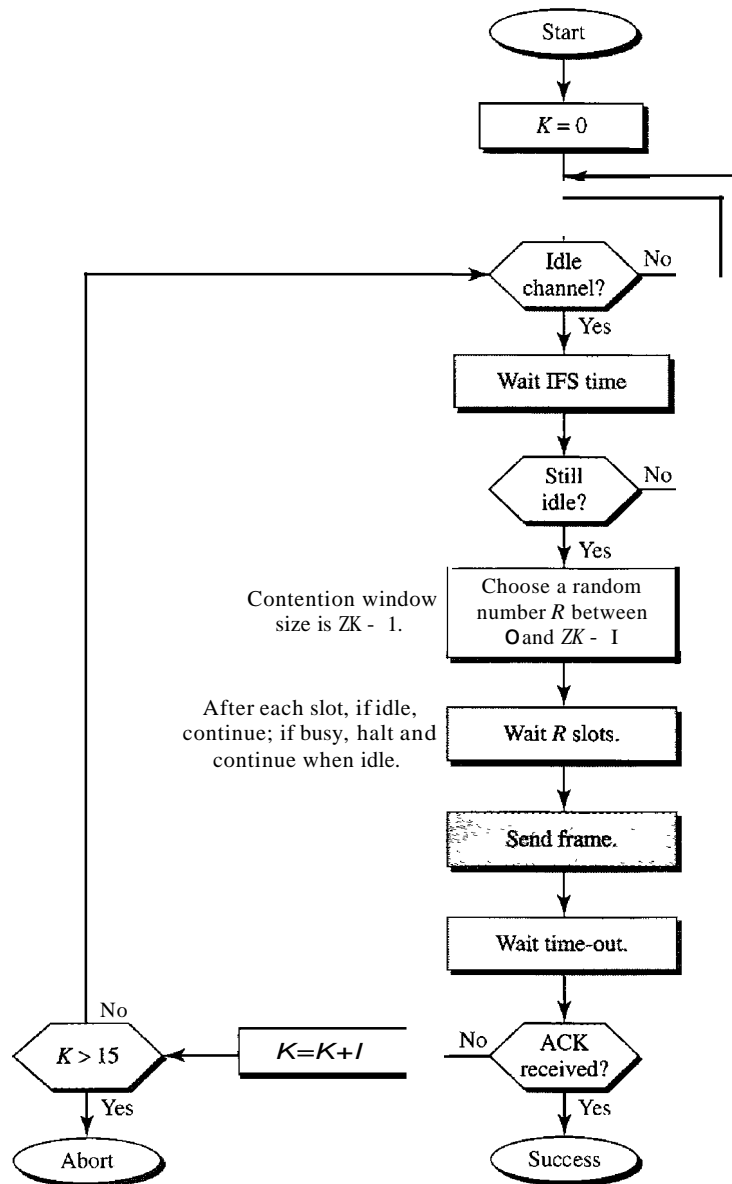
Procedure

Figure 12.17 shows the procedure. Note that the channel needs to be sensed before and after the IFS. The channel also needs to be sensed during the contention time. For each time slot of the contention window, the channel is sensed. If it is found idle, the timer continues; if the channel is found busy, the timer is stopped and continues after the timer becomes idle again.

CSMA/CA and Wireless Networks

CSMA/CA was mostly intended for use in wireless networks. The procedure described above, however, is not sophisticated enough to handle some particular issues related to wireless networks, such as hidden terminals or exposed terminals. We will see how these issues are solved by augmenting the above protocol with hand-shaking features. The use of CSMA/CA in wireless networks will be discussed in Chapter 14.

Figure 12.17 Flow diagram for CSMA/CA



12.2 CONTROLLED ACCESS

In controlled access, the stations consult one another to find which station has the right to send. A station cannot send unless it has been authorized by other stations. We discuss three popular controlled-access methods.

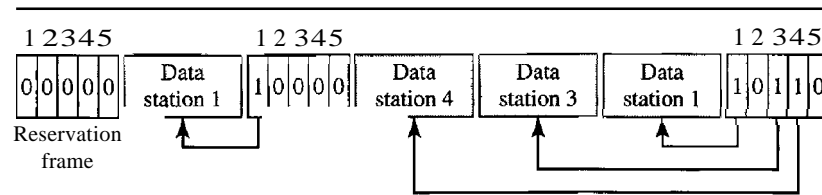
Reservation

In the reservation method, a station needs to make a reservation before sending data. Time is divided into intervals. In each interval, a reservation frame precedes the data frames sent in that interval.

If there are N stations in the system, there are exactly N reservation minislots in the reservation frame. Each minislot belongs to a station. When a station needs to send a data frame, it makes a reservation in its own minislot. The stations that have made reservations can send their data frames after the reservation frame.

Figure 12.18 shows a situation with five stations and a five-minislot reservation frame. In the first interval, only stations 1, 3, and 4 have made reservations. In the second interval, only station 1 has made a reservation.

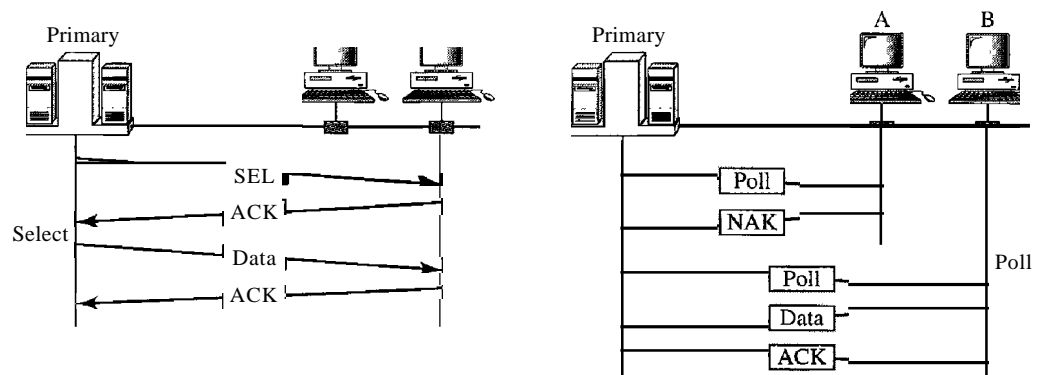
Figure 12.18 Reservation access method



Polling

Polling works with topologies in which one device is designated as a primary station and the other devices are secondary stations. All data exchanges must be made through the primary device even when the ultimate destination is a secondary device. The primary device controls the link; the secondary devices follow its instructions. It is up to the primary device to determine which device is allowed to use the channel at a given time. The primary device, therefore, is always the initiator of a session (see Figure 12.19).

Figure 12.19 Select and poll functions in polling access method



If the primary wants to receive data, it asks the secondaries if they have anything to send; this is called poll function. If the primary wants to send data, it tells the secondary to get ready to receive; this is called select function.

Select

The *select* function is used whenever the primary device has something to send. Remember that the primary controls the link. If the primary is neither sending nor receiving data, it knows the link is available.

If it has something to send, the primary device sends it. What it does not know, however, is whether the target device is prepared to receive. So the primary must alert the secondary to the upcoming transmission and wait for an acknowledgment of the secondary's ready status. Before sending data, the primary creates and transmits a select (SEL) frame, one field of which includes the address of the intended secondary.

Poll

The *poll* function is used by the primary device to solicit transmissions from the secondary devices. When the primary is ready to receive data, it must ask (poll) each device in turn if it has anything to send. When the first secondary is approached, it responds either with a NAK frame if it has nothing to send or with data (in the form of a data frame) if it does. If the response is negative (a NAK frame), then the primary polls the next secondary in the same manner until it finds one with data to send. When the response is positive (a data frame), the primary reads the frame and returns an acknowledgment (ACK frame), verifying its receipt.

Token Passing

In the token-passing method, the stations in a network are organized in a logical ring. In other words, for each station, there is a *predecessor* and a *successor*. The predecessor is the station which is logically before the station in the ring; the successor is the station which is after the station in the ring. The current station is the one that is accessing the channel now. The right to this access has been passed from the predecessor to the current station. The right will be passed to the successor when the current station has no more data to send.

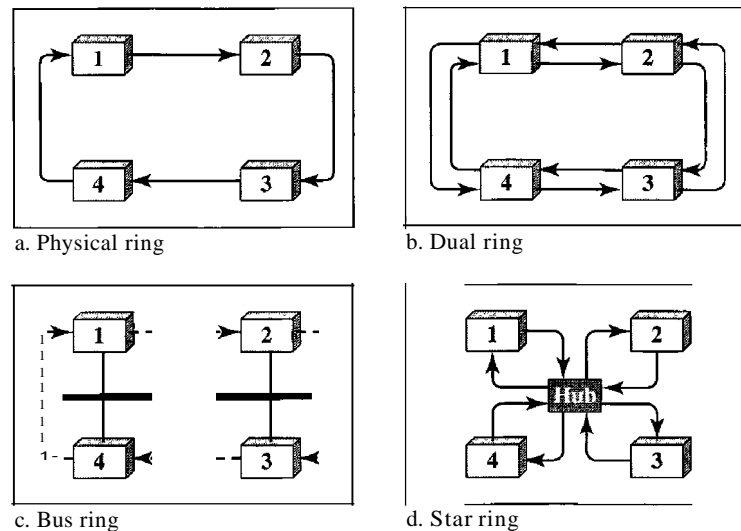
But how is the right to access the channel passed from one station to another? In this method, a special packet called a token circulates through the ring. The possession of the token gives the station the right to access the channel and send its data. When a station has some data to send, it waits until it receives the token from its predecessor. It then holds the token and sends its data. When the station has no more data to send, it releases the token, passing it to the next logical station in the ring. The station cannot send data until it receives the token again in the next round. In this process, when a station receives the token and has no data to send, it just passes the data to the next station.

Token management is needed for this access method. Stations must be limited in the time they can have possession of the token. The token must be monitored to ensure it has not been lost or destroyed. For example, if a station that is holding the token fails, the token will disappear from the network. Another function of token management is to assign priorities to the stations and to the types of data being transmitted. And finally, token management is needed to make low-priority stations release the token to high-priority stations.

Logical Ring

In a token-passing network, stations do not have to be physically connected in a ring; the ring can be a logical one. Figure 12.20 show four different physical topologies that can create a logical ring.

Figure 12.20 *Logical ring and physical topology in token-passing access method*



In the physical ring topology, when a station sends the token to its successor, the token cannot be seen by other stations; the successor is the next one in line. This means that the token does not have to have the address of the next successor. The problem with this topology is that if one of the links—the medium between two adjacent stations—fails, the whole system fails.

The dual ring topology uses a second (auxiliary) ring which operates in the reverse direction compared with the main ring. The second ring is for emergencies only (such as a spare tire for a car). If one of the links in the main ring fails, the system automatically combines the two rings to form a temporary ring. After the failed link is restored, the auxiliary ring becomes idle again. Note that for this topology to work, each station needs to have two transmitter ports and two receiver ports. The high-speed Token Ring networks called FDDI (Fiber Distributed Data Interface) and CDDI (Copper Distributed Data Interface) use this topology.

In the bus ring topology, also called a token bus, the stations are connected to a single cable called a bus. They, however, make a logical ring, because each station knows the address of its successor (and also predecessor for token management purposes). When a station has finished sending its data, it releases the token and inserts the address of its successor in the token. Only the station with the address matching the destination address of the token gets the token to access the shared media. The Token Bus LAN, standardized by IEEE, uses this topology.

In a star ring topology, the physical topology is a star. There is a hub, however, that acts as the connector. The wiring inside the hub makes the ring; the stations are connected to this ring through the two wire connections. This topology makes the network

less prone to failure because if a link goes down, it will be bypassed by the hub and the rest of the stations can operate. Also adding and removing stations from the ring is easier. This topology is still used in the Token Ring LAN designed by IBM.

12.3 CHANNELIZATION

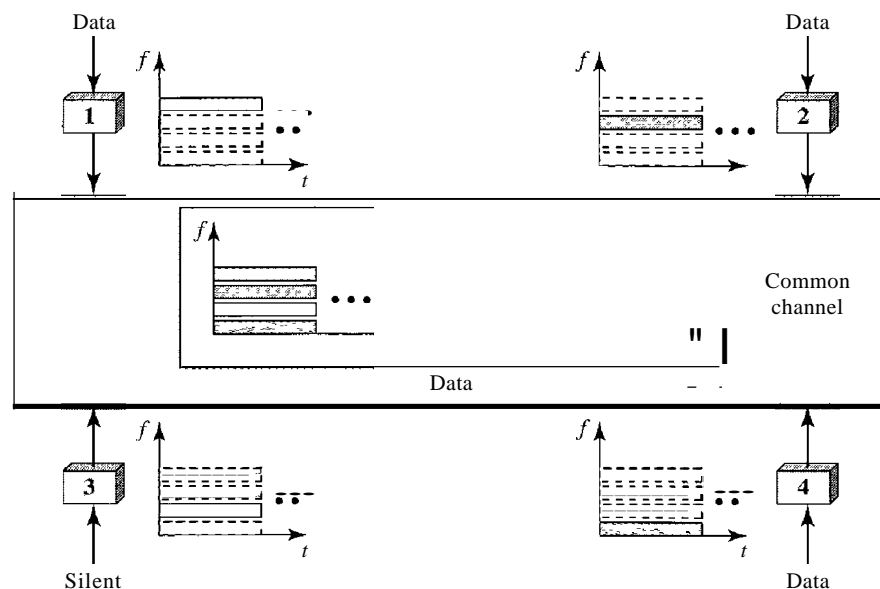
Channelization is a multiple-access method in which the available bandwidth of a link is shared in time, frequency, or through code, between different stations. In this section, we discuss three channelization protocols: FDMA, TDMA, and CDMA.

We see the application of all these methods in Chapter 16
when we discuss cellular phone systems.

Frequency-Division Multiple Access (FDMA)

In frequency-division multiple access (FDMA), the available bandwidth is divided into frequency bands. Each station is allocated a band to send its data. In other words, each band is reserved for a specific station, and it belongs to the station all the time. Each station also uses a bandpass filter to confine the transmitter frequencies. To prevent station interferences, the allocated bands are separated from one another by small *guard bands*. Figure 12.21 shows the idea of FDMA.

Figure 12.21 *Frequency-division multiple access (FDMA)*



In FDMA, the available bandwidth of the common channel
is divided into bands that are separated by guard bands.

FDMA specifies a predetermined frequency band for the entire period of communication. This means that stream data (a continuous flow of data that may not be packetized) can easily be used with FDMA. We will see in Chapter 16 how this feature can be used in cellular telephone systems.

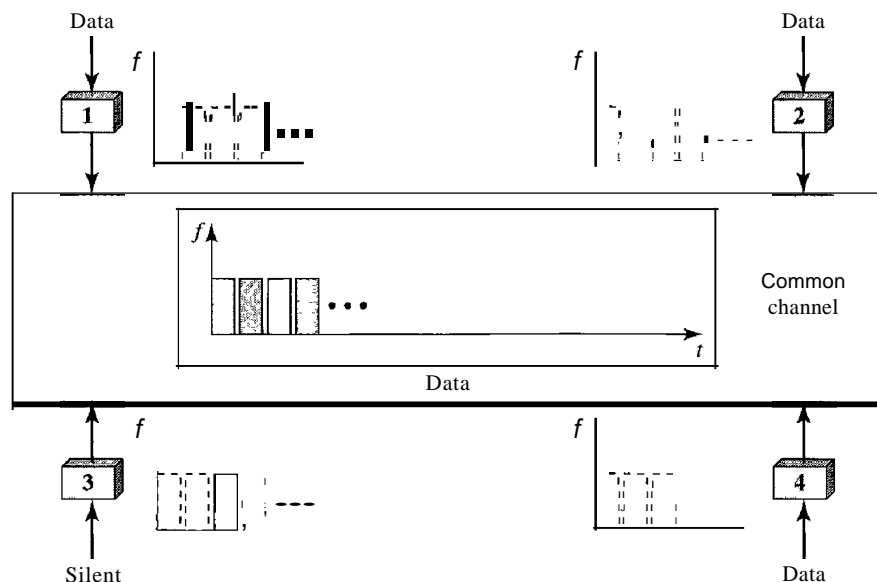
We need to emphasize that although FDMA and FDM conceptually seem similar, there are differences between them. FDM, as we saw in Chapter 6, is a physical layer technique that combines the loads from low-bandwidth channels and transmits them by using a high-bandwidth channel. The channels that are combined are low-pass. The multiplexer modulates the signals, combines them, and creates a bandpass signal. The bandwidth of each channel is shifted by the multiplexer.

FDMA, on the other hand, is an access method in the data link layer. The data link layer in each station tells its physical layer to make a bandpass signal from the data passed to it. The signal must be created in the allocated band. There is no physical multiplexer at the physical layer. The signals created at each station are automatically bandpass-filtered. They are mixed when they are sent to the common channel.

Time-Division Multiple Access (TDMA)

In time-division multiple access (TDMA), the stations share the bandwidth of the channel in time. Each station is allocated a time slot during which it can send data. Each station transmits its data in its assigned time slot. Figure 12.22 shows the idea behind TDMA.

Figure 12.22 Time-division multiple access (TDMA)



The main problem with TDMA lies in achieving synchronization between the different stations. Each station needs to know the beginning of its slot and the location of its slot. This may be difficult because of propagation delays introduced in the system if the stations are spread over a large area. To compensate for the delays, we can insert *guard*

times. Synchronization is normally accomplished by having some synchronization bits (normally referred to as preamble bits) at the beginning of each slot.

In TDMA, the bandwidth is just one channel that is
timeshared between different stations.

We also need to emphasize that although TDMA and TDM conceptually seem the same, there are differences between them. TDM, as we saw in Chapter 6, is a physical layer technique that combines the data from slower channels and transmits them by using a faster channel. The process uses a physical multiplexer that interleaves data units from each channel.

TDMA, on the other hand, is an access method in the data link layer. The data link layer in each station tells its physical layer to use the allocated time slot. There is no physical multiplexer at the physical layer.

Code-Division Multiple Access (CDMA)

Code-division multiple access (CDMA) was conceived several decades ago. Recent advances in electronic technology have finally made its implementation possible. CDMA differs from FDMA because only one channel occupies the entire bandwidth of the link. It differs from TDMA because all stations can send data simultaneously; there is no timesharing.

In CDMA, one channel carries all transmissions simultaneously.

Analogy

Let us first give an analogy. CDMA simply means communication with different codes. For example, in a large room with many people, two people can talk in English if nobody else understands English. Another two people can talk in Chinese if they are the only ones who understand Chinese, and so on. In other words, the common channel, the space of the room in this case, can easily allow communication between several couples, but in different languages (codes).

Idea

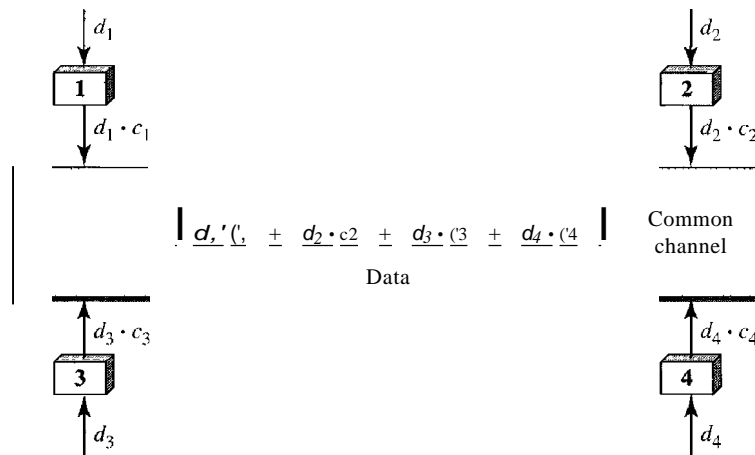
Let us assume we have four stations 1, 2, 3, and 4 connected to the same channel. The data from station 1 are d_1 , from station 2 are d_2 , and so on. The code assigned to the first station is c_1 , to the second is c_2 , and so on. We assume that the assigned codes have two properties.

1. If we multiply each code by another, we get 0.
2. If we multiply each code by itself, we get 4 (the number of stations).

With these two properties in mind, let us see how the above four stations can send data using the same common channel, as shown in Figure 12.23.

Station 1 multiplies (a special kind of multiplication, as we will see) its data by its code to get $d_1 \cdot c_1$. Station 2 multiplies its data by its code to get $d_2 \cdot c_2$. And so on. The

Figure 12.23 Simple idea of communication with code



data that go on the channel are the sum of all these terms, as shown in the box. Any station that wants to receive data from one of the other three multiplies the data on the channel by the code of the sender. For example, suppose stations 1 and 2 are talking to each other. Station 2 wants to hear what station 1 is saying. It multiplies the data on the channel by c_1 , the code of station 1.

Because $(c_1 \cdot c_1)$ is 4, but $(c_2 \cdot c_1)$, $(c_3 \cdot c_1)$, and $(c_4 \cdot c_1)$ are all 0s, station 2 divides the result by 4 to get the data from station 1.

$$\begin{aligned} \text{data} &= (d_1 \cdot c_1 + d_2 \cdot c_2 + d_3 \cdot c_3 + d_4 \cdot c_4) \cdot c_1 \\ &= d_1 \cdot c_1 \cdot c_1 + d_2 \cdot c_2 \cdot c_1 + d_3 \cdot c_3 \cdot c_1 + d_4 \cdot c_4 \cdot c_1 = 4 \times d_1 \end{aligned}$$

Chips

CDMA is based on coding theory. Each station is assigned a code, which is a sequence of numbers called chips, as shown in Figure 12.24. The codes are for the previous example.

Figure 12.24 Chip sequences



Later in this chapter we show how we chose these sequences. For now, we need to know that we did not choose the sequences randomly; they were carefully selected. They are called orthogonal sequences and have the following properties:

1. Each sequence is made of N elements, where N is the number of stations.

2. If we multiply a sequence by a number, every element in the sequence is multiplied by that element. This is called multiplication of a sequence by a scalar. For example,

$$2 \cdot [+1 +1 -1 -1] = [+2 +2 -2 -2]$$

3. If we multiply two equal sequences, element by element, and add the results, we get N , where N is the number of elements in the each sequence. This is called the inner product of two equal sequences. For example,

$$[+1 +1 -1 -1] \cdot [+1 +1 -1 -1] = 1 + 1 + 1 + 1 = 4$$

4. If we multiply two different sequences, element by element, and add the results, we get 0. This is called inner product of two different sequences. For example,

$$[+1 +1 -1 -1] \cdot [+1 +1 +1 +1] = 1 + 1 - 1 - 1 = 0$$

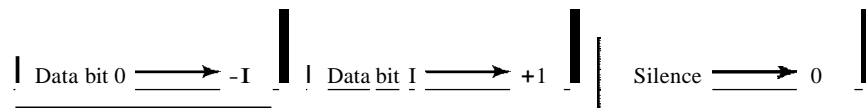
5. Adding two sequences means adding the corresponding elements. The result is another sequence. For example,

$$[+1 +1 -1 -1] + [+1 +1 +1 +1] = [+2 +2 \ 00]$$

Data Representation

We follow these rules for encoding: If a station needs to send a 0 bit, it encodes it as -1 ; if it needs to send a 1 bit, it encodes it as $+1$. When a station is idle, it sends no signal, which is interpreted as a 0. These are shown in Figure 12.25.

Figure 12.25 Data representation in CDMA



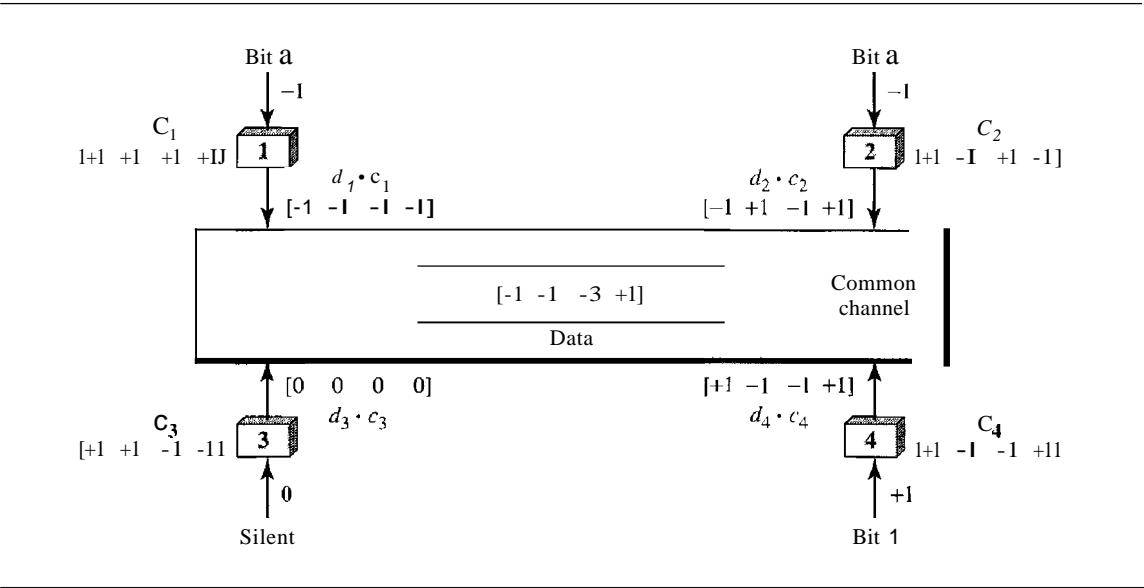
Encoding and Decoding

As a simple example, we show how four stations share the link during a 1-bit interval. The procedure can easily be repeated for additional intervals. We assume that stations 1 and 2 are sending a 0 bit and channel 4 is sending a 1 bit. Station 3 is silent. The data at the sender site are translated to -1 , -1 , 0 , and $+1$. Each station multiplies the corresponding number by its chip (its orthogonal sequence), which is unique for each station. The result is a new sequence which is sent to the channel. For simplicity, we assume that all stations send the resulting sequences at the same time. The sequence on the channel is the sum of all four sequences as defined before. Figure 12.26 shows the situation.

Now imagine station 3, which we said is silent, is listening to station 2. Station 3 multiplies the total data on the channel by the code for station 2, which is $[+1 -1 +1 -1]$, to get

$$[-1 -1 -3 +1] \cdot [+1 -1 +1 -1] = -4/4 = -1 \longrightarrow \text{bit 1}$$

Figure 12.26 Sharing channel in CDMA



Signal Level

The process can be better understood if we show the digital signal produced by each station and the data recovered at the destination (see Figure 12.27). The figure shows the corresponding signals for each station (using NRZ-L for simplicity) and the signal that is on the common channel.

Figure 12.27 Digital signal created by four stations in CDMA

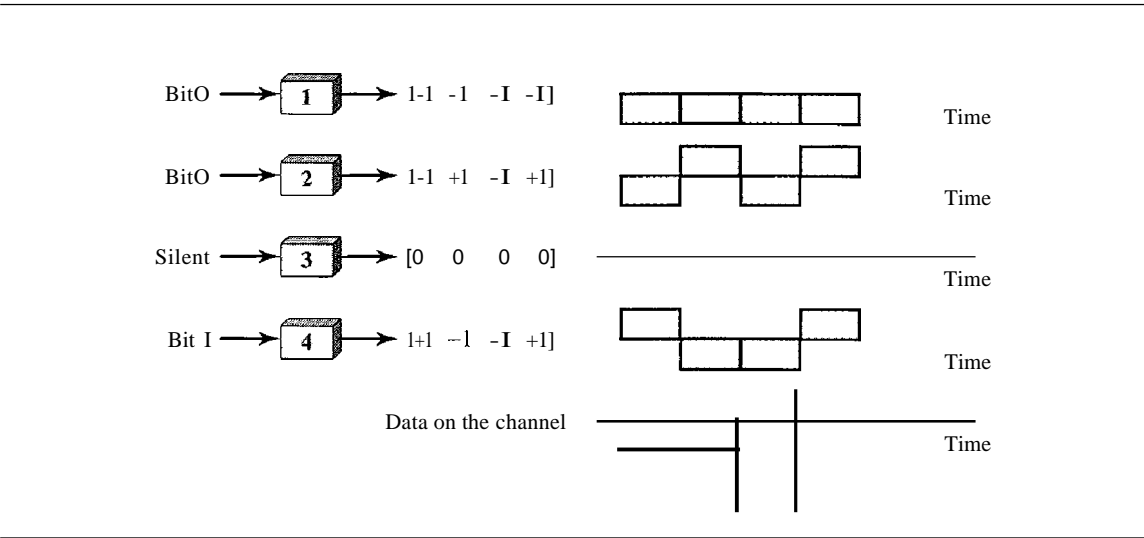
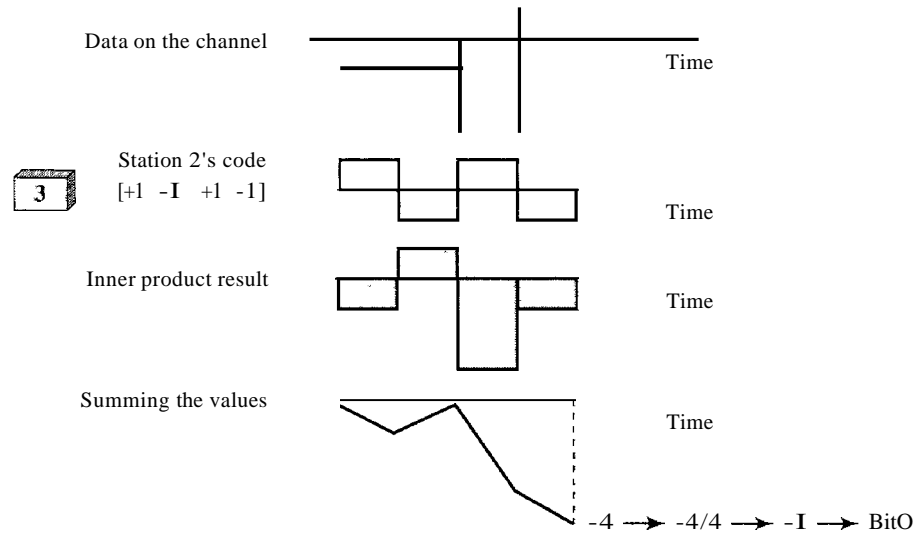


Figure 12.28 shows how station 3 can detect the data sent by station 2 by using the code for station 2. The total data on the channel are multiplied (inner product operation) by the signal representing station 2 chip code to get a new signal. The station then integrates and adds the area under the signal, to get the value -4, which is divided by 4 and interpreted as bit 0.

Figure 12.28 Decoding of the composite signal for one in CDMA

Sequence Generation

To generate chip sequences, we use a **Walsh table**, which is a two-dimensional table with an equal number of rows and columns, as shown in Figure 12.29.

Figure 12.29 General rule and examples of creating Walsh tables

$$W_1 = \begin{bmatrix} +1 \end{bmatrix} \qquad W_{2N} = \begin{bmatrix} W_N & W_N \\ W_N & \overline{W_N} \end{bmatrix}$$

a. Two basic rules

$$W_1 = \begin{bmatrix} +1 \end{bmatrix} \qquad W_2 = \begin{bmatrix} +1 & +1 \\ +1 & -1 \end{bmatrix} \qquad W_4 = \begin{bmatrix} +1 & +1 & +1 & +1 \\ +1 & -1 & +1 & -1 \\ +1 & +1 & -1 & -1 \\ +1 & -1 & -1 & +1 \end{bmatrix}$$

b. Generation of W_1 , W_2 , and W_4

In the Walsh table, each row is a sequence of chips. W_1 for a one-chip sequence has one row and one column. We can choose -1 or $+1$ for the chip for this trivial table (we chose $+1$). According to Walsh, if we know the table for N sequences W_N we can create the table for $2N$ sequences W_{2N} , as shown in Figure 12.29. The W_N with the overbar $\overline{W_N}$ stands for the complement of W_N where each $+1$ is changed to -1 and vice versa. Figure 12.29 also shows how we can create W_2 and W_4 from W_1 . After we select W_1 , W_2

can be made from four W_j 's, with the last one the complement of W_1 . After W_2 is generated, W_4 can be made of four W_2 's, with the last one the complement of W_2 . Of course, W_8 is composed of four W_4 's, and so on. Note that after W_N is made, each station is assigned a chip corresponding to a row.

Something we need to emphasize is that the number of sequences N needs to be a power of 2. In other words, we need to have $N = 2^m$.

The number of sequences in a Walsh table needs to be $N = 2^m$.

Example 12.6

Find the chips for a network with

- a. Two stations
- b. Four stations

Solution

We can use the rows of W_2 and W_4 in Figure 12.29:

- a. For a two-station network, we have $[+1 +1]$ and $[+1 -1]$.
- b. For a four-station network we have $[+1 +1 +1 +1]$, $[+1 -1 +1 -1]$, $[+1 +1 -1 -1]$, and $[+1 -1 -1 +1]$.

Example 12.7

What is the number of sequences if we have 90 stations in our network?

Solution

The number of sequences needs to be 2^m . We need to choose $m = 7$ and $N = 2^7$ or 128. We can then use 90 of the sequences as the chips.

Example 12.8

Prove that a receiving station can get the data sent by a specific sender if it multiplies the entire data on the channel by the sender's chip code and then divides it by the number of stations.

Solution

Let us prove this for the first station, using our previous four-station example. We can say that the data on the channel $D = d_1 \cdot (1 + d_2 \cdot (2 + d_3 \cdot (3 + d_4 \cdot (4)))$. The receiver which wants to get the data sent by station 1 multiplies these data by c_1 :

$$\begin{aligned} D \cdot c_1 &= (d_1 \cdot (1 + d_2 \cdot (2 + d_3 \cdot (3 + d_4 \cdot (4))) \cdot c_1 \\ &= d_1 \cdot c_1 \cdot (1 + d_2 \cdot (2 + d_3 \cdot (3 + d_4 \cdot (4))) \cdot c_1 \\ &= d_1 \times N + d_2 \times 0 + d_3 \times 0 + d_4 \times 0 \\ &= d_1 \times N \end{aligned}$$

When we divide the result by N , we get d_1 .

12.4 RECOMMENDED READING

For more details about subjects discussed in this chapter, we recommend the following books. The items in brackets [...] refer to the reference list at the end of the text.

Books

Multiple access is discussed in Chapter 4 of [Tan03], Chapter 16 of [Sta04], Chapter 6 of [GW04], and Chapter 8 of [For03]. More advanced materials can be found in [KMK04].

12.5 KEY TERMS

I-persistent method	multiple access (MA)
ALOHA	nonpersistent method
binary exponential backup	orthogonal sequence
carrier sense multiple access (CSMA)	polling
carrier sense multiple access with collision avoidance (<i>CSMA/CA</i>)	p-persistent method
carrier sense multiple access with collision detection (<i>CSMA/CD</i>)	primary station
channelization	propagation time
code-division multiple access (CDMA)	pure ALOHA
collision	random access
contention	reservation
controlled access	secondary station
frequency-division multiple access (FDMA)	slotted ALOHA
inner product	time-division multiple access (TDMA)
interframe space (IFS)	token
jamming signal	token passing
	vulnerable time
	Walsh table

12.6 SUMMARY

- O We can consider the data link layer as two sublayers. The upper sublayer is responsible for data link control, and the lower sublayer is responsible for resolving access to the shared media.
- U Many formal protocols have been devised to handle access to a shared link. We categorize them into three groups: random access protocols, controlled access protocols, and channelization protocols.
- O In random access or contention methods, no station is superior to another station and none is assigned the control over another.
- O ALOHA allows multiple access (MA) to the shared medium. There are potential collisions in this arrangement. When a station sends data, another station may attempt to do so at the same time. The data from the two stations collide and become garbled.
- O To minimize the chance of collision and, therefore, increase the performance, the CSMA method was developed. The chance of collision can be reduced if a station

senses the medium before trying to use it. Carrier sense multiple access (CSMA) requires that each station first listen to the medium before sending. Three methods have been devised for carrier sensing: I-persistent, nonpersistent, and p-persistent.

- O Carrier sense multiple access with collision detection (CSMA/CD) augments the CSMA algorithm to handle collision. In this method, a station monitors the medium after it sends a frame to see if the transmission was successful. If so, the station is finished. If, however, there is a collision, the frame is sent again.
- D To avoid collisions on wireless networks, carrier sense multiple access with collision avoidance (CSMA/CA) was invented. Collisions are avoided through the use three strategies: the interframe space, the contention window, and acknowledgments.
- O In controlled access, the stations consult one another to find which station has the right to send. A station cannot send unless it has been authorized by other stations. We discussed three popular controlled-access methods: reservation, polling, and token passing.
- O In the reservation access method, a station needs to make a reservation before sending data. Time is divided into intervals. In each interval, a reservation frame precedes the data frames sent in that interval.
- O In the polling method, all data exchanges must be made through the primary device even when the ultimate destination is a secondary device. The primary device controls the link; the secondary devices follow its instructions.
- D In the token-passing method, the stations in a network are organized in a logical ring. Each station has a predecessor and a successor. A special packet called a token circulates through the ring.
- O Channelization is a multiple-access method in which the available bandwidth of a link is shared in time, frequency, or through code, between different stations. We discussed three channelization protocols: FDMA, TDMA, and CDMA.
- D In frequency-division multiple access (FDMA), the available bandwidth is divided into frequency bands. Each station is allocated a band to send its data. In other words, each band is reserved for a specific station, and it belongs to the station all the time.
- D In time-division multiple access (TDMA), the stations share the bandwidth of the channel in time. Each station is allocated a time slot during which it can send data. Each station transmits its data in its assigned time slot.
- O In code-division multiple access (CDMA), the stations use different codes to achieve multiple access. CDMA is based on coding theory and uses sequences of numbers called chips. The sequences are generated using orthogonal codes such the Walsh tables.

12.7 PRACTICE SET

Review Questions

1. List three categories of multiple access protocols discussed in this chapter.
2. Define random access and list three protocols in this category.

3. Define controlled access and list three protocols in this category.
4. Define channelization and list three protocols in this category.
5. Explain why collision is an issue in a random access protocol but not in controlled access or channelizing protocols.
6. Compare and contrast a random access protocol with a controlled access protocol.
7. Compare and contrast a random access protocol with a channelizing protocol.
8. Compare and contrast a controlled access protocol with a channelizing protocol.
9. Do we need a multiple access protocol when we use the localloop of the telephone company to access the Internet? Why?
10. Do we need a multiple access protocol when we use one CATV channel to access the Internet? Why?

Exercises

11. We have a pure ALOHA network with 100 stations. If $T_{fr} = 1 \mu s$, what is the number of frames/s each station can send to achieve the maximum efficiency.
12. Repeat Exercise 11 for slotted ALOHA.
13. One hundred stations on a pure ALOHA network share a 1-Mbps channel. If frames are 1000 bits long, find the throughput if each station is sending 10 frames per second.
14. Repeat Exercise 13 for slotted ALOHA.
15. In a *CDMA/CD* network with a data rate of 10 Mbps, the minimum frame size is found to be 512 bits for the correct operation of the collision detection process. What should be the minimum frame size if we increase the data rate to 100 Mbps? To 1 Gbps? To 10 Gbps?
16. In a *CDMA/CD* network with a data rate of 10 Mbps, the maximum distance between any station pair is found to be 2500 m for the correct operation of the collision detection process. What should be the maximum distance if we increase the data rate to 100 Mbps? To 1 Gbps? To 10 Gbps?
17. In Figure 12.12, the data rate is 10 Mbps, the distance between station A and C is 2000 m, and the propagation speed is 2×10^8 m/s. Station A starts sending a long frame at time $t_1 = 0$; station C starts sending a long frame at time $t_2 = 3 \mu s$. The size of the frame is long enough to guarantee the detection of collision by both stations. Find:
 - a. The time when station C hears the collision $(t_3)'$
 - b. The time when station A hears the collision $(t_4)'$
 - c. The number of bits station A has sent before detecting the collision.
 - d. The number of bits station C has sent before detecting the collision.
18. Repeat Exercise 17 if the data rate is 100 Mbps.
19. Calculate the Walsh table W_5 from W_4 in Figure 12.29.
20. Recreate the W_2 and W_4 tables in Figure 12.29 using $W_1 = [-1]$. Compare the recreated tables with the ones in Figure 12.29.
21. Prove the third and fourth orthogonal properties of Walsh chips for W_4 in Figure 12.29.

22. Prove the third and fourth orthogonal properties of Walsh chips for W_4 recreated in Exercise 20.
23. Repeat the scenario depicted in Figures 12.27 to 12.28 if both stations 1 and 3 are silent.
24. A network with one primary and four secondary stations uses polling. The size of a data frame is 1000 bytes. The size of the poll, ACK, and NAK frames are 32 bytes each. Each station has 5 frames to send. How many total bytes are exchanged if there is no limitation on the number of frames a station can send in response to a poll?
25. Repeat Exercise 24 if each station can send only one frame in response to a poll.

Research Activities

26. Can you explain why the vulnerable time in ALOHA depends on T_{Fr} , but in CSMA depends on T_p ?
27. In analyzing ALOHA, we use only one parameter, time; in analyzing CSMA, we use two parameters, space and time. Can you explain the reason?