UNIT 2

The Data Link Layer: Design issues: Services Provided to the Network Layer, Framing, Error Control, Flow Control, Error Detection: Parity check, Checksum, CRC, Error Correction: Hamming Code, Linear block codes, FEC.

Elementary Data Link Protocols: simplex protocol, Simplex stop and wait, Simplex protocol for Noisy Channel

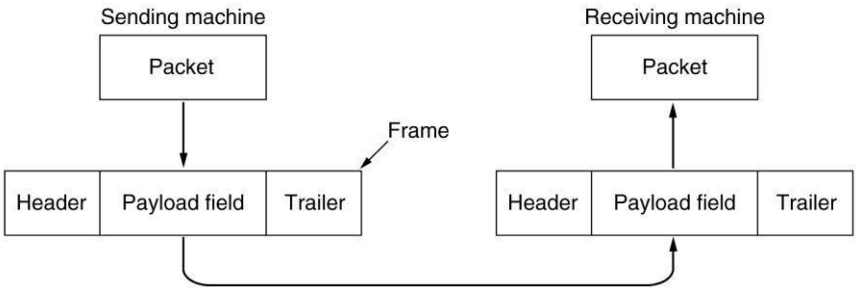
Sliding Window Protocols: One Bit Sliding Window Protocol, A Protocol Using Go-Back-N, Selective Repeat.

Design issues:

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. Determining how the bits of the physical layer are grouped into frames.
3. Dealing with transmission errors.
4. Regulating the flow of data so that slow receivers are not swamped by fast senders.

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 below figure. Frame management forms the heart of what the data link layer does.



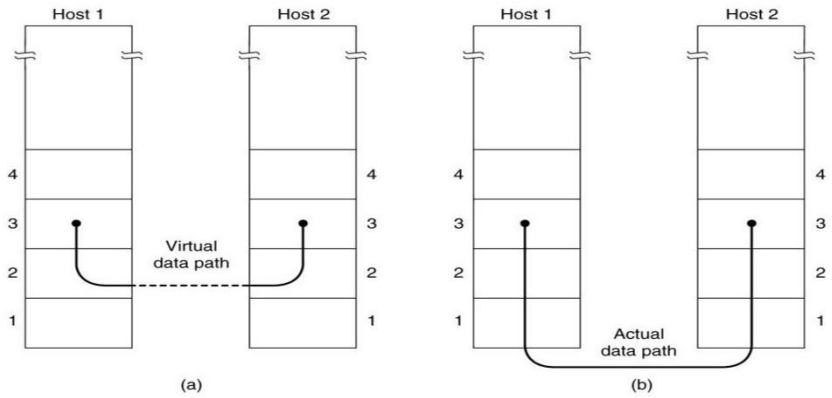
Relationship between packets and frames.

Services Provided to the Network Layer:

The function of the data link layer is to provide services 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. On the source machine is an entity, call it a process, in the network layer that 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 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.

The data link layer can be designed to offer various services. The actual services offered can vary from system to system. Three reasonable possibilities that are commonly provided are:

1. Unacknowledged connectionless service.
2. Acknowledged connectionless service.
3. Acknowledged connection-oriented service.



1. 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 logical connection is established beforehand or released afterward. If a frame is lost due to noise on the line, no attempt is made to detect the loss or recover from it in the data link layer. This class of service is appropriate when the error rate is very low so that recovery is left to higher layers. It is also appropriate for real-time traffic, such as voice, in which late data are worse than bad data. Most LANs use unacknowledged connectionless service in the data link layer.

2. Acknowledged connectionless service.

When this service is offered, there are still no logical connections used, but each frame sent is individually acknowledged. In this way, the sender knows whether a frame has arrived correctly. If it has not arrived within a specified time interval, it can be sent again. This service is useful over unreliable channels, such as wireless systems.

3. 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 indeed received. Furthermore, it guarantees that each frame is received exactly once and that all frames are received in the right order. With connectionless service, in contrast, it is conceivable that a lost acknowledgement causes a packet to be sent several times and thus received several times. Connection-oriented service, in contrast, provides the network layer processes with the equivalent of a reliable bit stream.

When connection-oriented service is used, transfers go through three distinct phases. In the first phase, the connection is established by having both sides initialize variables and counters needed to keep track of which frames have been received and which ones have not. In the second phase, one or more frames are actually transmitted. In the third and final phase, the connection is released, freeing up the variables, buffers, and other resources used to maintain the connection.

Framing

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 a raw bit stream and attempt to deliver it to the destination. This bit stream is not guaranteed to be error free. The number of bits received may be less than, equal to, or more than the number of bits transmitted, and they may have different values. 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 a frame arrives at the destination, the checksum is recomputed. If the newly-computed checksum is different from the one contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it (e.g., discarding the bad frame and possibly also sending back an error report).

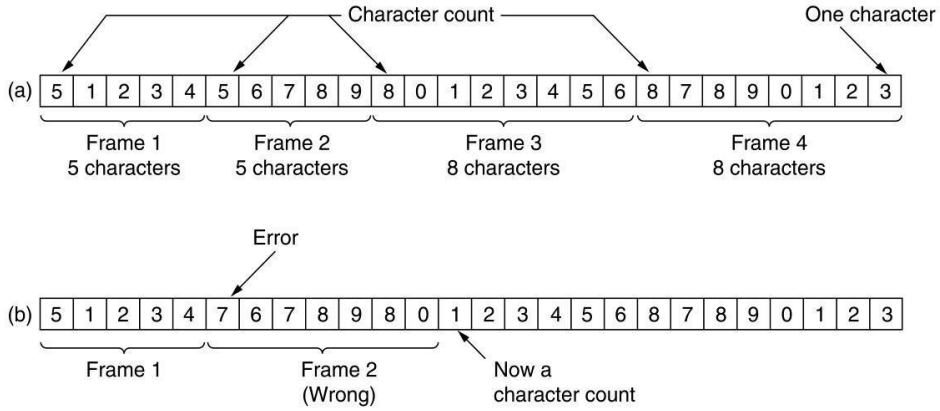
Breaking the bit stream up into frames is more difficult than it at first appears. One way to achieve this framing is to insert time gaps between frames, much like the spaces between words in ordinary text. However, networks rarely make any guarantees about timing, so it is possible these gaps might be squeezed out or other gaps might be inserted during transmission.

Since it is too risky to count on timing to mark the start and end of each frame, other methods have been devised. In this section we will look at four methods:

1. Character count.
2. Flag bytes with byte stuffing (or) character stuffing.
3. Starting and ending flags, with bit stuffing.
4. Physical layer coding violations.

1. Character count

The first framing method 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 and hence where the end of the frame is. This technique is shown in Fig.-(a) for four frames of sizes 5, 5, 8, and 8 characters, respectively.



The trouble with this algorithm is that the count can be garbled by a transmission error. For example, if the character count of 5 in the second frame of Fig-(b) becomes a 7, the destination will get out of synchronization and will be unable to locate the start of the next frame. Even if the checksum is incorrect so the destination knows that the frame is bad, it still has no way of telling where the next frame starts. Sending a frame back to the source asking for a retransmission does not help either, since the destination does not know how many characters to skip over to get to the start of the retransmission. For this reason, the character count method is rarely used anymore.

2. Flag bytes with byte stuffing (or) character stuffing.

The second framing method gets around the problem of resynchronization after an error by having each frame start and end with special bytes. In the past, the starting and ending bytes were different, but in recent years most protocols have used the same byte, called a flag byte, as both the starting and ending delimiter, as shown in below Fig.-(a) as FLAG. In this way, if the receiver ever loses synchronization, it can just search for the flag byte to find the end of the current frame. Two consecutive flag bytes indicate the end of one frame and start of the next one.

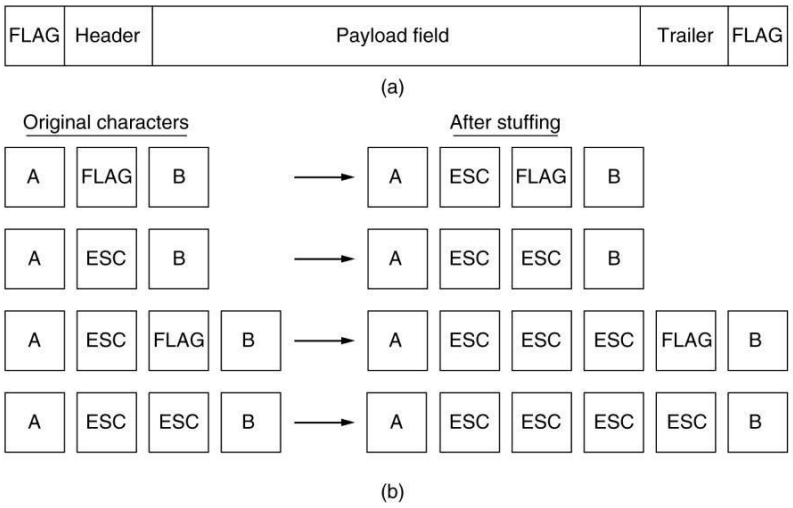


Fig.-(b). In all cases, the byte sequence delivered after destuffing is exactly the same as the original byte sequence.

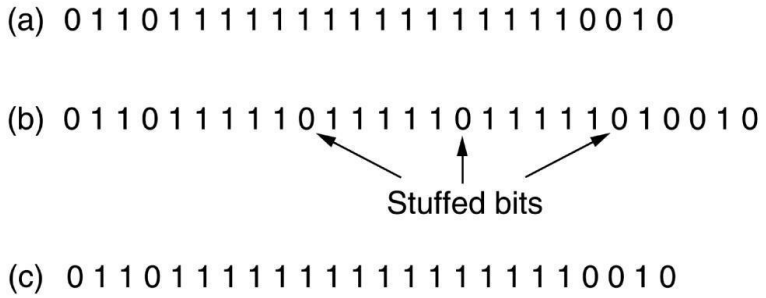
A major disadvantage of using this framing method is that it is closely tied to the use of 8-bit characters

3. Starting and ending flags, with bit stuffing.

The new technique allows data frames to contain an arbitrary number of bits and allows character codes with an arbitrary number of bits per character. It works like this. Each frame begins and ends with a special bit pattern, 01111110 (in fact, a flag byte). Whenever the sender's data link layer encounters five consecutive 1s in the data, it automatically stuffs a 0 bit into the outgoing bit stream. This bit stuffing is analogous to byte stuffing, in which an escape byte is stuffed into the outgoing character stream before a flag byte in the data.

When the receiver sees five consecutive incoming 1 bits, followed by a 0 bit, it automatically destuffs (i.e., deletes) the 0 bit. Just as byte stuffing is completely transparent to the network layer in both computers, so is bit stuffing. If the user data contain the flag pattern, 01111110, this flag is transmitted as 011111010 but stored in the receiver's memory as 01111110.

Figure gives an example of bit stuffing.



(a) The original data.
(b) The data as they appear on the line.
(c) The data as they are stored in the receiver's memory after destuffing.

With bit stuffing, the boundary between two frames can be unambiguously recognized by the flag pattern. Thus, if the receiver loses track of where it is, all it has to do is scan the input for flag sequences, since they can only occur at frame boundaries and never within the data.

As a final note on framing, many data link protocols use a combination of a character count with one of the other methods for extra safety. When a frame arrives, the count field is used to locate the end of the frame. Only if the appropriate delimiter is present at that position and the checksum is correct is the frame accepted as valid. Otherwise, the input stream is scanned for the next delimiter.

Error Control

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. Typically, the protocol calls for the receiver to send back special control frames bearing positive or negative acknowledgements about the incoming frames. If the sender receives a positive acknowledgement about a frame, it knows the frame has arrived safely. On the other hand, a negative acknowledgement means that something has gone wrong, and the frame must be transmitted again.

An additional complication comes from the possibility that hardware troubles may cause a frame to vanish completely (e.g., in a noise burst). In this case, the receiver will not react at all, since it has no reason to react. It should be clear that a protocol in which the sender transmits a frame and then waits for an acknowledgement, positive or negative, will hang forever if a frame is ever lost due to, for example, malfunctioning hardware.

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 expire after an interval long enough for the frame to reach the destination, be processed there, and have the acknowledgement propagate back to the sender. Normally, the frame will be correctly received and the acknowledgement will get back before the timer runs out, in which case the timer will be canceled.

However, if either the frame or the acknowledgement is lost, the timer will go off, alerting the sender to a potential problem. The obvious solution is to just transmit the frame again. However, when frames may be transmitted multiple times there is a danger that the receiver will accept the same frame two or more times and pass it to the network layer more than once. To prevent this from happening, it is generally necessary to assign sequence numbers to outgoing frames, so that the receiver can distinguish retransmissions 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 once, no more and no less, is an important part of the data link layer's duties.

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 the receiver can accept them. This situation can easily occur when the sender is running on a fast (or lightly loaded) computer and the receiver is running on a slow (or heavily loaded) machine. The sender keeps pumping the frames out at a high rate until the receiver is completely swamped. Even if the transmission is error free, at a certain point the receiver will simply be unable to handle the frames as they arrive and will start to lose some.

Two approaches are commonly used. In the first one, feedback-based flow control, the receiver sends back information to the sender giving it permission to send more data or at least telling the sender how

the receiver is doing. In the second one, rate-based flow control, the protocol has a built-in mechanism that limits the rate at which senders may transmit data, without using feedback from the receiver.

Various feedback-based flow control schemes are known, but most of them use the same basic principle. The protocol contains well-defined rules about when a sender may transmit the next frame. These rules often prohibit frames from being sent until the receiver has granted permission, either implicitly or explicitly. For example, when a connection is set up, the receiver might say: "You may send me n frames now, but after they have been sent, do not send any more until I have told you to continue."

Error Detection and Correction

Error-Correcting Codes

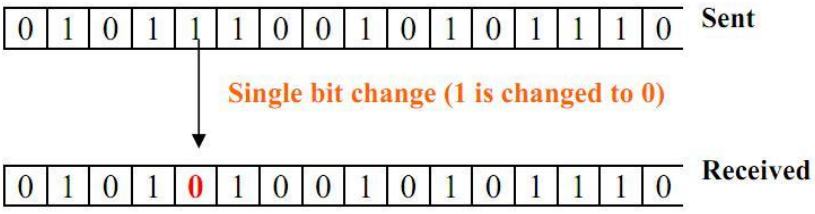
Two basic strategies for dealing with errors. One way is to include enough redundant information along with each block of data sent, to enable the receiver to deduce what the transmitted data must have been. The other way is to include only enough redundancy to allow the receiver to deduce that an error occurred, but not which error, and have it request a retransmission. The former strategy uses error-correcting codes and the latter uses error-detecting codes. The use of error-correcting codes is often referred to as forward error correction.

To understand how errors can be handled, it is necessary to look closely at what an error really is. Normally, a frame consists of m data (i.e., message) bits and r redundant, or check, bits. Let the total length be n (i.e., n = m + r). An n-bit unit containing data and check bits is often referred to as an n-bit codeword.

Type of Errors:

Single-bit Error

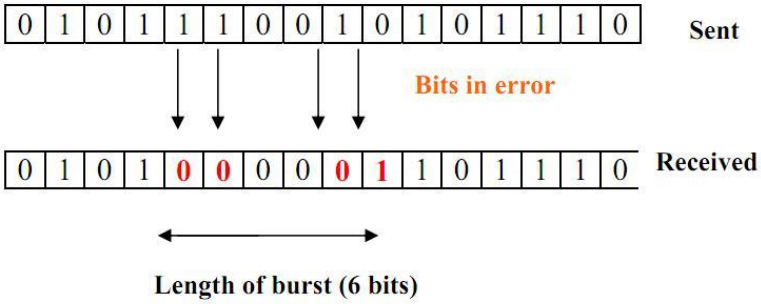
The term single-bit error means that only one bit of given data unit (such as a byte, character, or data unit) is changed from 1 to 0 or from 0 to 1 as shown in



Single bit errors are least likely type of errors in serial data transmission. To see why,I magine a sender sends data at 10 Mbps.This means that each bit lasts only for 0.1 μs (micro-second). For a single bit error to occur noise must have duration of only 0.1 μs (micro-second), which is very rare. However, a single-bit error can happen if we are having a parallel data transmission. For example, if 16 wires are used to send all 16 bits of a word at the same time and one of the wires is noisy, one bit is corrupted in each word.

Burst Error

The term burst error means that two or more bits in the data unit have changed from 0 to 1 or vice-versa. Note that burst error doesn't necessary means that error occurs in consecutive bits. The length of the burst error is measured from the first corrupted bit to the last corrupted bit. Some bits in between may not be corrupted.



Burst errors are mostly likely to happen in serial transmission. The duration of the noise is normally longer than the duration of a single bit, which means that the noise affects data; it affects a set of bits as shown in Fig. 3.2.2. The number of bits affected depends on the data rate and duration of noise.

Error Correction can be handled in two ways.

- One is when an error is discovered; the receiver can have the sender retransmit the entire data unit. This is known as backward error correction.

- In the other, receiver can use an error-correcting code, which automatically corrects certain errors. This is known as forward error correction.

In theory it is possible to correct any number of errors atomically. Error-correcting codes are more sophisticated than error detecting codes and require more redundant bits. The number of bits required to correct multiple-bit or burst error is so high that in most of the cases it is inefficient to do so. For this reason, most error correction is limited to one, two or at the most three-bit errors.

Single-bit error correction

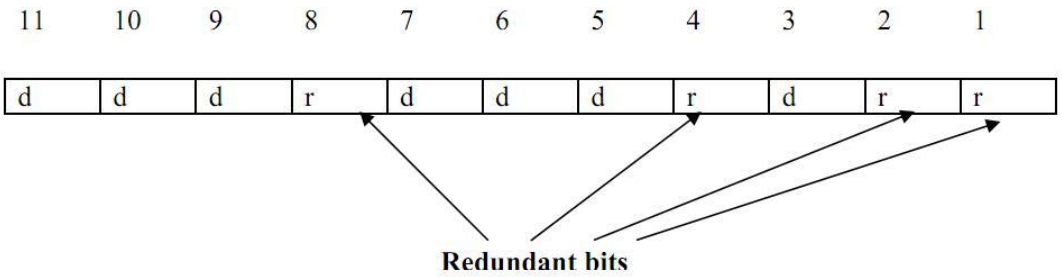
Concept of error-correction can be easily understood by examining the simplest case of single-bit errors. As we have already seen that a single-bit error can be detected by addition of a parity bit (VRC) with the data, which needed to be send. A single additional bit can detect error, but it's not sufficient enough to correct that error too. For correcting an error one has to know the exact position of error, i.e. exactly which bit is in error (to locate the invalid bits). For example, to correct a single-bit error in an ASCII character, the error correction must determine which one of the seven bits is in error. To this, we have to add some additional redundant bits.

To calculate the numbers of redundant bits (r) required to correct d data bits, let us find out the relationship between the two. So we have (d+r) as the total number of bits, which are to be transmitted; then r must be able to indicate at least d+r+1 different values. Of these, one value means no error, and remaining d+r values indicate error location of error in each of d+r locations. So, d+r+1 states must be distinguishable by r bits, and r bits can indicates 2^r states. Hence, 2^r must be greater than d+r+1.

$$\mathbf{2^r >= d+r+1}$$

The value of r must be determined by putting in the value of d in the relation. For example, if d is 7, then the smallest value of r that satisfies the above relation is 4. So the total bits, which are to be transmitted is 11 bits (d+r = 7+4 =11). So the total bits, which are to be transmitted is 11 bits (d+r = 7+4 =11).

Now let us examine how we can manipulate these bits to discover which bit is in error. A technique developed by R.W.Hamming provides a practical solution. The solution or coding scheme he developed is commonly known as Hamming Code. Hamming code can be applied to data units of any length and uses the relationship between the data bits and redundant bits as discussed.



Positions of redundancy bits in hamming code

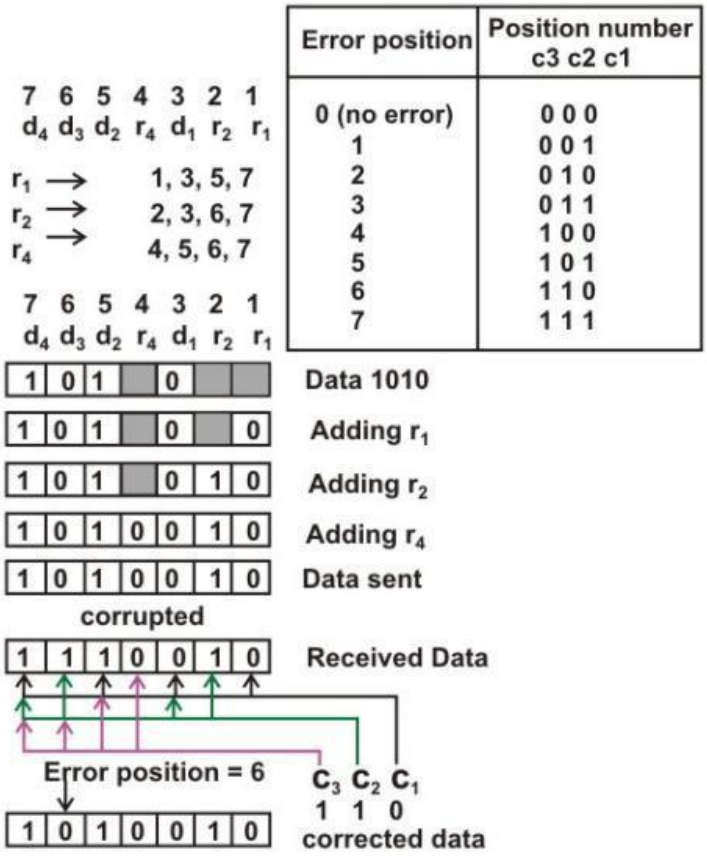
Basic approach for error detection by using Hamming code is as follows:

• To each group of m information bits k parity bits are added to form (m+k) bit code as shown in above Fig.

• Location of each of the (m+k) digits is assigned a decimal value.

• The k parity bits are placed in positions 1, 2… 2k-1 positions.–K parity checks are performed on selected digits of each codeword.

• At the receiving end the parity bits are recalculated. The decimal value of the k parity bits provides the bit-position in error, if any.



Use of Hamming code for error correction for a 4-bit data

Figure shows how hamming code is used for correction for 4-bit numbers ($d_4d_3d_2d_1$) with the help of three redundant bits ($r_3r_2r_1$). For the example data 1010, first $r_1(0)$ is calculated considering the parity of the bit positions, 1, 3, 5 and 7. Then the parity bits r_2 is calculated considering bit positions 2, 3, 6 and 7. Finally, the parity bits r_4 is calculated considering bit positions 4, 5, 6 and 7 as shown. If any corruption occurs in any of the transmitted code 1010010, the bit position in error can be found out by calculating $r_3r_2r_1$ at the receiving end. For example, if the received code word is 1110010, the recalculated value of $r_3r_2r_1$ is 110, which indicates that bit position in error is 6, the decimal value of 110.

Error-Detecting Codes

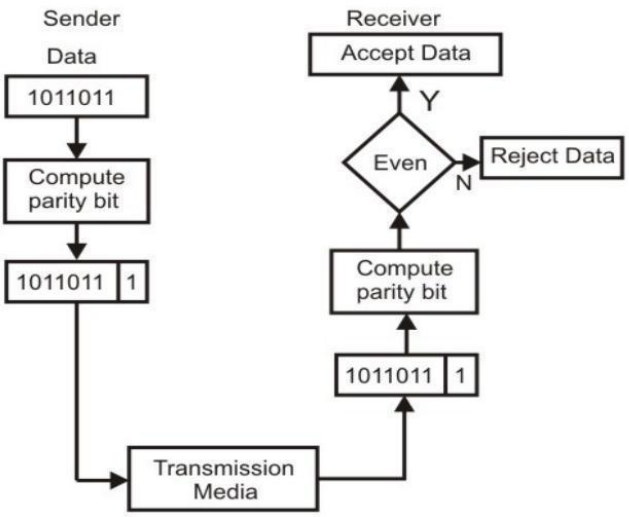
Basic approach used for error detection is the use of redundancy, where additional bits are added to facilitate detection and correction of errors. Popular techniques are:

- Simple Parity check
- Two-dimensional Parity check
- Checksum
- Cyclic redundancy check

- Simple Parity Checking or One-dimension Parity Check

The most common and least expensive mechanism for error- detection is the simple parity check. In this technique, a redundant bit called parity bit, is appended to every data unit so that the number of 1s in the unit (including the parity becomes even).

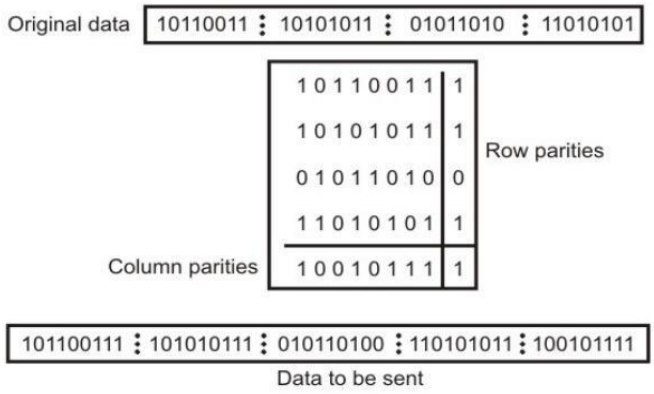
Blocks of data from the source are subjected to a check bit or Parity bit generator form, where a parity of 1 is added to the block if it contains an odd number of 1's (ON bits) and 0 is added if it contains an even number of 1's. At the receiving end the parity bit is computed from the received data bits and compared with the received parity bit, as shown in Fig. This scheme makes the total number of 1's even, that is why it is called even parity checking.



Even-parity checking scheme

- Two-dimension Parity Check

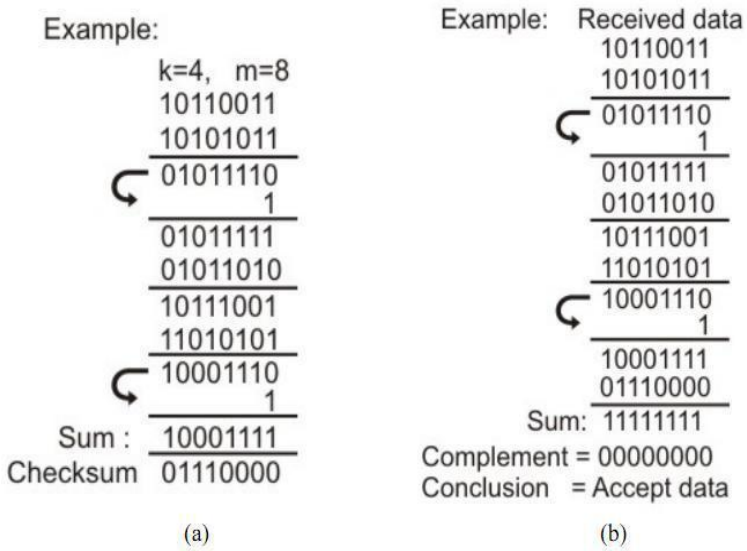
Performance can be improved by using two-dimensional parity check, which organizes the block of bits in the form of a table. Parity check bits are calculated for each row, which is equivalent to a simple parity check bit. Parity check bits are also calculated for all columns then both are sent along with the data. At the receiving end these are compared with the parity bits calculated on the received data.



Two-dimension Parity Checking

Checksum

In checksum error detection scheme, the data is divided into k segments each of m bits. In the sender's end the segments are added using 1's complement arithmetic to get the sum. The sum is complemented to get the checksum. The checksum segment is sent along with the data segments as shown in Fig. (a). At the receiver's end, all received segments are added using 1's complement arithmetic to get the sum. The sum is complemented. If the result is zero, the received data is accepted; otherwise discarded, as shown in Fig. (b).



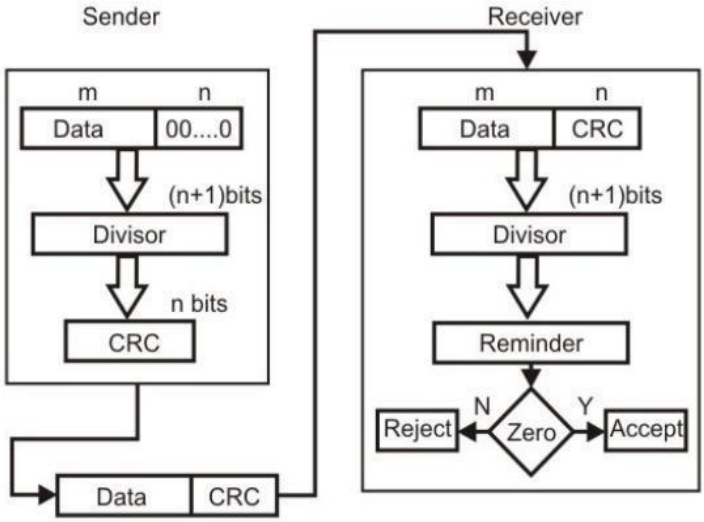
(a) Sender's end for the calculation of the checksum, (b) Receiving end for checking the checksum

Cyclic Redundancy Checks (CRC)

This Cyclic Redundancy Check is the most powerful and easy to implement technique. Unlike checksum scheme, which is based on addition, CRC is based on binary division. In CRC, a sequence of redundant bits, called **cyclic redundancy check bits**, are appended to the end of data unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number. At the 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 correct and is therefore accepted. A remainder indicates that the data unit has been damaged in transit and therefore must be rejected. The generalized technique can be explained as follows.

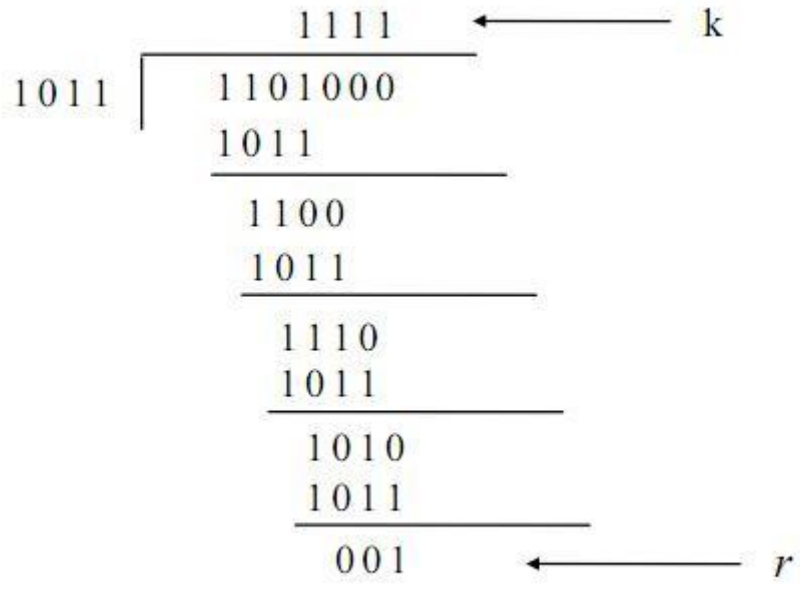
If a k bit message is to be transmitted, the transmitter generates an r-bit sequence, known as Frame Check Sequence (FCS) so that the (k+r) bits are actually being transmitted. Now this r-bit FCS is generated by

dividing the original number, appended by r zeros, by a predetermined number. This number, which is (r+1) bit in length, can also be considered as the coefficients of a polynomial, called Generator Polynomial. The remainder of this division process generates the r-bit FCS. On receiving the packet, the receiver divides the (k+r) bit frame by the same predetermined number and if it produces no remainder, it can be assumed that no error has occurred during the transmission. Operations at both the sender and receiver end are shown in Fig.



Basic scheme for Cyclic Redundancy Checking

This mathematical operation performed is illustrated in the below Fig. by dividing a sample 4-bit number by the coefficient of the generator polynomial x3+x+1, which is 1011, using the modulo-2 arithmetic. Modulo-2 arithmetic is a binary addition process without any carry over, which is just the Exclusive-OR operation. Consider the case where k=1101. Hence we have to divide 1101000 (i.e. k appended by 3 zeros) by 1011, which produces the remainder r=001, so that the bit frame (k+r) =1101001 is actually being transmitted through the communication channel. At the receiving end, if the received number, i.e., 1101001 is divided by the same generator polynomial 1011 to get the remainder as 000, it can be assumed that the data is free of errors.



Cyclic Redundancy Checks (CRC)

The transmitter can generate the CRC by using a feedback shift register circuit. The same circuit can also be used at the receiving end to check whether any error has occurred. All the values can be

expressed as polynomials of a dummy variable X. For example, for P = 11001 the corresponding polynomial is X4+X3+1. A polynomial is selected to have at least the following properties:

- It should not be divisible by X.
- It should not be divisible by (X+1).

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

CRC process can be expressed as XnM(X)/P(X) = Q(X) + R(X) / P(X)

Commonly used divisor polynomials are:

- CRC-16 = X16 + X15 + X2 + 1
- CRC-CCITT = X16 + X12 + X5 + 1
- CRC-32 = X32 + X26 + X23 + X22 + X16 + X12 + X11 + X10 + X8 + X7 + X5 + X4 + X2 + 1

This mathematical operation performed is illustrated in the below Fig. by dividing a sample 4-bit number by the coefficient of the generator polynomial x3+x+1, which is 1011, using the modulo-2 arithmetic. Modulo-2 arithmetic is a binary addition process without any carry over, which is just the Exclusive-

Performance

CRC is a very effective error detection technique. If the divisor is chosen according to the previously mentioned rules, its performance can be summarized as follows:

- CRC can detect all single-bit errors
- CRC can detect all double-bit errors (three 1's)
- CRC can detect any odd number of errors (X+1)
- CRC can detect all burst errors of less than the degree of the polynomial.
- CRCdetects most of the larger burst errors with a high probability.
  For example CRC-12 detects 99.97% of errors with a length 12 or more.

A serious problem occurs with this method when binary data, such as object programs or floating-point numbers, are being transmitted. It may easily happen that the flag byte's bit pattern occurs in the data. This situation will usually interfere with the framing. One way to solve this problem is to have the sender's data link layer insert a special escape byte (ESC) just before each "accidental" flag byte in the data. The data link layer on the receiving end removes the escape byte before the data are given to the network layer. This technique is called byte stuffing or character stuffing. Thus, a framing flag byte can be distinguished from one in the data by the absence or presence of an escape byte before it.

Of course, the next question is: What happens if an escape byte occurs in the middle of the data? The answer is that it, too, is stuffed with an escape byte. Thus, any single escape byte is part of an escape sequence, whereas a doubled one indicates that asingle escape occurred naturally in the data. Some examples are shown in above

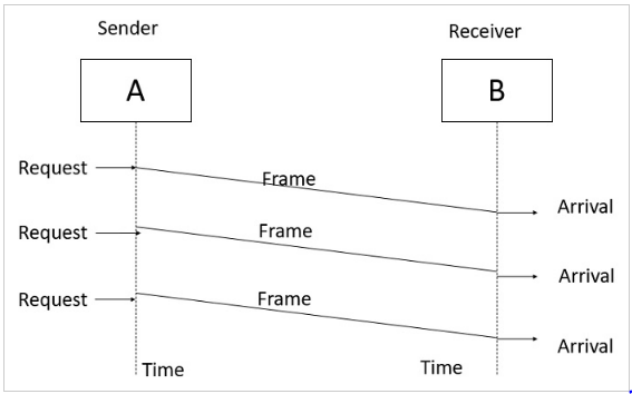
Elementary Data Link Protocols

An Unrestricted Simplex Protocol

The protocol consists of two distinct procedures, a sender and a receiver. The sender runs in the data link layer of the source machine, and the receiver runs in the data link layer of the destination machine. No sequence numbers or acknowledgements are used here.The only event type possible is frame_arrival (i.e., the arrival of an undamaged frame).

Data transmitting is carried out in one direction only. The transmission (Tx) and receiving (Rx) are always ready and the processing time can be ignored. In this protocol, infinite buffer space is available, and no errors are occurring that is no damage frames and no lost frames.

The Unrestricted Simplex Protocol is diagrammatically represented as follows –



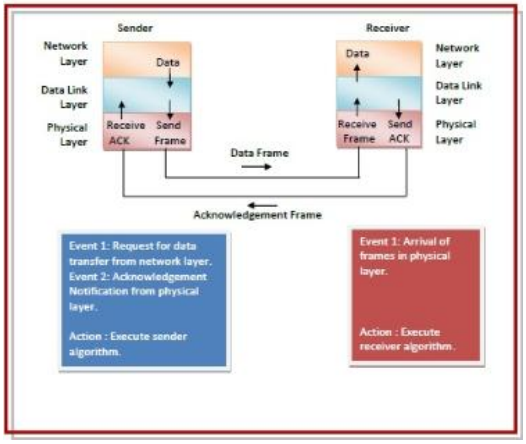
A Simplex Stop-and-Wait Protocol

Stop – and – Wait protocol is data link layer protocol for transmission of frames over noiseless channels. It provides unidirectional data transmission with flow control facilities but without error control facilities.

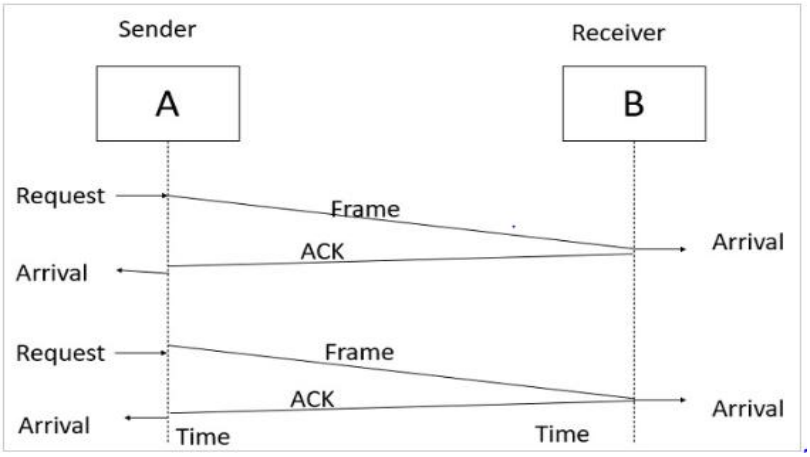
This protocol takes into account the fact that the receiver has a finite processing speed. If data frames arrive at the receiver's end at a rate which is greater than its rate of processing, frames be dropped out. In order to avoid this, the receiver sends an acknowledgement for each frame upon its arrival. The sender sends the next frame only when it has received a positive acknowledgement from the receiver that it is available for further data processing.

Design

- **Sender Site**: The data link layer in the sender site waits for the network layer for a data packet. It then checks whether it can send the frame. If it receives a positive notification from the physical layer, it makes frames out of the data and sends it. It then waits for an acknowledgement before sending the next frame.
- **Receiver Site**: The data link layer in the receiver site waits for a frame to arrive. When it arrives, the receiver processes it and delivers it to the network layer. It then sends an acknowledgement back to the sender.



The Simplex Stop and Wait Protocol is diagrammatically represented as follows −

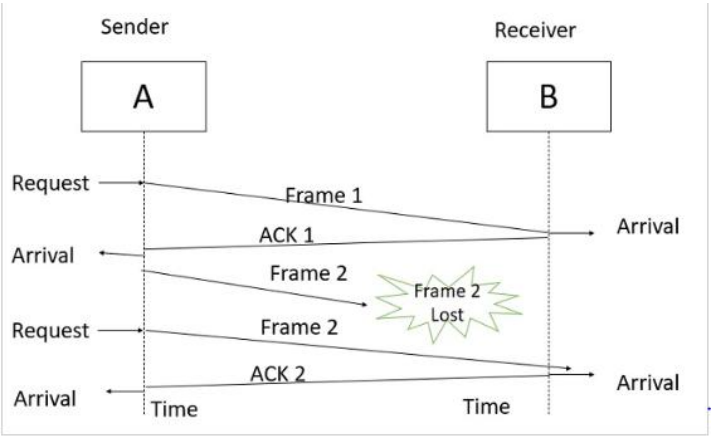


Simplex Protocol for Noisy Channel

Data transfer is only in one direction, consider separate sender and receiver, finite processing capacity and speed at the receiver, since it is a noisy channel, errors in data frames or acknowledgement frames are expected. Every frame has a unique sequence number.

After a frame has been transmitted, the timer is started for a finite time. Before the timer expires, if the acknowledgement is not received , the frame gets retransmitted, when the acknowledgement gets corrupted or sent data frames gets damaged, how long the sender should wait to transmit the next frame is infinite.

The Simplex Protocol for Noisy Channel is diagrammatically represented as follows −



Sliding Window Protocol

In the previous protocols, data frames were transmitted in one direction only. In most practical situations, there is a need for transmitting data in both directions. One way of achieving full-duplex data transmission is to have two separate communication channels and use each one for simplex data traffic (in different directions). If this is done, we have two separate physical circuits, each with a "forward" channel (for data) and a "reverse" channel (for acknowledgements). In both cases the bandwidth of the reverse channel is almost entirely wasted. In effect, the user is paying for two circuits but using only the capacity of one.

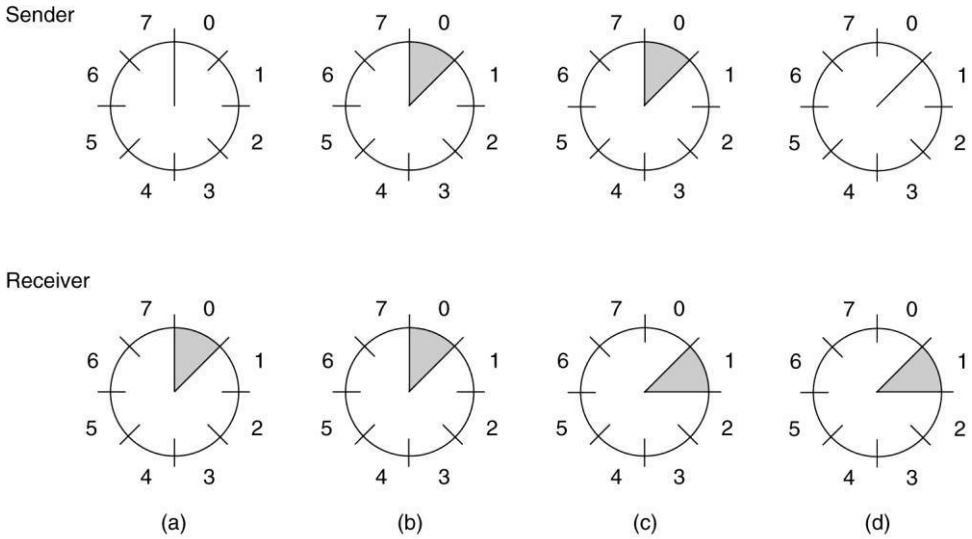
A better idea is to use the same circuit for data in both directions. After all, in protocols 2 and 3 it was already being used to transmit frames both ways, and the reverse channel has the same capacity as the forward channel

When a data frame arrives, instead of immediately sending a separate control frame, the receiver restrains itself and waits until the network layer passes it the next packet. The acknowledgement is attached to the outgoing data frame (using the ack field in the frame header). In effect, the acknowledgement gets a free ride on the next outgoing data frame. The technique of temporarily delaying outgoing acknowledgements so that they can be hooked onto the next outgoing data frame is known as **piggybacking.**

The principal advantage of using piggybacking over having distinct acknowledgement frames is a better use of the available channel bandwidth. The ack field in the frame header costs only a few bits, whereas a separate frame would need a header, the acknowledgement, and a checksum

The next three protocols are bidirectional protocols that belong to a class called **sliding window protocols**. The essence of all sliding window protocols is that at any instant of time, the sender maintains a set of sequence numbers corresponding to frames it is permitted to send. These frames are said to fall within the **sending window**. Similarly, the receiver also maintains a **receiving window** corresponding to the set of frames it is permitted to accept. The sender's window and the receiver's window need not have the same lower and upper limits or even have the same size. In some protocols they are fixed in size, but in others they can grow or shrink over the course of time as frames are sent and received.

The sequence numbers within the sender's window represent frames that have been sent or can be sent but are as yet not acknowledged. Whenever a new packet arrives from the network layer, it is given the next highest sequence number, and the upper edge of the window is advanced by one. When an acknowledgement comes in, the lower edge is advanced by one. In this way the window continuously maintains a list of unacknowledged frames. Figure shows an example.



A sliding window of size 1, with a 3-bit sequence number. (a) Initially. (b) After the first frame has been sent. (c) After the first frame has been received. (d) After the first acknowledgement has been received.

Since frames currently within the sender's window may ultimately be lost or damaged in transit, the sender must keep all these frames in its memory for possible retransmission. Thus, if the maximum window size is n, the sender needs n buffers to hold the unacknowledged frames. If the window ever grows to its maximum size, the sending data link layer must forcibly shut off the network layer until another buffer becomes free.

The receiving data link layer's window corresponds to the frames it may accept. Any frame falling outside the window is discarded without comment. When a frame whose sequence number is equal to the lower edge of the window is received, it is passed to the network layer, an acknowledgement is generated, and the window is rotated by one. Unlike the sender's window, the receiver's window always remains at its initial size. Note that a window size of 1 means that the data link layer only accepts frames in order, but for larger windows this is not so. The network layer, in contrast, is always fed data in the proper order, regardless of the data link layer's window size.

A One-Bit Sliding Window Protocol

Under normal circumstances, one of the two data link layers goes first and transmits the first frame. In other words, only one of the data link layer programs should contain the to_physical_layer and start_timer procedure calls outside the main loop. In the event that both data link layers start off simultaneously, a peculiar situation arises, as discussed later. The starting machine fetches the first packet from its network layer, builds a frame from it, and sends it. When this (or any) frame arrives, the receiving data link layer checks to see if it is a duplicate, just as in protocol 3. If the frame is the one expected, it is passed to the network layer and the receiver's window is slid up.

The acknowledgement field contains the number of the last frame received without error. If this number agrees with the sequence number of the frame the sender is trying to send, the sender knows it is done with the frame stored in buffer and can fetch the next packet from its network layer. If the sequence number disagrees, it must continue trying to send the same frame. Whenever a frame is received, a frame is also sent back.

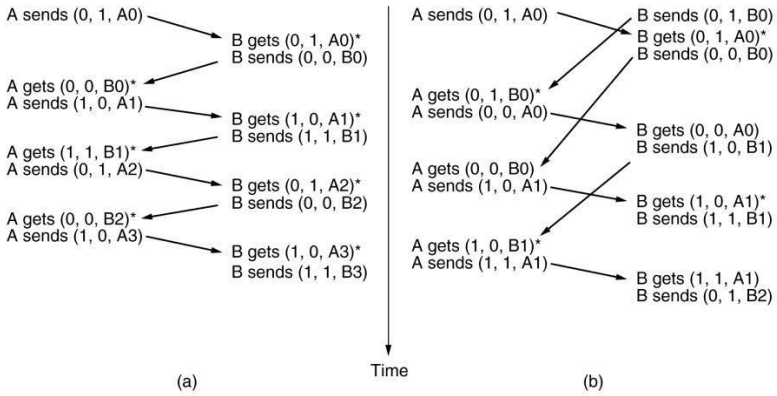
Now let us examine protocol 4 to see how resilient it is to pathological scenarios. Assume that computer A is trying to send its frame 0 to computer B and that B is trying to send its frame 0 to A. Suppose that A sends a frame to B, but A's timeout interval is a little too short. Consequently, A may time out repeatedly, sending a series of identical frames, all with seq = 0 and ack = 1.

When the first valid frame arrives at computer B, it will be accepted and frame_expected will be set to 1. All the subsequent frames will be rejected because B is now expecting frames with sequence number 1, not 0. Furthermore, since all the duplicates have ack = 1 and B is still waiting for an acknowledgement of 0, B will not fetch a new packet from its network layer.

After every rejected duplicate comes in, B sends A a frame containing seq = 0 and ack = 0. Eventually, one of these arrives correctly at A, causing A to begin sending the next packet. No combination of lost frames or premature timeouts can cause the protocol to deliver duplicate packets to either network layer, to skip a packet, or to deadlock.

However, a peculiar situation arises if both sides simultaneously send an initial packet. This synchronization difficulty is illustrated by Fig. In part (a), the normal operation of the protocol is shown. In (b) the peculiarity is illustrated. If B waits for A's first frame before sending one of its own, the sequence is as shown in (a), and every frame is accepted. However, if A and B simultaneously initiate communication, their first frames cross, and the data link layers then get into situation (b). In (a) each frame arrival brings a new packet for the network layer; there are no duplicates. In (b) half of the

frames contain duplicates, even though there are no transmission errors. Similar situations can occur as a result of premature timeouts, even when one side clearly starts first. In fact, if multiple premature timeouts occur, frames may be sent three or more times.

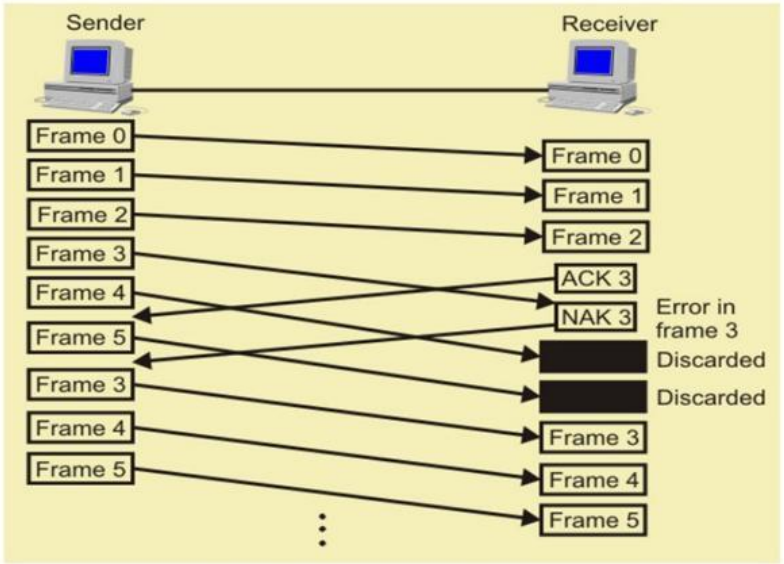


Two scenarios for protocol 4. (a) Normal case. (b) Abnormal case. The notation is (seq, ack, packet number). An asterisk indicates where a network layer accepts a packet.

A Protocol Using Go Back N

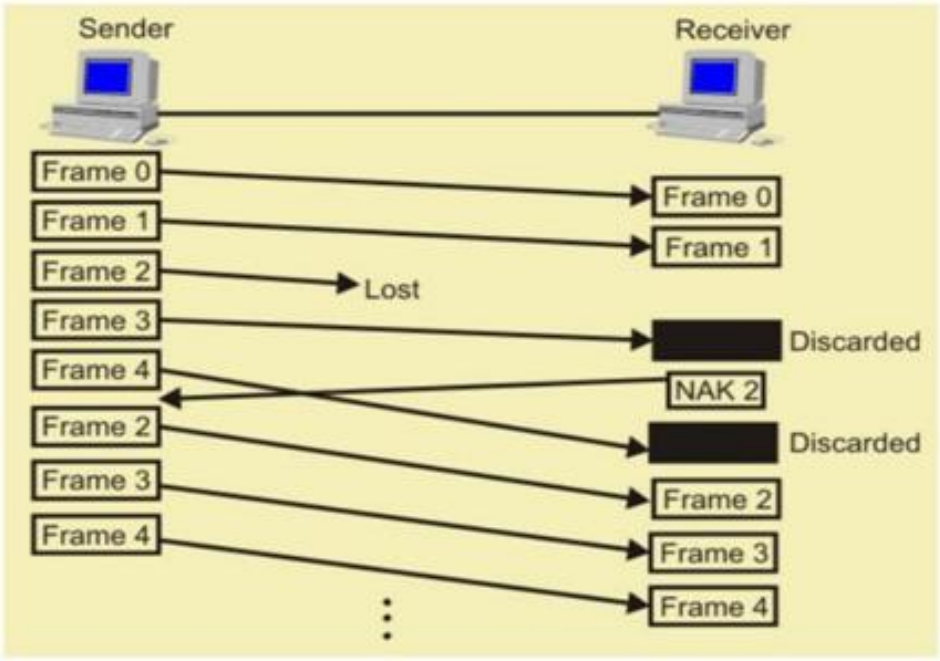
The most popular protocol is the go-back-N, where the sender sends the frames continuously without waiting for acknowledgement.. As the receiver receives the frames, it keeps on sending ACKs or a NACK, in case a frame is incorrectly received. When the sender receives a NACK, it retransmits the frame in error plus all the succeeding frames as shown in Fig. Hence, the name of the protocol is go-back-N ARQ. If a frame is lost, the receiver sends NAK after receiving the next frame as shown in Fig.. In case there is long delay before sending the NAK, the sender will resend the lost frame after its timer times out. If the ACK frame sent by the receiver is lost, the sender resends the frames after its timer times out as shown in Fig.

Assuming full-duplex transmission, the receiving end sends piggybacked acknowledgement by using some number in the ACK field of its data frame. Let us assume that a 3-bit sequence number is used and suppose that a station sends frame 0 and gets back an RR1, and then sends frames 1, 2, 3, 4, 5, 6, 7, 0 and gets another RR1.This might either mean that RR1 is a cumulative ACK or all 8 frames were damaged. This ambiguity can be overcome if the maximum window size is limited to 7, i.e. for a k-bit sequence number field it is limited to 2k-1. The number N (=2k-1) specifies how many frames can be sent without receiving acknowledgement.

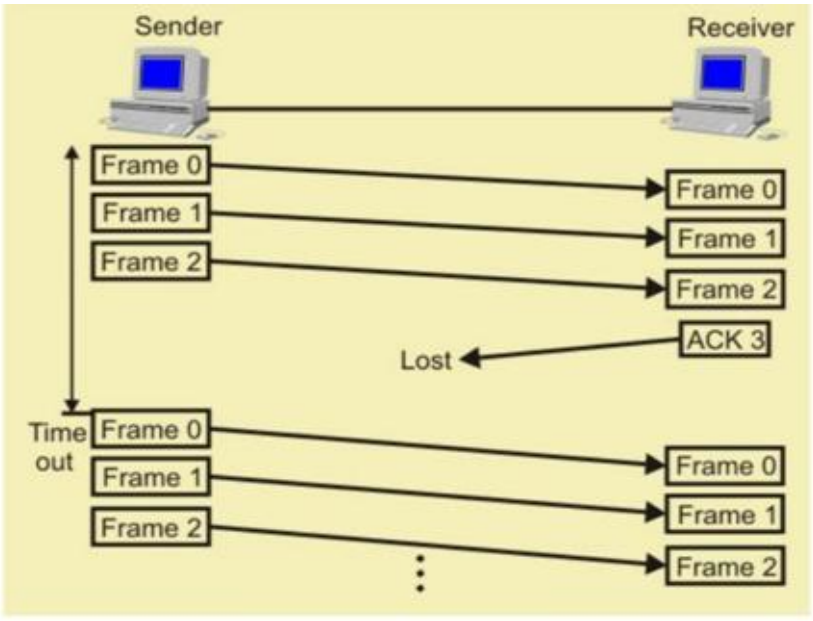


Frames in error in go-Back-N

If no acknowledgement is received after sending N frames, the sender takes the help of a timer. After the time-out, it resumes retransmission. The go-back-N protocol also takes care of damaged frames and damaged ACKs. This scheme is little more complex than the previous one but gives much higher throughput.



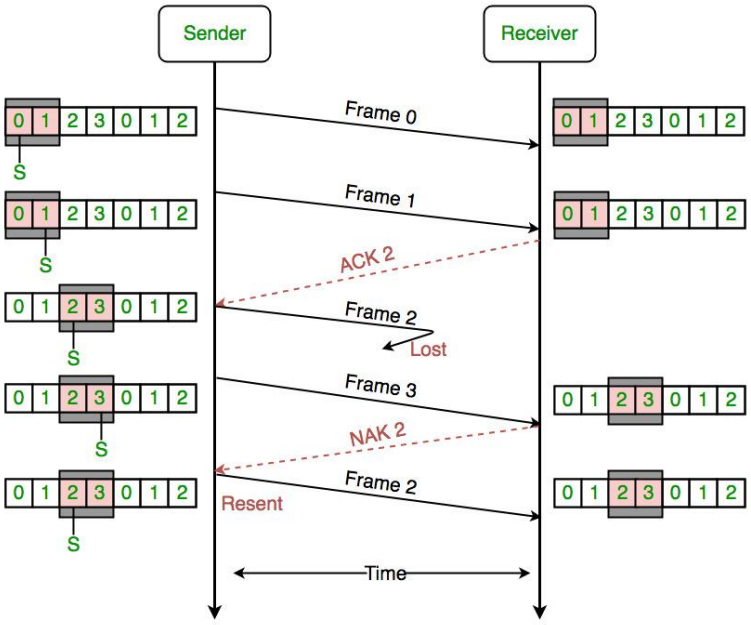
Lost Frames in Go-Back-N



Lost ACK in Go-Back-N

A Protocol Using Selective Repeat

The selective-repetitive ARQ scheme retransmits only those for which NAKs are received or for which timer has expired, this is shown in the Fig. This is the most efficient among the ARQ schemes, but the sender must be more complex so that it can send out-of-order frames. The receiver also must have storage space to store the post-NAK frames and processing power to reinsert frames in proper sequence.


Sender
Receiver
Frame 0
Frame 1
ACK 2
Frame 2
Lost
Frame 3
NAK 2
Frame 2
Resent
Time
S