
KTU NOTES

Module 4 Link Layer Communication
Data Link Layer – design issues - Error Detection: Parity Check, Checksum, CRC, Error Correction: Hamming code - Flow Control: Stop-and-Wait, Go-Back-N, and Selective-Repeat - Random Access: ALOHA, CSMA, CSMA/CD, CSMA/CA, Controlled Access: Reservation, Polling, Token Passing, Ethernet- Ethernet Cabling, Encoding, Frame Format, Binary Exponential Back Off Algorithm.

DATA LINK LAYER DESIGN ISSUES

DATA LINK LAYER

The data link layer has a number of specific functions it can carry out. These functions include

1. Providing a well-defined service interface to the network layer.
2. Provides services for the reliable interchange of data across a data link established by the physical layer.
3. Link layer protocols manage the establishment, maintenance, and release of data-link connections.
4. Data-link protocols control the flow of data, dealing with transmission errors, and supervise data error recovery.
5. Regulating the flow of data so that slow receivers are not swamped by fast senders.
6. Recovery from abnormal conditions.

To accomplish these goals, the data link layer takes the packets it gets from the network layer and encapsulates them into frames for transmission. Each frame contains a frame header, a payload field for holding the packet, and a frame trailer, as illustrated in Fig. 2-1. Frame management forms the heart of what the data link layer does.

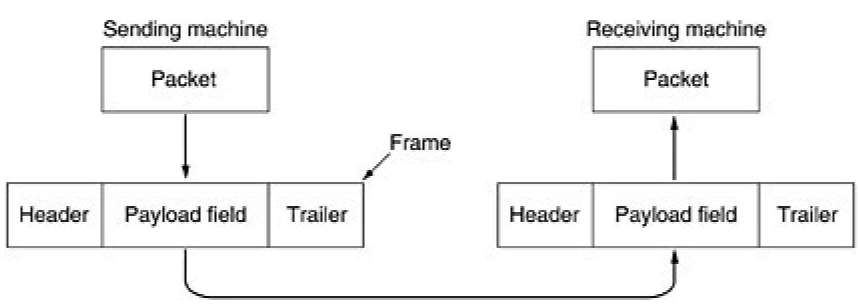


Figure 2-1. Relationship between packets and frames.

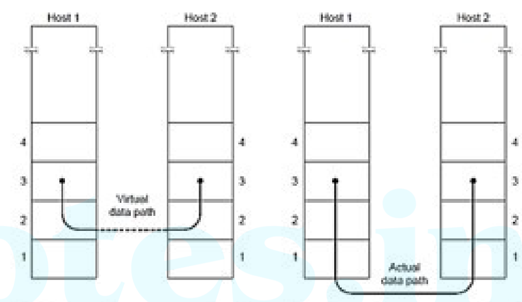
Data Link Layer Design Issues

1. Providing a well-defined service interface to the network layer
2. Determining how the bits of the physical layer are grouped into frames
3. Dealing with transmission errors
4. Regulating the flow of frames so that slow receivers are not swamped by fast senders.

1. Services Provided to the Network Layer

- The function of the data link layer is to provide service to the network layer.
- The principal service is transferring data from the network layer on the source machine to the network layer on the destination machine.
- The network layer hands some bits to the data link layer for transmission to the destination, the job of the data link layer is to transmit the bits to the destination machine, so they can be handed over to the network layer on the destination machine.

The job of the data link layer is to transmit the bits to the destination machine so they can be handed over to the network layer there, as shown in Fig.(a). The actual transmission follows the path of Fig.(b), but it is easier to think in terms of two data link layer processes communicating using a data link protocol.



(a) Virtual communication. (b) Actual communication.

- The data link layer can be designed to offer various services, Three possibilities that are commonly provided are:

1. Unacknowledged connectionless service.

2. Acknowledged connectionless service.

3. Acknowledged connection-oriented service.

Unacknowledged connectionless service

Unacknowledged connectionless service consists of having the source machine send independent frames to the destination machine without having the destination machine acknowledge them. No connection is established beforehand or released afterward. Good channels with low eror rates, for real-time traffic, such as speech.

Acknowledged connectionless service

When this service is offered, there are still no connections used, but each frame sent is individually acknowledged. This way, the sender knows whether or not a frame has arrived safely. Good for unreliable channels, such as wireless.

Acknowledged connection-oriented service

With this service, the source and destination machines establish a connection before any data are transferred. Each frame sent over the connection is numbered, and the data link layer guarantees that each frame sent is received. Furthermore, it guarantees that each frame is received exactly once and that all frames are received in the right order.

When connection-oriented service is used, transfers have three distinct phases.

1. In the first phase the connection is established by having both sides initialize variable and counter need to keep track of which frames have been received and which ones have not.
2. In the second phase, one or more frames are actually transmitted.
3. In the third phase, the connection is released, freeing up the variables, buffers, and other resources used to maintain the connection.

2. Framing

- In order to provide service to the network layer, the data link layer must use the service provided to it by the physical layer.
- What the physical layer does is accept raw bit stream and attempt to deliver it to the destination. This bit stream is not guaranteed to be error free.
- It is up to the data link layer to detect, and if necessary, correct errors.
- The usual approach is for the data link layer to break the bit stream up into discrete frames and compute the checksum for each frame. When the frames arrive at the destination , the checksum is re-computed.

There are four methods of breaking up the bit stream

1. Character count.
2. Starting and ending character stuffing.
3. Starting and ending flags, with bit stuffing.
4. Physical layer coding violations.

Character count, uses a field in the header to specify the number of characters in the frame. When the data link layer at the destination sees the character count, it knows how many characters follow. Problem: count can possible be misrepresented by a transmission error. This method is rarely used anymore.

Starting and ending character stuffing, gets around the problem of resynchronization after an error by having each frame start with the ASCII character sequence DLE STX and end with the sequence DLE ETX. (DLE is Data Link Escape, STX is Start of Text, and ETX is End of Text). **Problem:** a serious problem occurs with this method when binary data, such as object programs or floating-point numbers, are being transmitted it is possible that the DLE, STX, and ETX characters can occur, which will interfere with the framing. One way to solve this problem is to have the sender's data link layer insert and DLE character just before each "accidental" DLE and the data link layer on the other machine removes them before it gives the data to the network layer, this is called Character stuffing.

Starting and ending flags with bit stuffing, allows data frames to contain and arbitrary number of bits and allows character codes with an arbitrary number of bits per character. Each frame begins and ends with a special bit pattern, 01111110, called a flag byte. Whenever the sender's data link layer encounters five consecutive ones in the data, it automatically stuffs a 0 bit into the outgoing bit stream, which is called bit stuffing. The receiving machine destuffs the 0 bit. • The fourth method, Physical coding violations, is only applicable to networks in which the encoding on the physical medium contains some redundancy. For example, some LANs encode 1 bit of data by using 2 physical bits.

3. **Error Control**
   - The next problem to deal with is to make sure all frames are eventually delivered to the network layer at the destination, and in proper order.
   - The usual way to ensure reliable delivery is to provide the sender with some feedback about what is happening at the other end of the line.
   - One complication with this is that the frame may vanish completely, in which case, the receiver will not react at all, since it has no reason to react.
   - This possibility is dealt with by introducing timers into the data link layer. When the sender transmits a frame, it generally also starts a timer. The timer is set to go off after an interval long enough for the frame to reach the destination machine. If the frame or acknowledgment is lost the timer will go off. The obvious solution is to transmit the frame again. This creates the problem of possible sending frames multiple times. To prevent this from happening, it is generally necessary to assign sequence numbers to outgoing frames, so that the receiver can distinguish retransmission from originals.
   - The whole issue of managing the timers and sequence numbers so as to ensure that each frame is ultimately passed to the network layer at the destination exactly one, no more no less, is an important part of the data link layer's duties.

4. **Flow Control**
   - Another important design issue that occurs in the data link layer (and higher layers as well) is what to do with a sender that systematically wants to transmit frames faster than a receiver can accept them.
   - This situation can easily occur when the sender is running on a fast computer and the receiver is running on a slow machine.
   - The usual solution is to introduce flow control to throttle the sender into sending no faster than the receiver can handle the traffic.
   - Various flow control schemes are known, but most of them use the same basic principle, eg HDLC. The protocol contains well-defined rules about when a sender may transmit the next frame.

ERROR DETECTION AND CORRECTION

Error sources are present when data is transmitted over a medium. Even if all possible error-reducing measures are used during the transmission, an error invariably creeps in and begins to disrupt data transmission. Any computer or communication network must deliver accurate messages.

Error detection is applied mostly in the data-link layer but is also performed in other layers. In some cases, the transport layer includes some sort of error-detection scheme. When a packet arrives at the destination, the destination may extract an error-checking code from the transport header and perform error detection. Sometimes, network-layer protocols apply an error-detection code in the network-layer header. In this case, the error detection is performed only on the IP header, not on the data field. At the application layer, some type of error check, such as detecting lost packets, may also be possible. But the most common place to have errors is still the data-link layer. Possible and common forms of errors at this level are described here and are shown in Figure 2.1

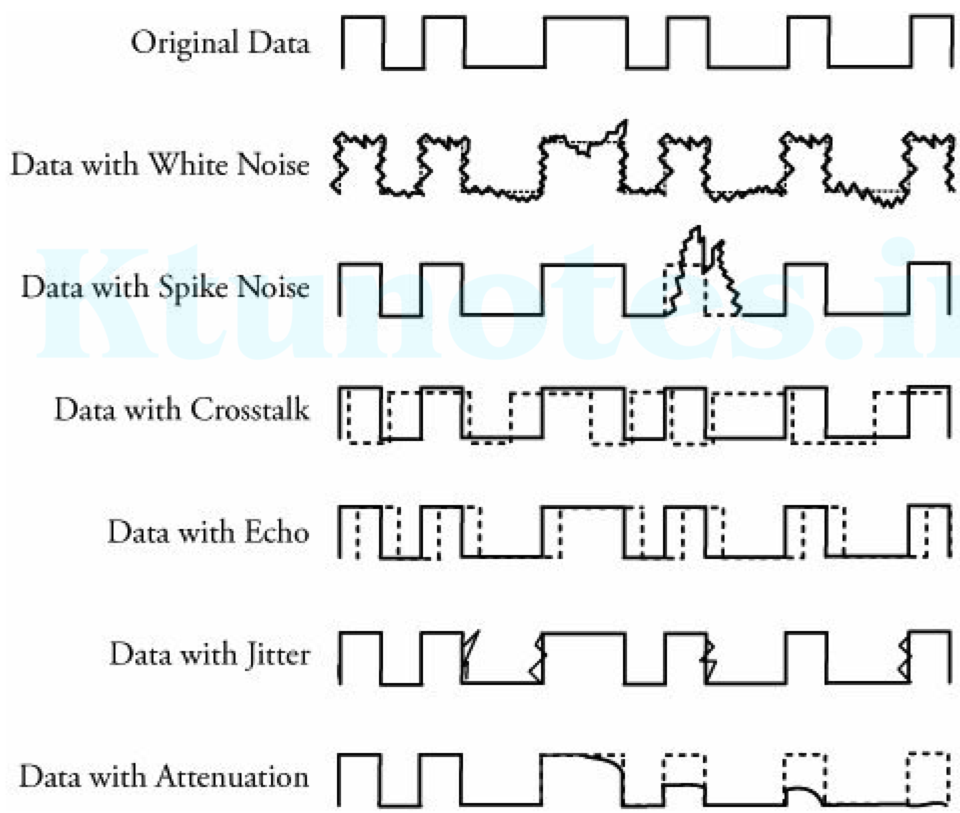


Figure 2.1. Common forms of data errors at the data-link level

Networks must be able to transfer data from one device to another with complete accuracy. Data can be corrupted during transmission. For reliable communication, errors must be detected and corrected.

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; leads to an error in data transmission.

Basic types of errors are

i. Single-Bit Error
ii. Burst Error

i. Single-Bit Error

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

Figure 2.1 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 the figure, 00000010 (ASCII STX) was sent, meaning start of text, but 00001010 (ASCII LF) was received, meaning line feed.

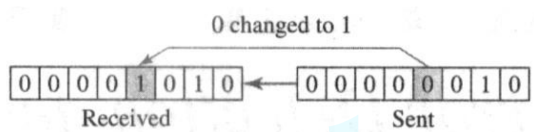


Fig 2.1 Single-bit error

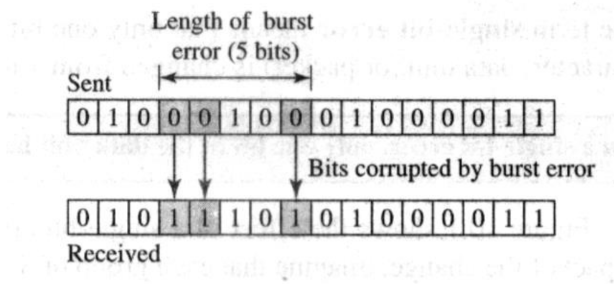
Single-bit errors are the least likely type of error in serial data transmission. To understand Why, imagine a sender sends data at 1 Mbps means that each bit lasts only 1/1,000,000 s, or 1 μs. For a single-bit error to occur, the noise must have a duration of on1y 1 μs, which is very rare; noise normally lasts much longer than this.

However, a single-bit error can happen if we are sending data using parallel transmission. For example, if eight wires are used to send all 8 bits of 1 byte at the same time and one of the wires is noisy, one bit can be corrupted in each byte. Think of parallel transmission inside a computer, between CPU and memory, for example.

ii. 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.

Figure below shows the effect of a burst error on a data unit. In this case, 0100010001000011 was sent, but 0101110101000011 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 been corrupted.



Burst error is most likely to occur in a serial transmission. The duration of noise is normally longer than the duration of one 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 1/100 s can affect 10 bits; if we are sending data at 1 Mbps, the same noise can affect 10,000 bits.

ERROR DETECTION

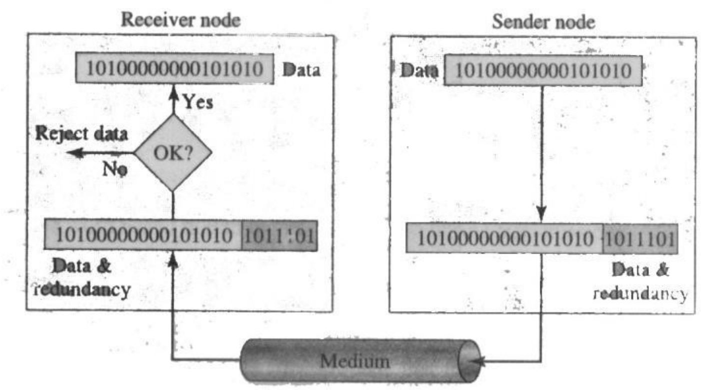
Most networking equipment at the data-link layer inserts some type of error-detection code. When a frame arrives at the next hop in the transmission sequence, the receiving hop extracts the error-detection code and applies it to the frame. When an error is detected, the message is normally discarded. In this case, the sender of the erroneous message is notified, and the message is sent again. However, in real-time applications, it is not possible to resend messages. The most common approaches to error detection are

Redundancy

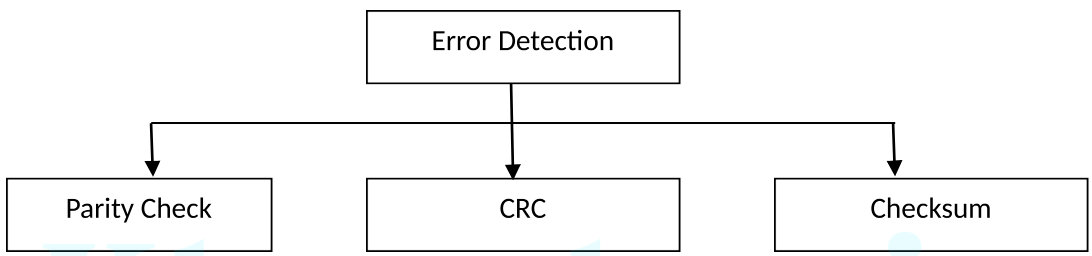
One error detection mechanism would be to send every data unit twice. The receiving device would then able to do a bit for bit comparison between the two versions of the data. Any discrepancy would indicate an error, and an appropriate correction mechanism could be set in place The system would be completely accurate (the odds of errors being introduced on to exactly the same bits in both sets of data are infinitesimally small), but it would also Not only would the transmission time double, but also the time it takes to compare every unit bit by bit must be added.

Error detection uses the concept of redundancy, which means adding extra bits for detecting errors at the destination.. This technique is called redundancy because the extra bits are redundant to the information; they are discarded as soon as the accuracy of the transmission has been determined.

Figure 1.3 shows the process of using redundant bits to check the accuracy of a data unit. Once the data stream has been generated, it passes through a device that analyzes it and adds on an appropriately coded data unit, now enlarged by several bits, travels over the link to the receiver. The receiver puts the entire stream through a checking function, if the received bit stream passes the checking criteria the data portion of the data unit is accepted and the redundant bits are discarded.



Three types of redundancy checks are common in data communications: Parity check, Cyclic Redundancy Check (CRC), and Checksum.

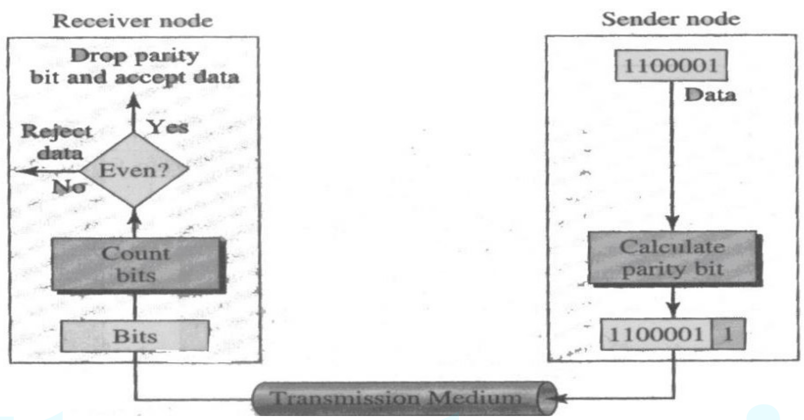


1. PARITY CHECK

The most common and least expensive mechanism for error detection is the parity check. Parity checking can be simple or two dimensional.

Simple Parity Check

In this technique, a redundant bit, called a parity bit, unit so that the total number of Is in the unit (including the parity bit) becomes even (or odd). Suppose we want to transmit the binary data unit 1100001 [ASII a (97)];



Adding the number of 1s gives us 3 an odd number. Before transmitting we pass the data unit through a parity generator. The parity generator counts the 1's and appends the parity bit (a 1 in this case end. The total number of is now 4 an even number. The system now transmits the entire expanded unit across the network link. When it reaches its destination, the receiver puts all 8 bits through an even-parity checking function. If the receiver sees 11000011, it counts four is, an even number, and the data unit passes. But if the data unit has been damaged in transit, means instead of 11000011, the receiver sees 11001011. Then when the parity checker counts the 1s, it gets 5, an odd number. The receiver knows that an error has been introduced into the data somewhere and therefore rejects the whole unit.

Some systems may use odd-parity checking, where the number of Is should be odd. The principle is the same.

Example: Suppose the sender wants to send the word world. In ASCII, the five characters are

coded as 1110111 1101111 1110010 1101100 1100100

w o r l d

Each of the first four characters has an even number of is, so the parity bit is a 0. The last character (d), however, has three 1s (an odd number), so the parity bit is a 1 to make the total number of 1s even. The following shows the actual bits sent (the parity bits are underlined).

1110111<u>0</u> 1101111<u>0</u> 1110010<u>0</u> 1101100<u>0</u> 1100100<u>1</u>

Now suppose the word *world* is received by the receiver without being corrupted in transmission as 1110111<u>0</u> 1101111<u>0</u> 1110010<u>0</u> 1101100<u>0</u> 1100100<u>1</u>, The receiver counts the is in each character and
comes up with even numbers (6, 6, 4, 4, 4). The data are accepted.

If the word *world* in is corrupted during transmission and receiver get the data stream as,

111**1**111<u>0</u> 1101111<u>0</u> 1110**1**10<u>0</u> 1101100<u>0</u> 1100100<u>1</u>

The receiver counts the Is in each character and comes up with even and odd numbers (7, 6, 5, 4, 4). The receiver knows that the data are corrupted, discards them, and asks for retransmission.

Performance

Simple parity check can detect all single bit error It can also detect burst errors as long as the total number of bits change to odd (1, 3, 5, etc.). Let's say we have an even-parity data unit where the total number of 1s, including the parity bit, is 6: 1000111011. If any 3 bits change value, the resulting parity will be odd and the error will be detected: 1**111**111011:9, 0110111011:7, 1**l**00**0**1**0**011:5 a11 odd. The checker would return a result of 1, and the data unit would be rejected. The same holds true for any odd number of errors.

Suppose, however, that 2 bits of the data unit are changed: 1**11**0111011:8, 1**1**00**0**11011:6, 1000**0**1101**0**:4. In each case the number of is in the data unit is still even. The parity checker will add them and return an even number although the data unit contains two errors. This method cannot detect errors where the total number of bits changed is even. If any two bits change in transmission, the changes cancel each other and the data unit will pass a parity check even though the data unit is damaged. The same holds true for any even number of errors.

Two Dimensional Parity Check

A better approach is the two dimensional parity check. In this method, block of bits is organized as a table (rows and columns). First we calculate the parity bit for each data unit. Then we organize them into a table format.

For example, shown in Figure 1.6, we have four data units shown in four rows and eight columns. We then calculate parity bit for each column and create a new row of 8 bit; they are the parity bits for the whole block. Note that the first parity bit in the fifth row is calculated based on all first bits; the second parity bit is calculated based on all second bits; and so on. We then attach the parity bits to the original data and send them to the receiver.

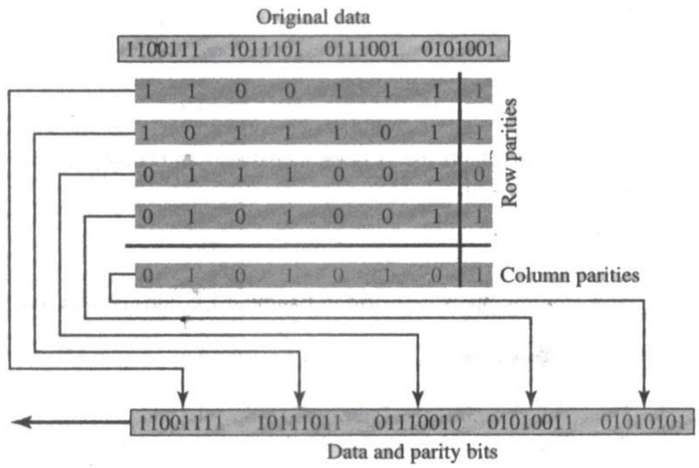


Figure 1.6 Two Dimensional Parity check

Performance

Two-dimensional parity check increases the likelihood of detecting burst errors. A redundancy of *n* bits can easily detect a burst error of n bits. A burst error of more than n bits is also detected by this method with a very high probability. There is, however, one pattern of errors that remains elusive. If 2 bits in one data unit are damaged and two bits in exactly the same positions in another data unit are also damaged, the checker will not detect an error.

2. CYCLIC REDUNDANCY CHECK (CRC)

The most powerful of the redundancy checking techniques is the cyclic redundancy check (CRC). Unlike the parity check which is based on addition, CRC is based on binary division. In CRC, instead of adding bits to achieve a desired parity, a sequence of redundant bits, called the CRC or the CRC remainder, is appended to the end of a data unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number. At its destination, the incoming data unit is divided by the same number. If at this step there is no remainder the data unit is assumed to be intact and is therefore accepted. A remainder indicates that the data unit has been damaged in transmit therefore must be rejected.

The redundancy bits used by CRC are derived by dividing the data unit by a pre determined divisor; the remainder is the CRC.

To be valid, a CRC must have two qualities; it must have exactly one less bit than the divisor, and appending it to the end of the data string must make the resulting bit sequence exactly divisible by the divisor.

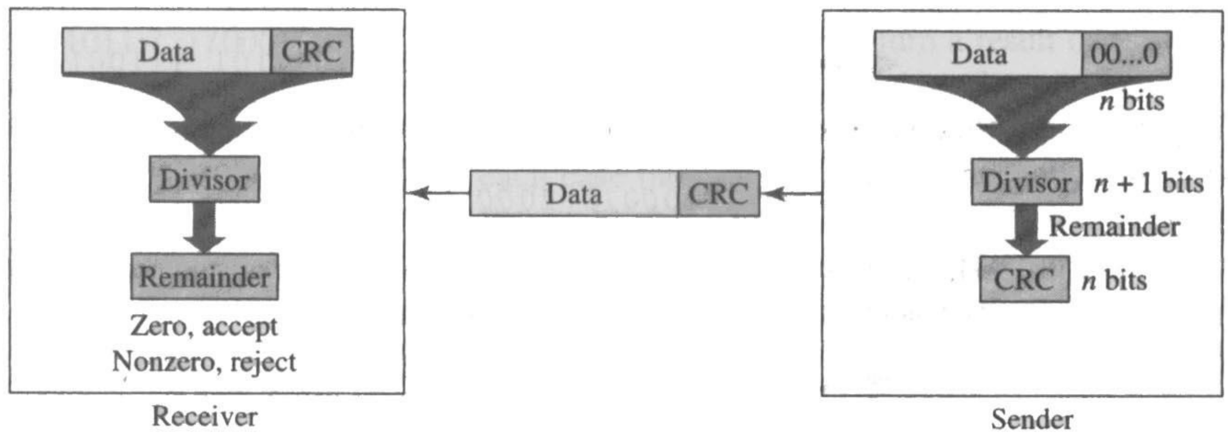


Figure 2.7 CRC generator and

checker Figure 2.7 provides an outline of the three basic steps in

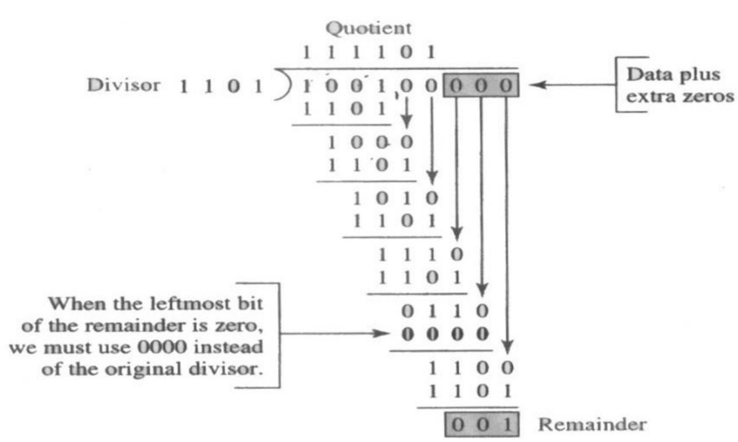
CRC.

1. A string of *n* 0s is appended to the data unit. The number *n* is 1 less than the number of bits in the predetermined divisor, which is *n + 1* bit.
2. The newly elongated data unit is divided by the divisor, using a process called binary division. The remainder resulting from this division is the CRC.
3. The CRC of n bits derived in step 2 replaces the appended 0s at the end of the data unit. Note that the CRC may consist of all 0s.

The data unit arrives at the receiver data first, followed by the CRC. The receiver treats the whole string as a unit and divides it by the same divisor that was used to find the CRC remainder.

If the string arrives without error, the CRC checker yields a remainder of zero and the data unit passes. If the string has been changed in transit, the division yields a nonzero remainder and the data unit does not pass.

The CRC Generator



A CRC generator uses modulo-2 division. Above figure shows this process. In the first step, the 4-bit divisor is subtracted from the first 4 bits of the dividend. Each bit of the divisor is subtracted from the corresponding bit of the dividend without disturbing the next-higher bit. In our example, the divisor, 1101, is subtracted from the first 4 bits of the dividend, 1001, yielding 100 (the leading 0 of the remainder is dropped). The next unused bit from the dividend is then pulled down to make the number of bits in the remainder equal to the number of bits in the divisor. The next step, therefore, is 1000 — 1101, which yields 101, and so on.

In this process, the divisor always begins with a 1; the divisor is subtracted from a portion of the previous that is equal to it in length; the divisor can only be subtracted from a dividend/remainder whose leftmost bit is 1. Anytime the leftmost bit of the dividend/remainder is 0, a string of 0s, of the same length as the divisor, replaces the divisor in that step of the process. For example, if the divisor is 4 bits long, it is replaced by four 0s. (Remember, we are dealing with bit patterns not with quantitative values; 0000 is not the same as 0.) This restriction means that, at any step, the leftmost subtraction will be either 0 - 0 or 1 - 1, both of which equal 0. So, after subtraction, the leftmost bit of the remainder will always be a leading zero, which is dropped, and the next unused bit of the dividend is pulled down to fill out the remainder. Note that only the first bit of the remainder is dropped—if the second bit is also 0, it is retained, and the dividend/remainder for the next step will begin with 0. This process repeats until the entire dividend has been used.

The CRC Checker

A CRC checker functions exactly as the generator does. After receiving the data appended with the CRC, it does the same modulo-2 division. If the remainder is all 0s, the CRC is dropped and the data are accepted; otherwise, the received stream of bits is discarded and data are resent.

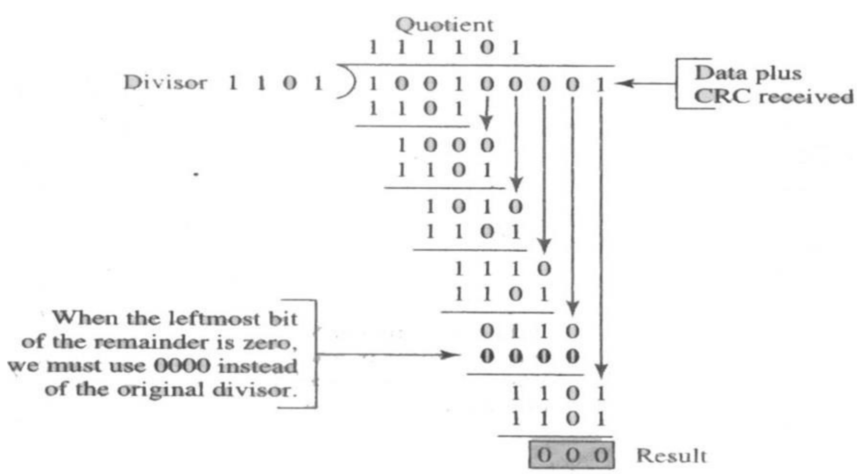


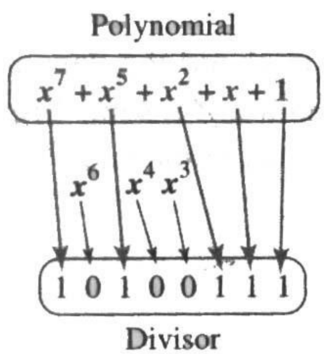
Figure above shows the same process of division in the receiver. We assume that there is no error. The remainder is therefore all 0s, and the data are accepted.

Polynomials

The divisor in the CRC generator is most often represented *not as a string* of 1s and 0s, but as an algebraic polynomial, like $\mathbf{x^7+x^5+x^2+x+1}$

The polynomial format is useful for two reasons: It is short, and it can be used to prove the concept mathematically.

The relationship of a polynomial to its corresponding binary representation is shown below.



A polynomial should be selected to have at least the following properties:

1. It should not be divisible by x.
2. It should be divisible by x + 1.

The first condition guarantees that all burst errors of a length equal to the degree of the polynomial are detected. The second condition guarantees that all burst errors affecting an odd number of bits are detected.

Example: It is obvious that we cannot choose x (binary 10) or x2 + x (binary 110) as the polynomial because both are divisible by x. However, we can choose x + 1 (binary 11) because it is not divisible by x, but is divisible by x + 1. We can also choose x + 1 (binary 101) because it is divisible by x + I (binary division).

Standard Polynomials

Some standard polynomials used by popular protocols for CRC generation are given below.

CRC-8 for ATM: $x^8 + x^2 + x + 1$

CRC-12: $x^{12} + x^{11} + x^3 + x + 1$

CRC-16: $x^{16} + x^{15} + x^2 + 1$

CRC-CCITT: $x^{16} + x^{15} + x^5 + 1$

CRC-32 used in IEEE 802:

$$x^{32} + x^{26} + x^{23} + x^{22} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x + 1$$

Performance

CRC is a very effective error detection method. If the divisor is chosen according to the previously mentioned rules,

1. CRC can detect all burst errors that affect an odd number of bits.
2. CRC can detect all burst errors of length less than or equal to the degree of the polynomial.
3. CRC can detect, with a very high probability, burst errors of length greater than the degree of the polynomial.

Example The CRC- 12 ($x^{12} + x^{11} + x^3 + x + 1$), which has a degree of 12, will detect all burst errors affecting an odd number of bits, will detect all burst errors with a length less than or equal to 12, and will detect, 99.97 percent of the time, burst errors with a length of 12 or more.

3. CHECKSUM

The third error detection method we discuss here is called the checksum. Like the parity checks and CRC, the checksum is based on the concept of redundancy.

Checksum Generator

In the sender, the checksum generator did the following actions.

1. The unit is subdivided into *k* sections, each of *n* bits.
2. All these *k* sections are added using ones complement arithmetic in such a way that the total is also
   n bits long.
3. The sum is complemented and appended to the end of the original data unit as redundancy bits, called the checksum field
4. The extended data unit is transmitted across the network. So, if the sum of the data segment is T, the checksum will be -T.

Checksum Checker

The receiver follows these steps

1. The unit is divided into k sections, each of n bits as the checksum generation.
2. All sections are added using ones complement to get the sum.
3. The sum is complemented.
4. If the result is zero, the data are accepted: otherwise, they are rejected.

Example: Suppose the following block of 16 bits is to be sent using a checksum of 8 bits.

10101001 00111001

The numbers are added using ones complement arithmetic

	10101001
	00111001
Sum	11100010
Checksum	00011101

The pattern sent is: 10101001 00111001 00011101
Data Checksum

Performance

The checksum detects all errors involving an odd number of bits as well as most errors involving an even number of bits. However, if one or more bits of a segment are damaged and the corresponding bit or opposite value in a second segment are also damaged, the sums of those columns will not change and the receiver will not detect a problem. If the last digit of one segment is a 0 and it gets changed to a 1 in transit, then the last 1 in another segment must be changed to a 0 if the error is to go undetected.

In two-dimensional parity check, two 0s could both change to is without altering the parity because carries were discarded. Checksum retains all carries; so although two Os becoming is would not alter the value of their own column, it would change the value of the next-higher column. But anytime a bit inversion is balanced by an opposite bit inversion in the corresponding digit of another data segment, the error is invisible.

FLOW AND ERROR CONTROL

Flow and error control are the main functions of the data link layer. Let us informally define each.

Flow Control

Flow control refers to a set of procedures used to restrict the amount of data that the sender can send before waiting for acknowledgment. Flow control coordinates the amount of data that can be sent before receiving acknowledgment and is one of the most important duties of the data link layer. 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.

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: Anytime an error is detected in an exchange, specified frames are retransmitted. This process is called ***automatic repeat request*** (**ARQ**).

FLOW AND ERROR CONTROL MECHANISMS

Three common flow and error control mechanisms are,

1. Stop- and-Wait ARQ,
2. Go-Back-N ARQ
3. Selective-Repeat ARQ.

These are sometimes referred to as ***sliding window protocols***.

1. STOP-AND-WAIT ARQ

Stop-and-Wait ARQ is the simplest flow and error control mechanism. It has the following features:

- The sending device keeps a copy of the last frame transmitted until it receives an acknowledgment for that frame. Keeping a copy allows the sender to retransmit lost or damaged frames until they are received correctly.
- For identification purposes, both data frames and acknowledgment (ACK) frames are numbered alternately 0 and 1. A data 0 frame is acknowledged by an ACK 1 frame, indicating that the receiver has received data frame 0 and is now expecting data frame 1. This numbering allows for identification of data frames in case of duplicate transmission (important in the case of lost acknowledgment or delayed acknowledgment).
- A damaged or lost frame is treated in the same manner by the receiver. If the receiver detects an error in the received frame, it simply discards the frame and sends no acknowledgment. If the receiver receives a frame that is out of order (0 instead of 1 or 1 instead of 0), it knows that a frame is lost. It discards the out-of-order received frame.
- The sender has a control variable, which we call S, that holds the number of the recently sent frame (0 or 1). The receiver has a control variable, which we call R, that holds the number of the next frame expected (0 or 1).
- The sender starts a timer when it sends a frame. If an acknowledgment is not received within an allotted time period, the sender assumes that the frame was lost or damaged and resends it.
- The receiver sends only positive acknowledgment for frames received safe and Sound; it is silent about the frames damaged or lost. The acknowledgment number always defines the number of the next expected frame. If frame 0 is received, ACK I is sent; if frame 1 is

received, ACK 0 is sent.

Operation

In the transmission of a frame, we can have four situations: *normal operation*, the *frame is lost*, *the acknowledgement is lost*, or *the acknowledgment is delayed*.

Normal Operation

In a normal transmission, the sender sends frame 0 and waits to receive ACK 1. When ACK 1 is received, it sends frame 1 and then waits to receive ACK 0, and so on. The ACK must be received before the timer set for each frame expires. Figure below shows successful frame transmissions

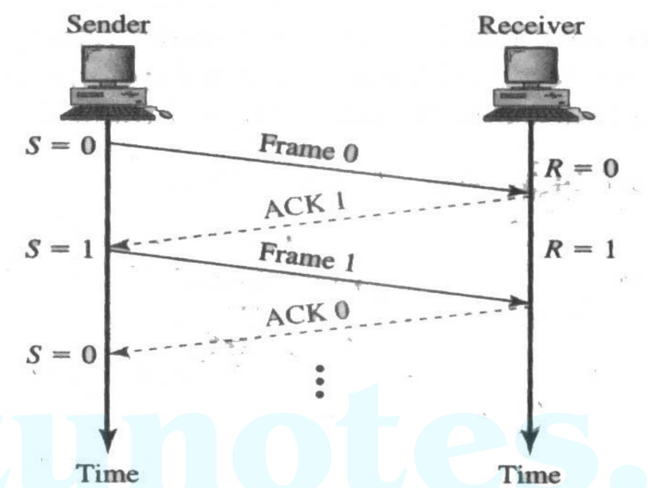


Figure: Stop and Wait ARQ, Normal operation

Lost or Damaged Frame

A lost or damaged frame is handled in the same way by the receiver; when the receiver receives a damaged frame, it discards it, which essentially means the frame is lost. The receiver remains silent about a lost frame and keeps its value of R. For example, in Figure below, the sender transmits frame 1, but it is lost. The receiver does nothing, retaining the value of R (1). After the timer at the sender site expires, another copy of frame 1 is sent.

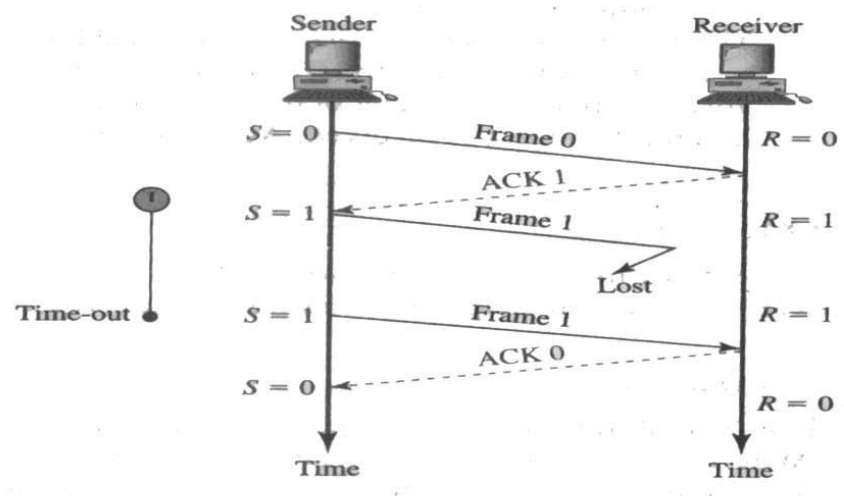


Figure: Stop and Wait ARQ, Lost or Damaged Frame

A lost or damaged acknowledgment

A lost or damaged acknowledgment is handled in the same way by the sender; if the sender receives a damaged acknowledgment, it discards it. Figure below shows a lost ACK 0. The waiting sender does not know if frame I has been received. When the timer for frame 1 expires, the sender retransmits frame 1. Note that the receiver has already received frame 1 and is expecting to receive frame 0 (R = 0). Therefore, it silently discards the second copy of frame 1.

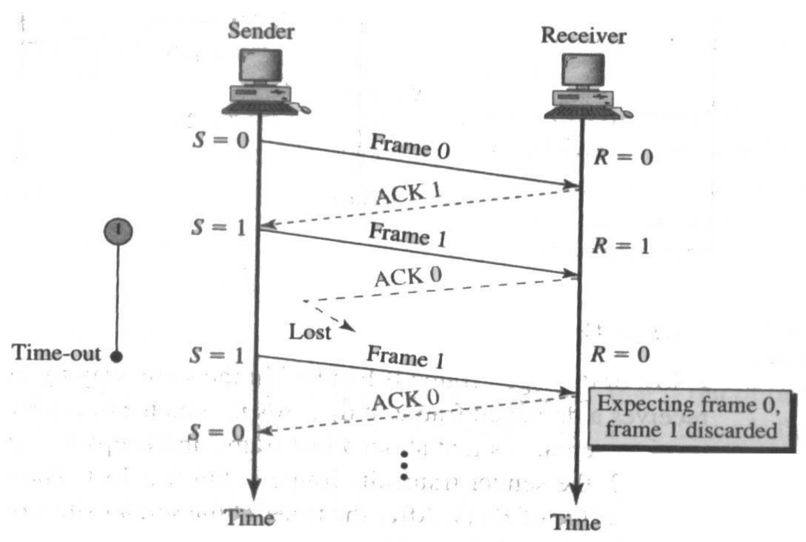


Figure: Stop and Wa1t ARQ, lost ACK frame

Delayed acknowledgment

Another problem that may occur is delayed acknowledgment. An acknowledgment can be delayed at the receiver or by some problem with the link. Figure 4 shows the delay of ACK 1; it is received after the timer for frame 0 has already expired. The sender has already retransmitted a copy of frame 0. However, the value of R at the receiver site is still 1, which means that the receiver expects to see frame 1. The receiver, therefore, discards the duplicate frame 0.

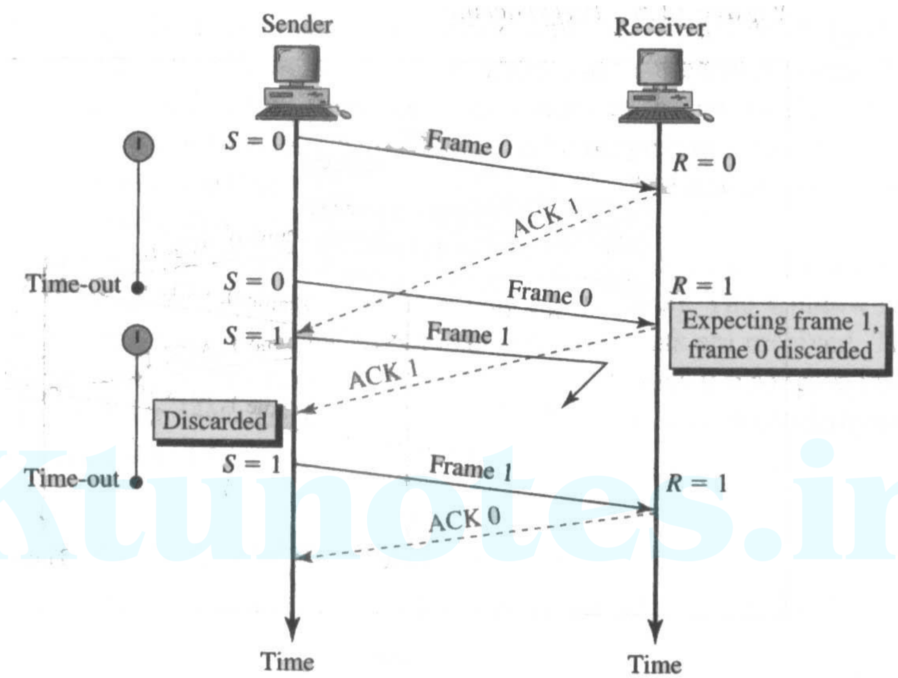


Figure 4: Stop and Wa1t ARQ, delayed ACK frame

The sender has now received two ACKs, one that was delayed and one that was sent after the duplicate frame 0 arrived. The second ACK 1 is discarded.

After the delayed ACK 1 reaches the sender, frame I is sent. However, frame 1 is lost and never reaches the receiver. The sender then receives an ACK 1 for the duplicate frame sent. If the ACKs were not numbered, the sender would interpret the second ACK as the acknowledgment for frame 1. Numbering the ACKs provides a method to keep track of the received data frames.

Bidirectional Transmission

If the two parties have two separate channels for full- duplex transmission or share the same channel for half-duplex transmission. In this case, each party needs both S and R variables to track frames sent and expected.

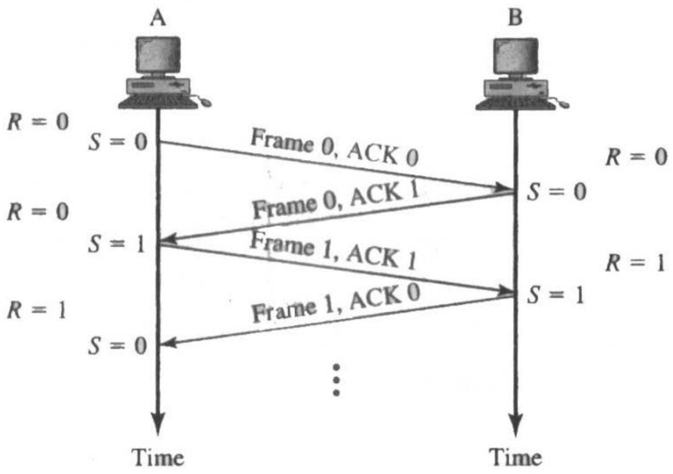


Figure 5: Bidirectional Transmission, Piggybacking

Piggybacking

Piggybacking is a method to combine a data frame with an acknowledgment. For example, in Figure 5, Stations A and B both have data to send. Instead of sending separate data and ACK frames, station A sends a data frame that includes an ACK. Station B behaves in a similar manner.

Piggybacking can save bandwidth because the overhead from a data frame and an ACK frame (addressees, CRC, etc.,) can be combined into just one frame.

2. GO-BACK-N ARQ

In Stop-and-Wait ARQ, at any point in time for a sender, there is only one frame, the outstanding frame, that is sent and waiting to be acknowledged. This is not a good use of the transmission medium. To improve the efficiency, multiple frames should be in transition while waiting for acknowledgment. In other words, we need to let more than one frame be outstanding. Two protocols use this concept: Go-Back-N ARQ and Selective Repeat ARQ.

In Go-Back-N ARQ, we can send up to W frames before worrying about acknowledgments; we keep a copy of these frames until the acknowledgments arrive. This procedure requires additional features to be added to Stop-and-Wait ARQ.

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 3, the only sequence numbers are 0 through 7 inclusive. However, we can repeat the sequence. So the sequence numbers are

0,l,2,3,4,5,6,7, 0,l,2,3,4,5,6,7….1

Sender Sliding Window

At the sender site, to hold the outstanding frames until they are acknowledged, we use the concept of a window. We imagine that all frames are stored in a buffer. The outstanding frames are enclosed in a window. The frames to the left of the window are those that have already been acknowledged and can be purged; those to the right of the Window cannot be sent until the window slides over them. The size of the window is at most $2^m — 1$.

The size of the window in this protocol is fixed, although we can have a variable- size window in other protocols such as TCP. The window slides to include new unsent frames when the correct acknowledgments are received. The Window is a sliding window. For example, in Figure 6a, frames 0 through 6 have been sent. In part b, the window slides two frames over because an acknowledgment was received for frames 0 and 1.

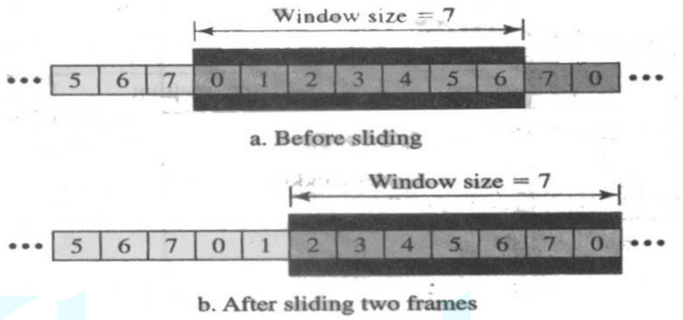


Figure 6: Sender Sliding Window

Receiver Sliding Window

The size of the window at the receiver side in this protocol is always 1. The receiver is always looking for a specific frame to arrive in a specific order. Any frame arriving out of order is discarded and needs to be resent. The receiver window also slides as shown in Figure 7. In part *a* the receiver is waiting for frame 0. When that arrives, the window slides over.

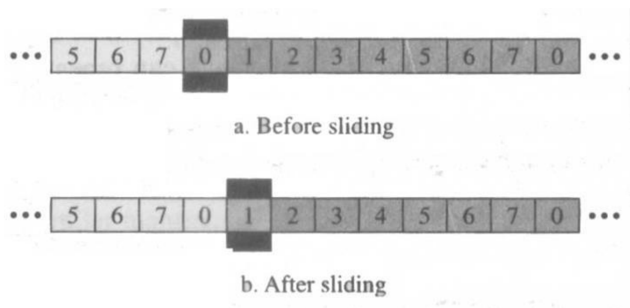


Figure 6: Sender Sliding Window

Control variables

The sender has three variables, S, S_F, and S_L. The S variable holds the sequence number of the recently sent frame; S_F holds the sequence number of the first frame in the window; and S_L holds the sequence number of the last frame in the window. The size of the window is W, where $W = S_L - S_F + 1$.

The receiver only has one variable, R that holds the sequence number of the frame it expects to receive. If the sequence number of the received frame is the same as the value of R, the frame is accepted; if not, it is rejected. Figure 8 shows the sender and receiver window with their control variables.

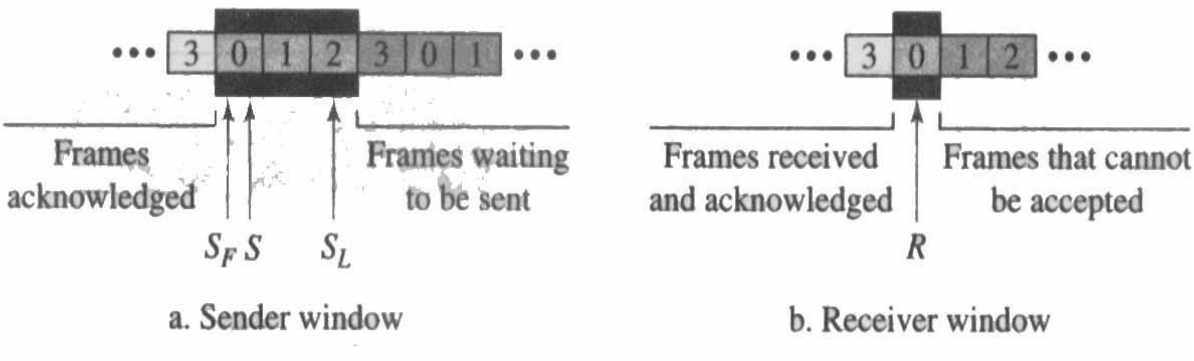


Figure 8, Control variables

Timers

The sender sets a timer for each frame sent. The receiver has no timers.

Acknowledgment

The receiver sends positive acknowledgments 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 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 Frame

When a frame is damaged, the sender goes back and sends a set of frames starting from the damaged one up to the last one sent. 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, so the sender goes back and sends frames 3, 4, 5, 6 again. That is why the protocol is called Go-Back-N ARQ.

OPERATION

Normal Operation

Figure 9 shows a normal operation of Go-Back-N ARQ. The sender keeps track of the outstanding frames and updates the variables and windows as the acknowledgments arrive.

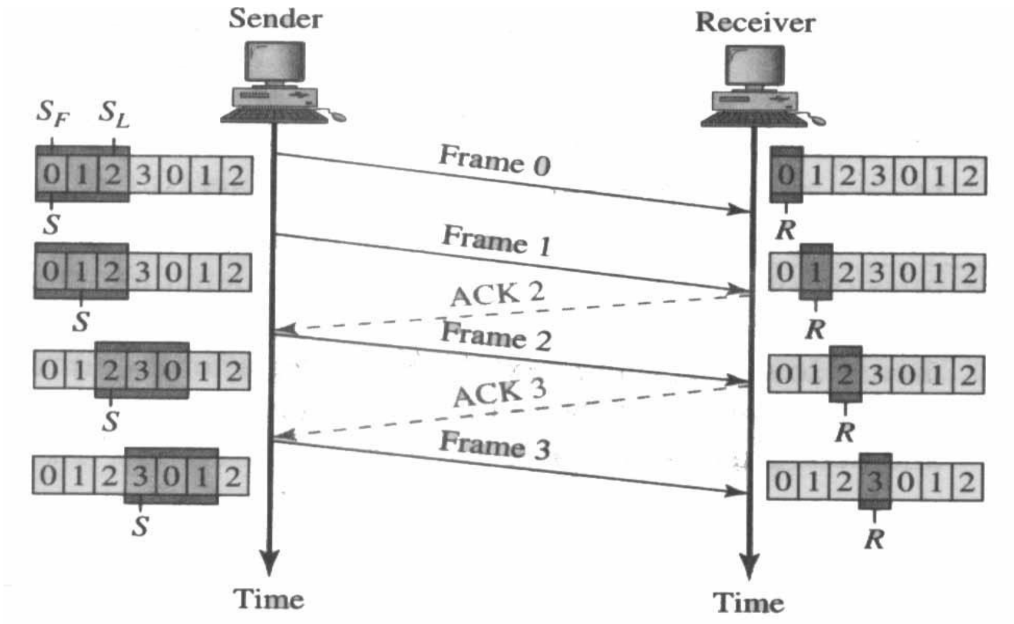


Figure 9 Go-Back-N ARQ, normal operation

Damaged or Lost Frame

Figure 10 shows, the situation when a frame is lost. It shows that frame 2 is lost. Note that when the receiver receives frame 3, it is discarded because the receiver is expecting frame 2, not frame 3 (according to its window). After the timer for frame 2 expires at the sender site, the sender sends frames 2 and 3 (it goes back to 2).

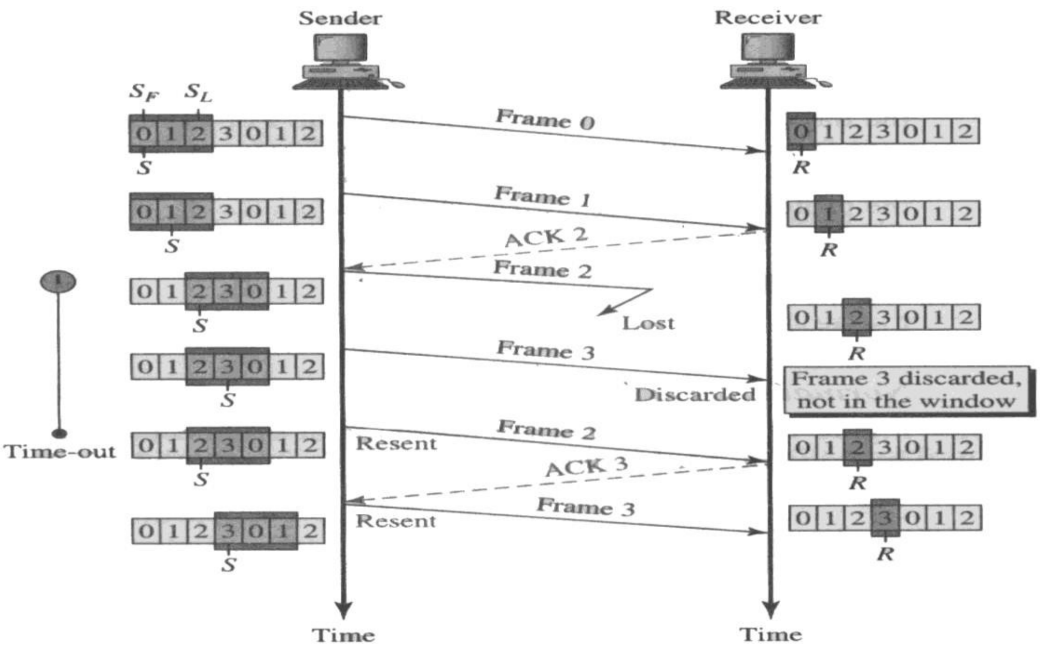


Figure 10: Damaged or Lost Frame

Damaged or lost Acknowledgment

If an acknowledgment is damaged or lost, we can have two situations. If the next acknowledgment arrives before the expiration of any timer, there is no need for retransmission of frames because acknowledgments are cumulative in this protocol. ACK 4 means ACK 1 to ACK 4. So if ACK 1, ACK 2, and ACK 3 are lost, ACK 4 covers them. However, if the next ACK arrives after the time-out, the frame and all the frames after that are resent. The receiver never resends an ACK.

Delayed Acknowledgment

A delayed acknowledgment triggers the resending of frames.

Sender Window Size

We can now show why the size of the sender 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 compares a window size of 3 and 4.

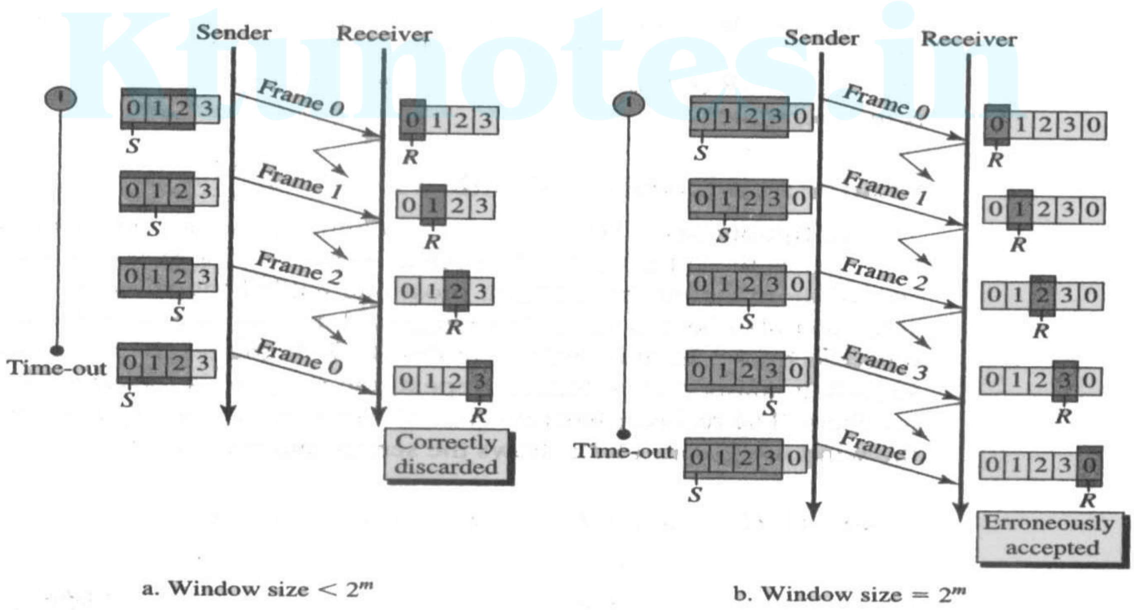


Figure 11 Go-Back-N ARQ: sender Window size

If the size of the window is 3 (less than 22) and all three acknowledgments are lost, the frame 0 timer expires and all three frames are resent. However, the window of 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 22) and all acknowledgments are lost, the sender will send the 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.

Bidirectional Transmission and Piggybacking

As in the case of Stop-and-Wait ARQ, Go-Back N ARQ can also be bidirectional We can also use piggybacking to improve the efficiency of the transmission However, each direction needs both a sender window and a receiver window.

3. SELECTIVE REPEAT ARQ

Go-Back-N ARQ simplifies the process at the receiver site. The receiver keeps3 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 Repeat ARQ. It is more efficient for noisy links, but the processing at the receiver is more complex.

Sender and Receiver Windows

The configuration of the sender and its control variables for Selective Repeat ARQ is the same as those for Go-Back-N ARQ. But the size of the window should be at most one-half of the value 2^m. The receiver window size must also be this size. This window, however, specifies the range of the accepted received frame. In other words, in Go-Back-N, the receiver is looking for one specific sequence number; in Selective Repeat, the receiver is looking for a range of sequence numbers. The receiver has two control variables R_F and R_L to define the boundaries of the window. Figure 12 shows the sender and receiver windows.

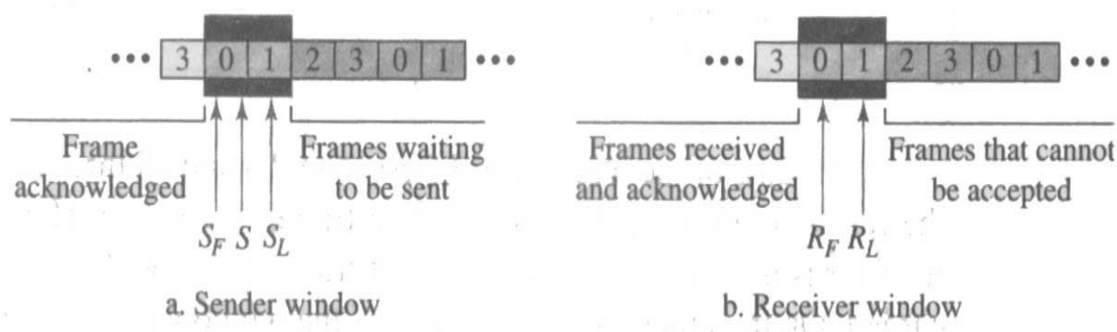


Figure 12 Selective Repeat ARQ, sender and receiver windows

Selective Repeat ARQ also defines a negative acknowledgment (NAK) that reports the sequence number of a damaged frame before the timer expires.

Operation

Figure 13 show the operation of the mechanism of Selective Repeat ARQ with an example of a lost frame.

Frames 0 and I are accepted when received because they are in the range specified by the receiver window. When frame 3 is received, it is also accepted for the same reason. However, the receiver sends a NAK 2 to show that frame 2 has not been received. When the sender receives the NAK 2, it resends only frame 2, which is then accepted because it is in the range of the window.

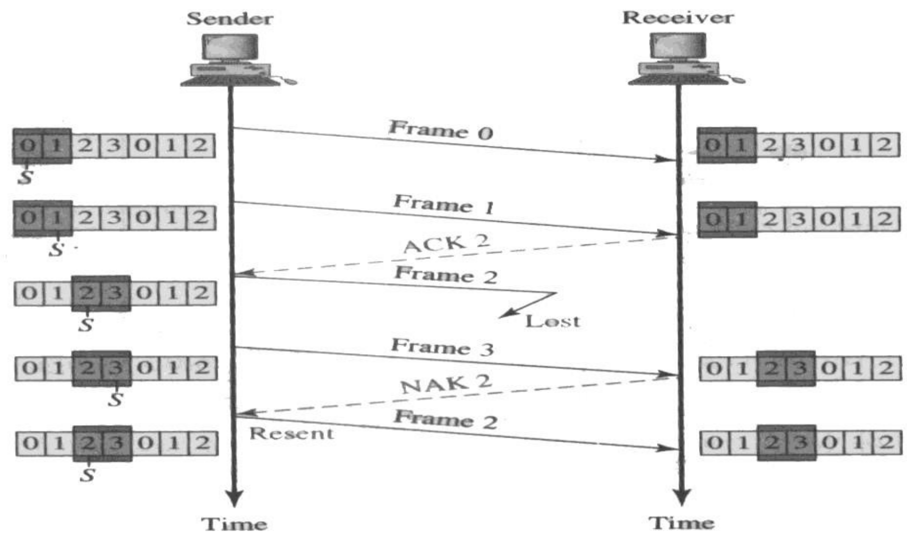


Figure 13 Selective Repeat ARQ, lost frame

Sender Window Size

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 should be $2^m/2$ or 2. Figure 14 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 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.

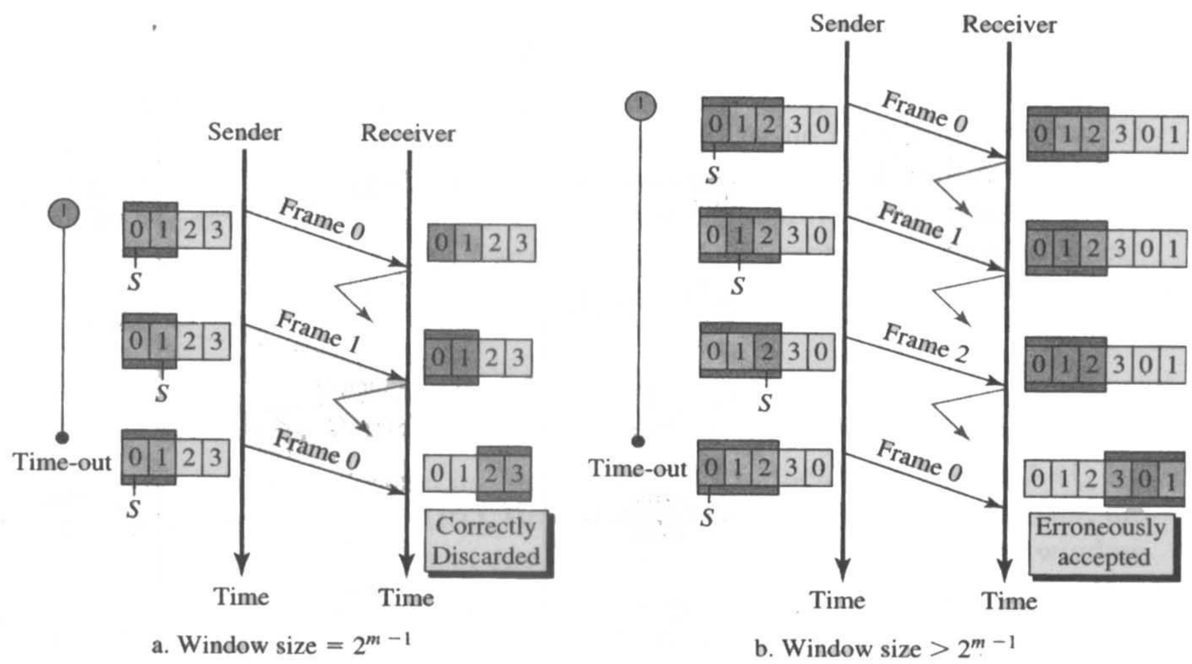


Figure 14 Selective Repeat ARQ, sender window size

Bidirectional Transmission and Piggybacking

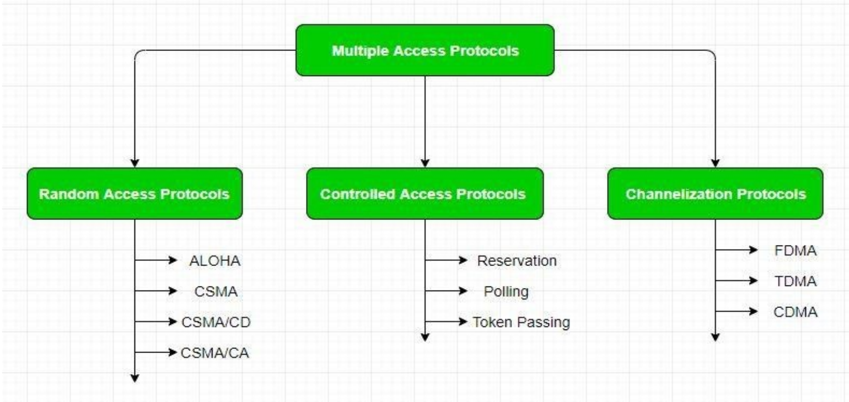
As in the case of Stop-and-Wait ARQ and the Go-Back-N ARQ, Selective Repeat ARQ can also be bidirectional. We can use piggybacking to improve the efficiency of the transmission. However each direction needs both a sender window and a receiver window.

Multiple access protocols

Multiple Access Control –

If there is a dedicated link between the sender and the receiver then data link control layer is sufficient, however if there is no dedicated link present then multiple stations can access the channel simultaneously. Hence multiple access protocols are required to decrease collision and avoid crosstalk. For example, in a classroom full of students, when a teacher asks a question and all the students (or stations) start answering simultaneously (send data at same time) then a lot of chaos is created (data overlap or data lost) then it is the job of the teacher (multiple access protocols) to manage the students and make them answer one at a time.

Thus, protocols are required for sharing data on non-dedicated channels. Multiple access protocols can be subdivided further as –



1. Random Access Protocol: In this, all stations have same superiority that is no station has more priority than another station. Any station can send data depending on medium's state (idle or busy). It has two features:

1. There is no fixed time for sending data
2. There is no fixed sequence of stations sending data

The Random-access protocols are further subdivided as:

ALOHA – It was designed for wireless LAN but is also applicable for shared medium. In this, multiple stations can transmit data at the same time and can hence lead to collision and data being garbled.

Pure Aloha:

When a station sends data, it waits for an acknowledgement. If the acknowledgement doesn't come within the allotted time then the station waits for a random amount of time called back-off time (Tb) and re-sends the data. Since different stations wait for different amount of time, the probability of further collision decreases.

Vulnerable Time = 2* Frame transmission

time Throughput = G exp{-2*G}

Maximum throughput = 0.184 for G=0.5

Slotted Aloha:

It is similar to pure aloha, except that we divide time into slots and sending of data is allowed only at the beginning of these slots. If a station misses out the allowed time, it must wait for the next slot. This reduces the probability of collision.

Vulnerable Time = Frame transmission

time Throughput = G exp{-*G}

Maximum throughput = 0.368 for G=1

<u>Carrier Sense Multiple Access protocols</u>

CSMA Carrier Sense Multiple Access ensures fewer collisions as the station is required to first sense the medium (for idle or busy) before transmitting data. If it is idle then it sends data, otherwise it waits till the channel becomes idle. However, there is still chance of collision in CSMA due to propagation delay. For example, if station A wants to send data, it will first sense the medium. If it finds the channel idle, it will start sending data. However, by the time the first bit of data is transmitted (delayed due to propagation delay) from station A, if station B requests to send data and senses the medium it will also find it idle and will also send data. This will result in collision of data from station A and B.

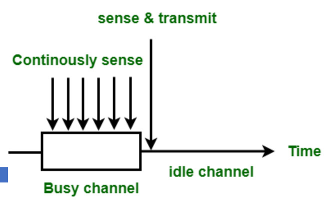
CSMA access modes-

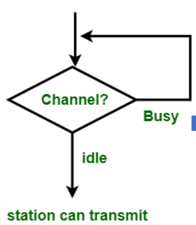
- 1-persistent: The node senses the channel, if idle it sends the data, otherwise it continuously keeps on checking the medium for being idle and transmits unconditionally (with 1 probability) as soon as the channel gets idle.
- Non-Persistent: The node senses the channel, if idle it sends the data, otherwise it checks the medium after a random amount of time (not continuously) and transmits when found idle.
- P-persistent: The node senses the medium, if idle it sends the data with p probability. If the data is not transmitted ((1-p) probability) then it waits for some time and checks the medium again, now if it is found idle then it sends with p probability. This repeat continues until the frame is sent. It is used in Wi-Fi and packet radio systems.
- O-persistent: Superiority of nodes is decided beforehand and transmission occurs in that order. If the medium is idle, node waits for its time slot to send data.

Persistent and non persistent CSMA

1-persistent CSMA:

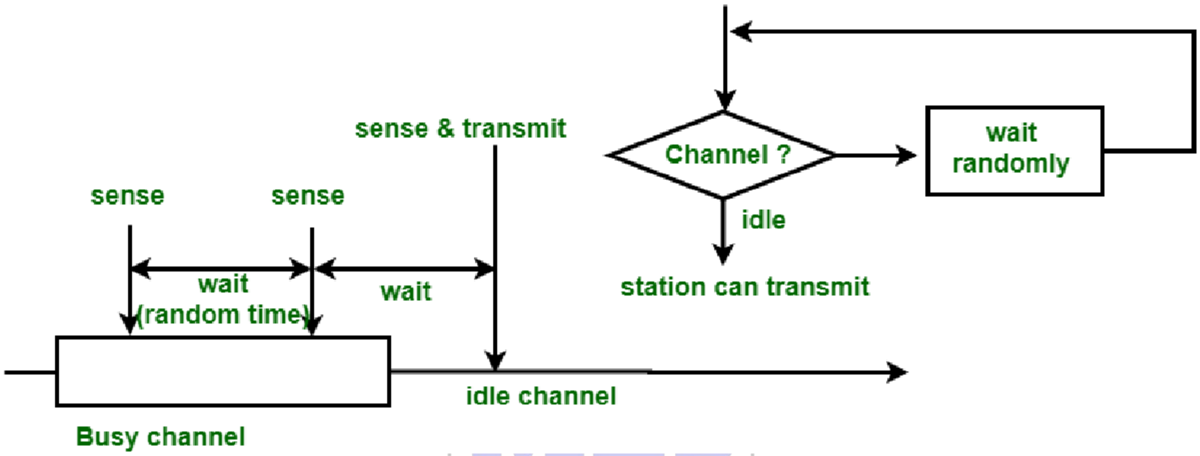
In 1-persistent CSMA, station continuously senses channel to check its state i.e., idle or busy so that it can transfer data. In case when channel is busy, station will wait for channel to become idle. When station finds an idle channel, it transmits frame to channel without any delay with probability 1. Due to probability 1, it is called 1-persistent CSMA. The problem with this method is that there is a huge chance of collision, as two or more stations can find channel in idle state and transmit frames at the same time. At the time when a collision occurs the station has to wait for random time for channel to be idle and to start all again.





Non-persistent CSMA :

In Non-persistent CSMA, station that has frames to send only senses for channel. In the case of an idle channel, it will send frames immediately to that channel. In case when channel is found busy, it will wait for a fixed amount of time and again sense for state of station to be idle or busy. In this method, station does not immediately sense for channel for only purpose of capturing it when it detects end of previous transmission. This method reduces the chances of a collision but reduces efficiency of network.



CSMA with Collition Detaection

CSMA/CD – Carrier sense multiple access with collision detection. Stations can terminate transmission of data if collision is detected. For more details refer – Efficiency of CSMA/CD

CSMA/CA – Carrier sense multiple access with collision avoidance. The process of collisions detection involves sender receiving acknowledgement signals. If there is just one signal (its own) then the data is successfully sent but if there are two signals (its own and the one with which it has collided) then it means a collision has occurred. To distinguish between these two cases, collision must have a lot of impact on received signal. However, it is not so in wired networks, so CSMA/CA is used in this case.

CSMA/CA avoids collision by:

1. Interframe space – Station waits for medium to become idle and if found idle it does not immediately send data (to avoid collision due to propagation delay) rather it waits for a period of time called Interframe space or IFS. After this time it again checks the medium for being idle. The IFS duration depends on the priority of station.
2. Contention Window – It is the amount of time divided into slots. If the sender is ready to send data, it chooses a random number of slots as wait time which doubles every

time medium is not found idle. If the medium is found busy it does not restart the entire process, rather it restarts the timer when the channel is found idle again.

3. Acknowledgement – The sender re-transmits the data if acknowledgement is not received before time-out.

2. Controlled Access:

In this, the data is sent by that station which is approved by all other stations. For further details refer – Controlled Access Protocols

3. Channelization:

In this, the available bandwidth of the link is shared in time, frequency and code to multiple stations to access channel simultaneously.

- Frequency Division Multiple Access (FDMA) – The available bandwidth is divided into equal bands so that each station can be allocated its own band. Guard bands are also added so that no to bands overlap to avoid crosstalk and noise.
- Time Division Multiple Access (TDMA) – In this, the bandwidth is shared between multiple stations. To avoid collision time is divided into slots and stations are allotted these slots to transmit data. However, there is an overhead of synchronization as each station needs to know its time slot. This is resolved by adding synchronization bits to each slot. Another issue with TDMA is propagation delay which is resolved by addition of guard bands.
- Code Division Multiple Access (CDMA) – One channel carries all transmissions simultaneously. There is neither division of bandwidth nor division of time. For example, if there are many people in a room all speaking at the same time, then also perfect reception of data is possible if only two person speak the same language.

  Similarly, data from different stations can be transmitted simultaneously in different code languages.

<u>Controlled Access: Reservation, Polling, Token Passing</u>

In controlled access, the stations seek information from one another to find which station has the right to send. It allows only one node to send at a time, to avoid collision of messages on shared medium.

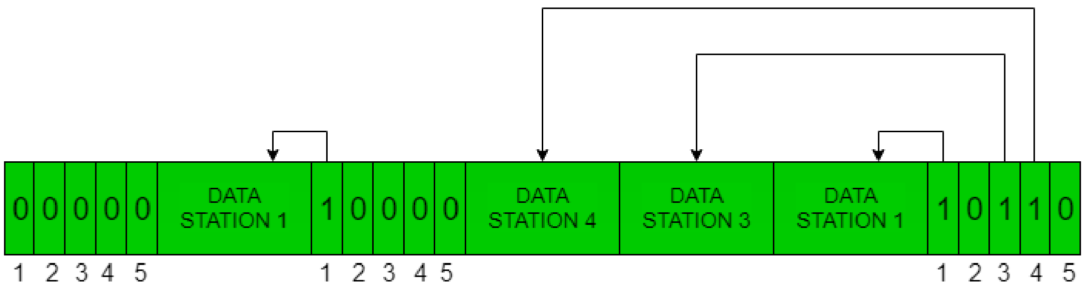
The three controlled-access methods are:

1. Reservation
2. Polling
3. Token Passing

Reservation

- In the reservation method, a station needs to make a reservation before sending data.
- The time line has two kinds of periods:
    1. Reservation interval of fixed time length
    2. Data transmission period of variable frames.
- If there are M stations, the reservation interval is divided into M slots, and each station has one slot.
- Suppose if station 1 has a frame to send, it transmits 1 bit during the slot 1. No other station is allowed to transmit during this slot.
- In general, i th station may announce that it has a frame to send by inserting a 1 bit into i th slot. After all N slots have been checked, each station knows which stations wish to transmit.
- The stations which have reserved their slots transfer their frames in that order.
- After data transmission period, next reservation interval begins.
- Since everyone agrees on who goes next, there will never be any collisions.

The following figure shows a situation with five stations and a five-slot 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.

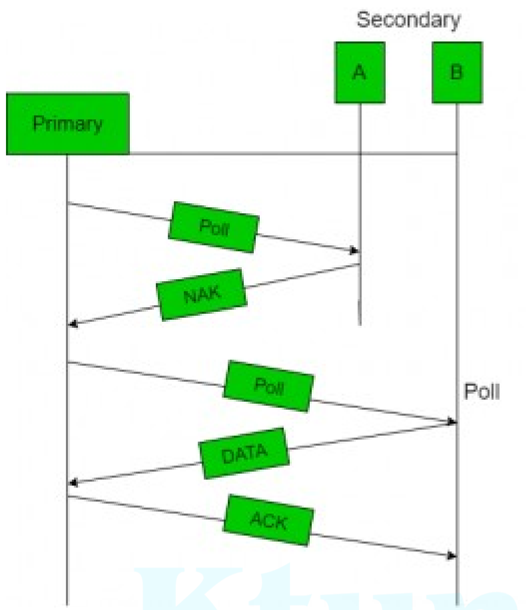


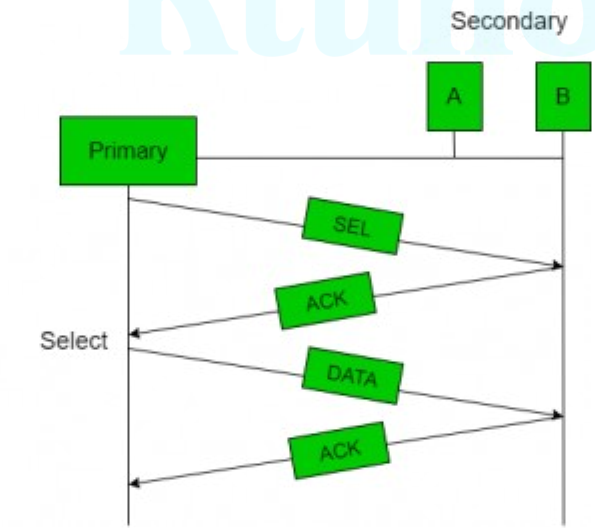
Reservation Frame

Polling

- Polling process is similar to the roll-call performed in class. Just like the teacher, a controller sends a message to each node in turn.
- In this, one acts as a primary station(controller) and the others are secondary stations. All data exchanges must be made through the controller.

- The message sent by the controller contains the address of the node being selected for granting access.
- Although all nodes receive the message but the addressed one responds to it and sends data, if any. If there is no data, usually a “poll reject”(NAK) message is sent back.
- Problems include high overhead of the polling messages and high dependence on the reliability of the controller.





Efficiency

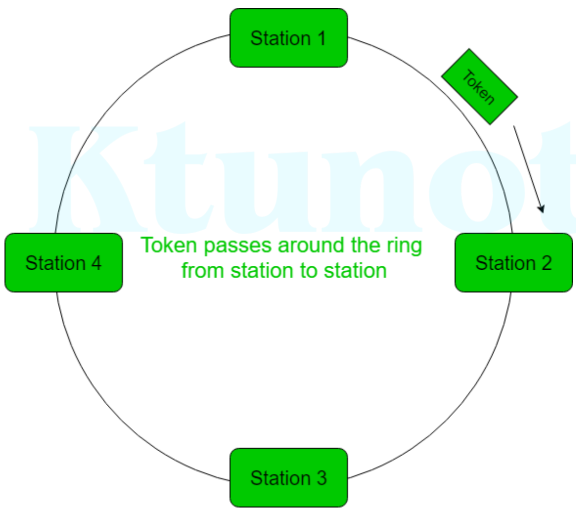
Let T_{poll} be the time for polling and T_t be the time required for transmission of data. Then,

Efficiency = $T_t/(T_t + T_{poll})$

Token Passing

- In token passing scheme, the stations are connected logically to each other in form of ring and access of stations is governed by tokens.

- A token is a special bit pattern or a small message, which circulate from one station to the next in some predefined order.
- In Token ring, token is passed from one station to another adjacent station in the ring whereas in case of Token bus, each station uses the bus to send the token to the next station in some predefined order.
- In both cases, token represents permission to send. If a station has a frame queued for transmission when it receives the token, it can send that frame before it passes the token to the next station. If it has no queued frame, it passes the token simply.
- After sending a frame, each station must wait for all N stations (including itself) to send the token to their neighbors and the other N – 1 stations to send a frame, if they have one.
- There exist problems like duplication of token or token is lost or insertion of new station, removal of a station, which need be tackled for correct and reliable operation of this scheme.



Performance

Performance of token ring can be concluded by 2 parameters:-

1. Delay, which is a measure of time between when a packet is ready and when it is delivered. So, the average time (delay) required to send a token to the next station = a/N.
2. Throughput, which is a measure of the successful traffic.

Throughput, $S = 1/(1 + a/N)$ for $a<1$

and

$S = 1/\{a(1 + 1/N)\}$ for $a>1$.

where N = number of stations

$$a = T_p/T_t$$

(T_p = propagation delay and T_t = transmission delay)

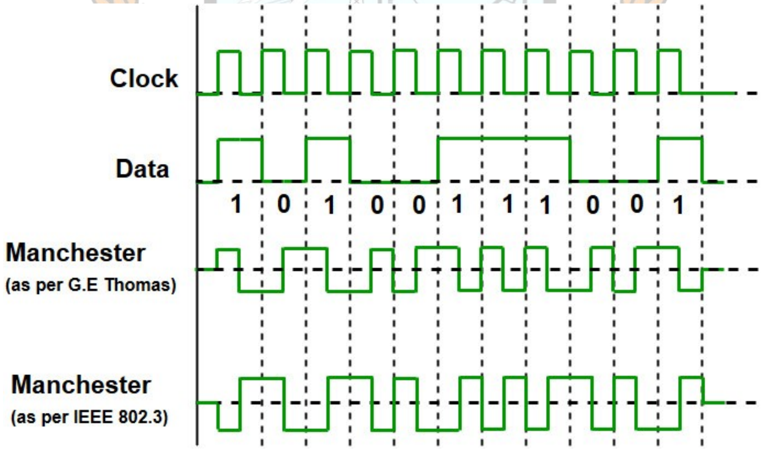
LAN Technologies

Local Area Network (LAN) is a data communication network connecting various terminals or computers within a building or limited geographical area. The connection among the devices could be wired or wireless. Ethernet, Token Ring and Wireless LAN using IEEE 802.11 are examples of standard LAN technologies.

Ethernet: -

Ethernet is most widely used LAN Technology, which is defined under IEEE standards 802.3. The reason behind its wide usability is Ethernet is easy to understand, implement, maintain and allows low-cost network implementation. Also, Ethernet offers flexibility in terms of topologies which are allowed. Ethernet operates in two layers of the OSI model, Physical Layer, and Data Link Layer. For Ethernet, the protocol data unit is Frame since we mainly deal with DLL. In order to handle collision, the Access control mechanism used in Ethernet is CSMA/CD.

Manchester Encoding Technique is used in Ethernet.



Since we are talking about IEEE 802.3 standard Ethernet therefore, 0 is expressed by a high-to-low transition, a 1 by the low-to-high transition. In both Manchester Encoding and Differential Manchester, Encoding Baud rate is double of bit rate.

Baud rate = 2* Bit rate

Ethernet LANs consist of network nodes and interconnecting media or link. The network nodes can be of two types:

Data Terminal Equipment (DTE): - Generally, DTEs are the end devices that convert the user information into signals or reconvert the received signals. DTEs devices are: personal computers, workstations, file servers or print servers also referred to as end stations. These devices are either the source or the destination of data frames. The data terminal equipment may be a single piece of equipment or multiple pieces of equipment that are interconnected and perform all the required functions to allow the user to communicate. A user can interact to DTE or DTE may be a user.

Data Communication Equipment (DCE): -DCEs are the intermediate network devices that receive and forward frames across the network. They may be either standalone devices such as repeaters, network switches, routers or maybe communications interface units such as interface cards and modems. The DCE performs functions such as signal conversion, coding and may be a part of the DTE or intermediate equipment.

Currently, these data rates are defined for operation over optical fibers and twisted-pair cables:

i) Fast Ethernet

Fast Ethernet refers to an Ethernet network that can transfer data at a rate of 100 Mbit/s.

ii) Gigabit Ethernet

Gigabit Ethernet delivers a data rate of 1,000 Mbit/s (1 Gbit/s).

iii) 10 Gigabit Ethernet

10 Gigabit Ethernet is the recent generation and delivers a data rate of 10 Gbit/s (10,000 Mbit/s). It is generally used for backbones in high-end applications requiring high data rates.

Ethernet Cabling

Types of Ethernet cabling: *These categories tell you the quality of the cabling. The quality determines, essentially, how much the cable can handle. Here are the categories that you need to know:*

- Cat 3 – used for voice cabling and 10Mb Ethernet
- Cat 5 – used for 10/100Mb Ethernet and works for voice as well

- Cat 5E – Enhanced Cat 5 cabling that helps to prevent cross-talk, works for 10/100Mb and 1000Mb (or Gigabit Ethernet)
- Cat 6 – Like Cat 5E but with larger gauge wires, works for 10/100/1000Mb. This cable is better than Cat 5e for Gigabit Ethernet.
- Cat 7 – Also called Class F, this is fully-shielded cabling and supports up to 600Mhz. This is a relatively new type of cabling and isn't used much.

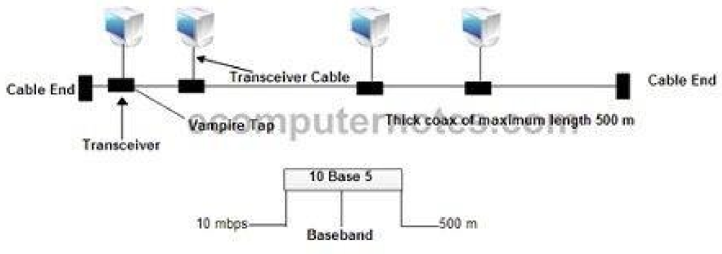
Most companies today are still using and even installing Category 5e as it works for the 100Mb Fast-Ethernet in use on almost every desktop PC.

The following four cable types are the most common among them:

Table : Types of 10 Mbps Ethernets

Name	Cable	Max. segment	Nodes /segment	Signaling technique	Topology used	Cable diameter	Access Method	advantage
10Base5	Thick coax	500 m	100	Baseband (Manchester)	Bus	10	CSMA/CD	Good For back bones
10Base2	Thin coax	200 m	30	Baseband (Manchester)	Bus	5	CSMA/CD	Cheapest system
10Base-T	Twisted pair	100 m	1024	Based band (Manchester)	Star	0.4 to 0.6	CSMA/CD	Easy maintenance
10Base-F	Fiber optics	2000m	1024	Manchester/ on-off	Star	62.5/125 pm	CSMA/CD	Best between buildings

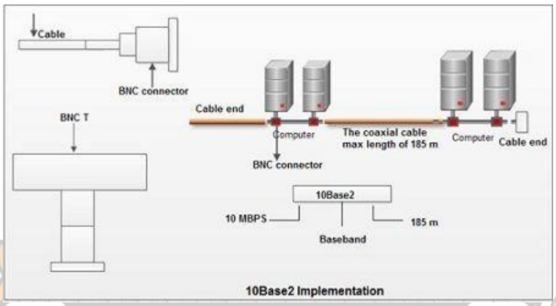
1. **10Base5 cable.** 10Base5 cable or **"Thick Ethernet"** or Thicknet is the cable which is the oldest in the category. it is called as thicknet because of the use of thick coaxial cable. The cable is marked after each 2.5 meters. The thicknet uses bus topology. These marks are provided for attaching tap points. The connections to this cable are made by "Vampire Taps". In this type of connection, a pin is carefully forced half a way into the coaxial cable core. The cable operates at 10Mbps and it can support a maximum segment length of 500 meters. It supports 100 nodes per segment. 10Base5 cable allows, at maximum, five segments (each of max of 500 meter) to be connected. These segments are connected with the help of repeaters. Therefore, four repeaters are allowed to be used providing the effective length of2.5 km.



10Base5 Implementation

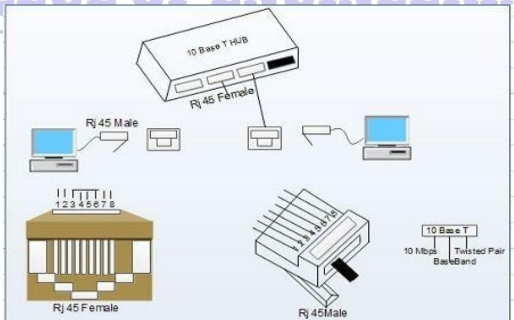
It makes use of transceiver (transmitter/receiver) connected via a vampire tap. The transceiver is responsible for transmitting, receiving and detecting collisions. This transceiver is connected to the station via a transceiver cable that provides separate path for sending and receiving.

2. **10Base2 Cable.** 10Base2 cable also called **"Thin Ethernet" or Thinnet or cheapnet or cheapernet,** was designed after the thick Ethernet cable. This type of cable is usually thin, flexible and bends easily. It also makes use of bus topology. It is also a coaxial cable that is having a smaller diameter than the 10Base5 cable. The connections to this cable are made by BNC connectors or UTP connectors. The connections are created by forming T junctions. These are easier to use and more reliable than vampire connections. The Ethernet based on this type of cables is cheaper and easy to install but it can run for 200 meters and it supports 30 nodes per segment. In both of these network cables, detecting cable breaks, bad taps or loose connections can be a major problem. Here, a special technique, called **"Time Domain Reflectometry"** is used to detect such kind of errors.



10Base2 Implementation

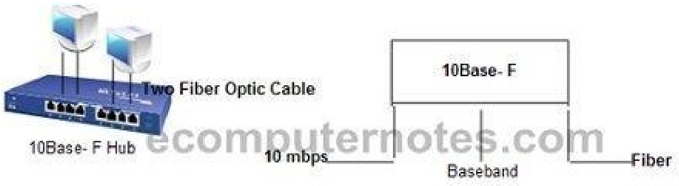
3. **10BaseT Cable.** l0BaseT cable or **"Twisted Pair"** Cable is cheapest and easiest to maintain. This type of cabling is most popular among local area networks. It make use of unshielded twisted pair and provides maximum segment length of 100 m. It make use of start topology. In this type of network, every station is having a wired link to a central device, called "Hub". Telephone company twisted pair cable of category 5 is used in this type of network. This is an older and known technology of connections. It can support 1024 nodes per cable segment. The maximum length of a segment from hub to station can be 150 meters. This type of networks involves the extra cost of hubs.



10BaseT Implementation

4. **l0BaseF Cable.** 10BaseF cable or **"Fiber Optics Cable"** is the most efficient and fastest cable in the category of cables for 802 LANs. The fiber optic cable is very expensive as compared to above discussed cables but it offers a very high data transmission speed and noise immunity. This type of cabling is preferred for running networks between buildings

or widely separated hubs. It has the highest length per cable segment i.e. 2000 meters and it can support 1024 nodes per cable segment.



10Base-F Implementation

Encoding Manchester
Encoding

None of the versions of Ethernet uses straight binary encoding with 0volts for a 0 bit and 5 volts for a 1 bit because it leads to ambiguities. If one station sends the bit string 0001000, others might falsely interpret itas 10000000 or 01000000 because they cannot tell the difference between an idle sender (0 volts) and a 0 bit (0 volts). This problem can be solved by using +1 volts for a 1 and -1 volts for a 0, but there is still the problem of a receiver sampling the signal at a slightly different frequency than the sender used to generate it. Different clock speeds can cause the receiver and sender to get out of synchronization about where the bit boundaries are, especially after a long run of consecutive 0s or along run of consecutive 1s.What is needed is a way for receivers to unambiguously determine the start, end, or middle of each bit without reference to an external clock. Two such approaches are called Manchester encoding and differential Manchester encoding. With Manchester encoding, each bit period is divided into two equal intervals. A binary 1 bit is sent by having the voltage set high during the first interval and low in the second one. A binary 0 is just the reverse: first low and then high. This scheme ensures that every bit period has a transition in the middle, making it easy for the receiver to synchronize with the sender. A disadvantage of Manchester encoding is that it requires twice as much bandwidth as straight binary encoding because the pulses are half the width. For example, to send data at 10 Mbps, the signal has to change 20 million times/sec.

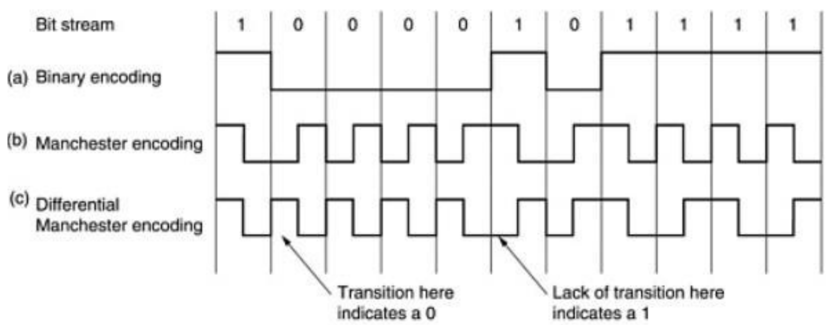


Figure: (a) Binary encoding. (b) Manchester encoding. (c) Differential Manchester encoding.

This a variation ofbasic Manchester encoding. In it, a 1 bit is indicated by the absence of a transition at the start of the interval. A 0 bit is indicated by the presence of a transition at the start of the interval. In both cases, there is a transition in the middle as well. The differential

scheme requires more complex equipment but offers better noise immunity. All Ethernet systems use Manchester encoding due to its simplicity. The high signal is + 0.85voltsand the low signal is - 0.85 volts, giving a DC value of 0 volts. Ethernet does not use differential Manchester encoding, but other LANs (e.g., the802.5 token ring) do use it.

Ethernet(IEEE 802.3) Frame Format

PREAMBLE	SFD	DESTINATION ADDRESS	SOURCE ADDRESS	LENGTH	DATA	CRC
7 Bytes	1 Byte	6 Bytes	6 Bytes	2 Bytes	46 - 1500 Bytes	4 Bytes

IEEE 802.3 ETHERNET Frame Format

- PREAMBLE – Ethernet frame starts with 7-Bytes Preamble. This is a pattern of alternative 0's and 1's which indicates starting of the frame and allow sender and receiver to establish bit synchronization. Initially, PRE (Preamble) was introduced to allow for the loss of a few bits due to signal delays. But today's high-speed Ethernet don't need Preamble to protect the frame bits.
- PRE (Preamble) indicates the receiver that frame is coming and allow the receiver to lock onto the data stream before the actual frame begins.
- Start of frame delimiter (SFD) – This is a 1-Byte field which is always set to 10101011. SFD indicates that upcoming bits are starting of the frame, which is the destination address. Sometimes SFD is considered the part of PRE, this is the reason Preamble is described as 8 Bytes in many places. The SFD warns station or stations that this is the last chance for synchronization.
- Destination Address – This is 6-Byte field which contains the MAC address of machine for which data is destined.
- Source Address – This is a 6-Byte field which contains the MAC address of source machine. As Source Address is always an individual address (Unicast), the least significant bit of first byte is always 0.
- Length – Length is a 2-Byte field, which indicates the length of entire Ethernet frame. This 16-bit field can hold the length value between 0 to 65534, but length cannot be larger than 1500 because of some own limitations of Ethernet.
- Data – This is the place where actual data is inserted, also known as Payload. Both IP header and data will be inserted here if Internet Protocol is used over Ethernet. The maximum data present may be as long as 1500 Bytes. In case data length is less than minimum length i.e. 46 bytes, then padding 0's is added to meet the minimum possible length.
- Cyclic Redundancy Check (CRC) – CRC is 4 Byte field. This field contains a 32-bits hash code of data, which is generated over the Destination Address, Source Address, Length, and Data field. If the checksum computed by destination is not the same as sent checksum value, data received is corrupted.

Brief overview on Extended Ethernet Frame (Ethernet II Frame) :

Standard IEEE 802.3 basic frame format is discussed above in detail. Now let's see the extended Ethernet frame header, using which we can get Payload even larger than 1500 Bytes.

DA [Destination MAC Address] :
6 bytes SA [Source MAC
Address] : 6 bytes Type [0x8870
(Ethertype)] : 2 bytes
DSAP [802.2 Destination Service Access
Point] : 1 byte SSAP [802.2 Source Service
Access Point] : 1 byte
Ctrl [802.2 Control Field] :
1 byte Data [Protocol
Data] :> 46 bytes FCS
[Frame Checksum] : 4 bytes

DA	SA	Type	DSAP	SSAP	Ctrl	Data	FCS
6 Bytes	6 Bytes	2 Bytes	1 Byte	1 Byte	1 Byte	> 46 Bytes	4 Bytes

Proposed ETHERNET Frame Extension

Binary Exponential Back Off Algorithm

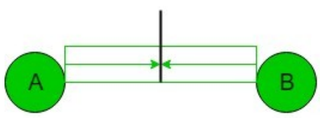
Back-off algorithm is a **collision resolution** mechanism which is used in random access MAC protocols (CSMA/CD). This algorithm is generally used in Ethernet to schedule re-transmissions after collisions.

If a collision takes place between 2 stations, they may restart transmission as soon as they can after the collision. This will always lead to another collision and form an infinite loop of collisions leading to a deadlock. To prevent such scenario back-off algorithm is used.

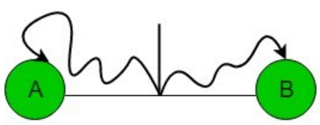
After a collision, time is divided into discrete slots (**Tslot**) whose length is equal to 2t, where t is the maximum propagation delay in the network.

consider an scenario of 2 stations A and B transmitting some data:

At t = 0 , both A and B start transmission



Packets of both A and B collide



Both stations A and B detect collision

The stations involved in the collision randomly pick an integer from the set K i.e {0, 1}. This set is called the contention window. If the sources collide again because they picked the same integer, the contention window size is doubled and it becomes {0, 1, 2, 3}. Now the sources involved in the second collision randomly pick an integer from the set {0, 1, 2, 3} and wait that number of time slots before trying again. Before they try to transmit, they listen to the channel and transmit only if the channel is idle. This causes the source which picked the smallest integer in the contention window to succeed in transmitting its frame.

So, Back-off algorithm defines a *waiting time for the stations involved in collision*, i.e. for how much time the station should wait to re-transmit.

Comparison of Fast and Gigabit Ethernet

Both Fast Ethernet and Gigabit Ethernet are used for network connection. They can work with fiber switch, fiber optic cable, Ethernet cable and some similar devices. The following are some key differences between Fast Ethernet and Gigabit Ethernet.

- The simplest difference between Fast Ethernet vs Gigabit Ethernet is their speed. Fast Ethernet runs at the maximum speed of 100 Mbps and Gigabit Ethernet offers up to 1 Gbps speed which is 10 times faster than Fast Ethernet.
- Round-trip delay of Fast Ethernet is 100-500 bit times. As against, Gigabit Ethernet has the delay of 4000-bit times.
- Configuration problems in Gigabit Ethernet are more complicated than Fast Ethernet. Sometimes Gigabit Ethernet needs high-compatibility fiber switch to work with, for instance, 10gbe switch.
- The distance covered by Fast Ethernet is at most 10 km. However, the Gigabit Ethernet has the limit of 70 km.
- Gigabit Ethernet is more expensive than Fast Ethernet. Upgrading of Fast Ethernet from Standard Ethernet is easy and cost-effective while upgrading of Gigabit Ethernet from Fast Ethernet is complex and expensive.
- Gigabit Ethernet requires specifically designed network devices that can support the standard 1000Mbps data rate like Gigabit Ethernet switch. Fast Ethernet requires no specific network devices.

Basis For Comparison	Fast Ethernet	Gigabit Ethernet
Basic	Offers 100 Mbps speed.	Provide 1 Gbps speed.
Delay	Generate more delay.	Less comparatively.
Configuration	Simple	Complicated and create more errors.
Coverage	Can cover distance up to 10 km.	Has the limit of 70 km.
Relation	Successor of 10-Base-T Ethernet.	A successor of fast Ethernet.
Round trip delay	100-500 bit times	4000 bit times

Question Bank

1. Explain binary exponential back off algorithm.
2. Draw the binary, Manchester and differential Manchester encoding for the bit stream 1010100011.
3. Explain the format of 802.3 MAC frame. Mention the restrictions imposed on minimum lengths of an 802.3 frame.
4. Why do Ethernet systems use Manchester Encoding?
5. Why does Ethernet frame have a minimum size? How does gigabit Ethernet solve this problem?
6. Prove that Slotted ALOHA gives better throughput than pure ALOHA.
7. List the various Ethernet standards.
8. Explain static channel allocation techniques.
9. List the Fast and Gigabit Ethernet standards.
10. What is the rationale behind p-persistent CSMA?
11. Derive expressions for efficiency of ALOHA variants.
12. How is differential Manchester encoding superior to Manchester encoding?