Unit-2

Data Link Layer:

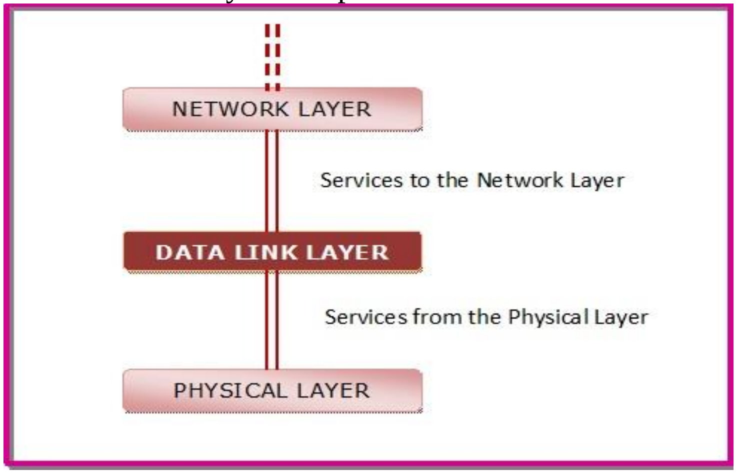
The data link layer in the OSI (Open System Interconnections) Model is in between the physical layer and the network layer. This layer converts the raw transmission facility provided by the physical layer to reliable and error-free frames.

Data Link layer Design Issues:

- Services to the network layer
- Framing
- Error Control
- Flow Control

Services to the Network Layer

In the OSI Model, each layer uses the services of the layer below it and provides services to the layer above it. The data link layer uses the services offered by the physical layer. The primary function of this layer is to provide a well defined service interface to network layer above it.


NETWORK LAYER
Services to the Network Layer
DATA LINK LAYER
Services from the Physical Layer
PHYSICAL LAYER

The types of services provided can be of three types −

- Unacknowledged connectionless service
- Acknowledged connectionless service
- Acknowledged connection - oriented service

1. Unacknowledged Connectionless Service

- In this type of service source machine sends frames to destination machine but the destination machine does not send any acknowledgement of these frames back to the source. Hence it is called unacknowledged service.
- There is no connection establishment between source and destination machine before data transfer or release after data transfer. Therefore it is known as connectionless service.
- There is no error control *i.e.* if any frame is lost due to noise on the line; no attempt is made to recover it.
- This type of service is used when error rate is low and it is suitable for real time traffic such as Ethernet (Ethernet is the traditional technology for connecting devices in a wired local area network (LAN) or wide area network (WAN). It enables devices to communicate with each other via a protocol, which is a set of rules or common network language).

2. Acknowledged Connectionless Service

- In this service, neither the connection is established before the data transfer nor is it released after the data transfer between source and destination.
- When the sender sends the data frames to destination, destination machine sends back the acknowledgement of these frames.
- This type of service provides additional reliability because source machine retransmit the frames if it does not receive the acknowledgement of these frames within the specified time.
- This service is useful over unreliable channels, such as wireless systems (wifi).

3. Acknowledged Connection - Oriented Service

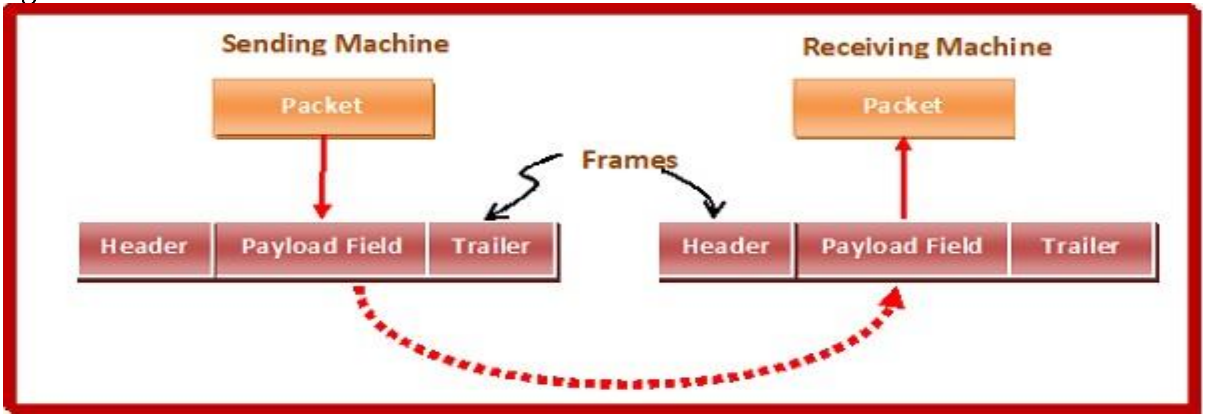
- This service is the most sophisticated service provided by data link layer to network layer.
- It is connection-oriented. It means that connection is establishment between source & destination before any data is transferred.
- In this service, data transfer has three distinct phases:-Connection establishment, Actual data transfer, Connection release.
- Here, each frame being transmitted from source to destination is given a specific number and is acknowledged by the destination machine.
- All the frames are received by destination in the same order in which they are send by the source.

Framing

In the physical layer, data transmission involves synchronized transmission of bits from the source to the destination. The data link layer packs these bits into frames.

Data-link layer takes the packets from the Network Layer and encapsulates them into frames. If the frame size becomes too large, then the packet may be divided into small sized frames. Smaller sized frames make flow control and error control more efficient.

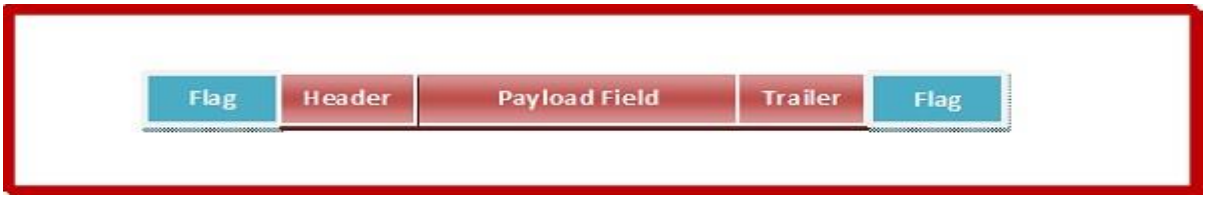
Then, it sends each frame bit-by-bit on the hardware. At receiver’s end, data link layer picks up signals from hardware and assembles them into frames.



Parts of a Frame

A frame has the following parts −

- Frame Header − It contains the source and the destination addresses of the frame.
- Payload field − It contains the message to be delivered.
- Trailer − It contains the error detection and error correction bits.
- Flag − It marks the beginning and end of the frame.



Types of Framing

Framing can be of two types, fixed sized framing and variable sized framing.

Fixed-sized Framing

Here the size of the frame is fixed and so the frame length acts as delimiter of the frame. Consequently, it does not require additional boundary bits to identify the start and end of the frame.

Example − ATM cells (Asynchronous Transfer Mode). Cell is basic data unit of ATM protocol, contains 53 bytes includes 5 bytes header and 48 bytes payload.

Variable – Sized Framing

Here, the size of each frame to be transmitted may be different. So additional mechanisms are kept to mark the end of one frame and the beginning of the next frame.

It is used in local area networks.

Two ways to define frame delimiters in variable sized framing are −

- Length Field − Here, a length field is used that determines the size of the frame. It is used in Ethernet (IEEE 802.3).
- End Delimiter − Here, a pattern is used as a delimiter to determine the size of frame. It is used in Token Rings. If the pattern occurs in the message, then two approaches are used to avoid the situation −
  - Byte Stuffing − A byte is stuffed in the message to differentiate from the delimiter. This is also called character-oriented framing.
  - Bit Stuffing − A pattern of bits of arbitrary length is stuffed in the message to differentiate from the delimiter. This is also called bit – oriented framing.

Error Control

Error control in data link layer is the process of detecting and correcting data frames that have been corrupted or lost during transmission.

In case of lost or corrupted frames, the receiver does not receive the correct data-frame and sender is ignorant about the loss. Data link layer follows a technique to detect transit errors and take necessary actions, which is retransmission of frames whenever error is detected or frame is lost. The process is called Automatic Repeat Request (ARQ).

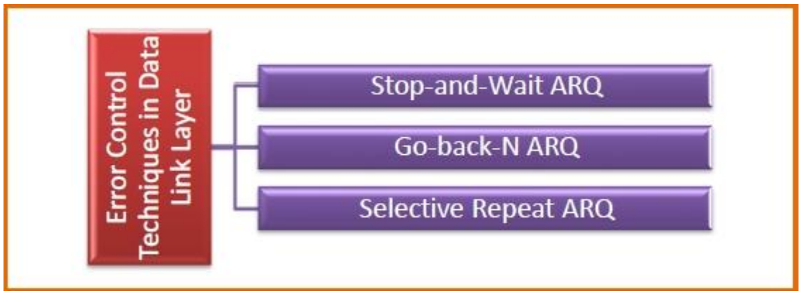
Phases in Error Control

The error control mechanism in data link layer involves the following phases −

- **Detection of Error** − Transmission error, if any, is detected by either the sender or the receiver.
- **Acknowledgment** − acknowledgment may be positive or negative.
  - **Positive ACK** − On receiving a correct frame, the receiver sends a positive acknowledge.
  - **Negative ACK** − On receiving a damaged frame or a duplicate frame, the receiver sends a negative acknowledgment back to the sender.
- **Retransmission** − The sender maintains a clock and sets a timeout period. If an acknowledgment of a data-frame previously transmitted does not arrive before the timeout, or a negative acknowledgment is received, the sender retransmits the frame.

Error Control Techniques

There are three main techniques for error control −



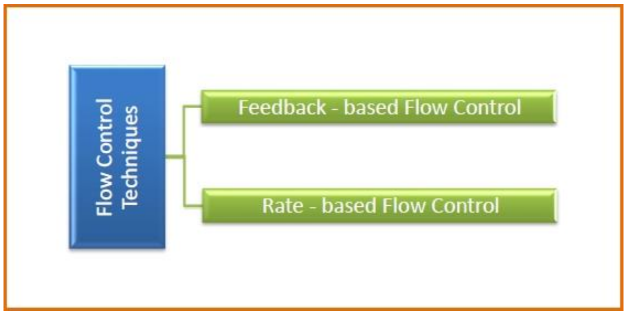
- **Stop and Wait ARQ**
  This protocol involves the following transitions −
  - A timeout counter is maintained by the sender, which is started when a frame is sent.
  - If the sender receives acknowledgment of the sent frame within time, the sender is confirmed about successful delivery of the frame. It then transmits the next frame in queue.
  - If the sender does not receive the acknowledgment within time, the sender assumes that either the frame or its acknowledgment is lost in transit. It then retransmits the frame.
  - If the sender receives a negative acknowledgment, the sender retransmits the frame.
- **Go-Back-N ARQ**
  The working principle of this protocol is −
  - The sender has buffers called sending window.
  - The sender sends multiple frames based upon the sending-window size, without receiving the acknowledgment of the previous ones.
  - The receiver receives frames one by one. It keeps track of incoming frame’s sequence number and sends the corresponding acknowledgment frames.
  - After the sender has sent all the frames in window, it checks up to what sequence number it has received positive acknowledgment.
  - If the sender has received positive acknowledgment for all the frames, it sends next set of frames.
  - If sender receives NACK or has not received any ACK for a particular frame, it retransmits all the frames after which it does not receive any positive ACK.
- **Selective Repeat ARQ**
  - Both the sender and the receiver have buffers called sending window and receiving window respectively.
  - The sender sends multiple frames based upon the sending-window size, without receiving the acknowledgment of the previous ones.
  - The receiver also receives multiple frames within the receiving window size.
  - The receiver keeps track of incoming frame’s sequence numbers, buffers the frames in memory.
  - It sends ACK for all successfully received frames and sends NACK for only frames which are missing or damaged.
  - The sender in this case, sends only packet for which NACK is received.

Flow Control

Flow control is a technique that allows two stations working at different speeds to communicate with each other. It is a set of measures taken to regulate the amount of data that a sender sends so that a fast sender does not overwhelm a slow receiver. In data link layer, flow control restricts the number of frames the sender can send before it waits for an acknowledgment from the receiver.

Approaches of Flow Control

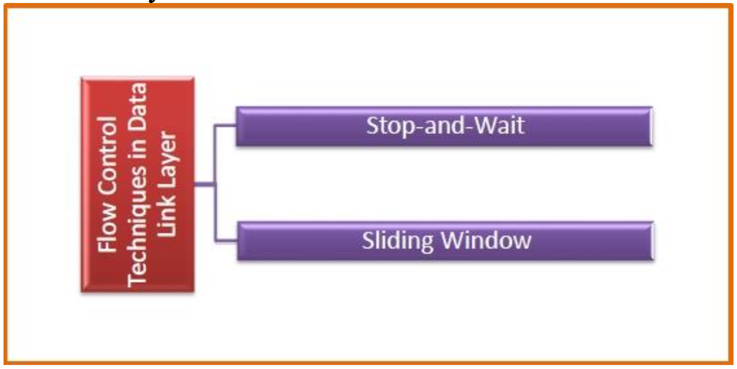
Flow control can be broadly classified into two categories −



- **Feedback based Flow Control** In these protocols, the sender sends frames after it has received acknowledgments from the user. This is used in the data link layer.
- **Rate based Flow Control** These protocols have built in mechanisms to restrict the rate of transmission of data without requiring acknowledgment from the receiver. This is used in the network layer and the transport layer
  .

Flow Control Techniques in Data Link Layer

Data link layer uses feedback based flow control mechanisms. There are two main techniques −



Stop and Wait

This protocol involves the following transitions −

- The sender sends a frame and waits for acknowledgment.
- Once the receiver receives the frame, it sends an acknowledgment frame back to the sender.
- On receiving the acknowledgment frame, the sender understands that the receiver is ready to accept the next frame.

Sliding Window

This protocol improves the efficiency of stop and waits protocol by allowing multiple frames to be transmitted before receiving an acknowledgment.

The working principle of this protocol can be described as follows −

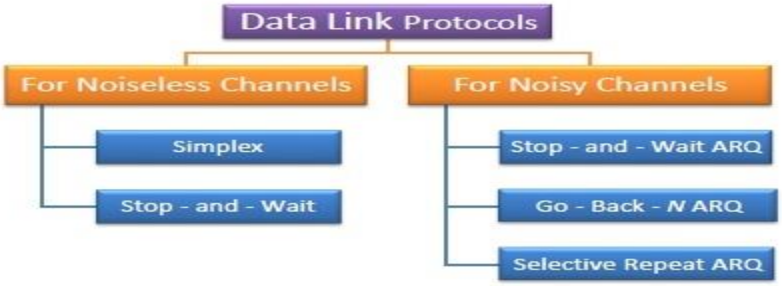
- Both the sender and the receiver have finite sized buffers called windows. The sender and the receiver agree upon the number of frames to be sent based upon the buffer size.
- The sender sends multiple frames in a sequence, without waiting for acknowledgment. When its sending window is filled, it waits for acknowledgment. On receiving acknowledgment, it advances the window and transmits the next frames, according to the number of acknowledgments received.

Error & Flow control mechanisms overview:-

Protocols in the data link layer are designed so that this layer can perform its basic functions: framing, error control and flow control. Framing is the process of dividing bit - streams from physical layer into data frames whose size ranges from a few hundred to a few thousand bytes. Error control mechanisms deals with transmission errors and retransmission of corrupted and lost frames. Flow control regulates speed of delivery and so that a fast sender does not drown a slow receiver.

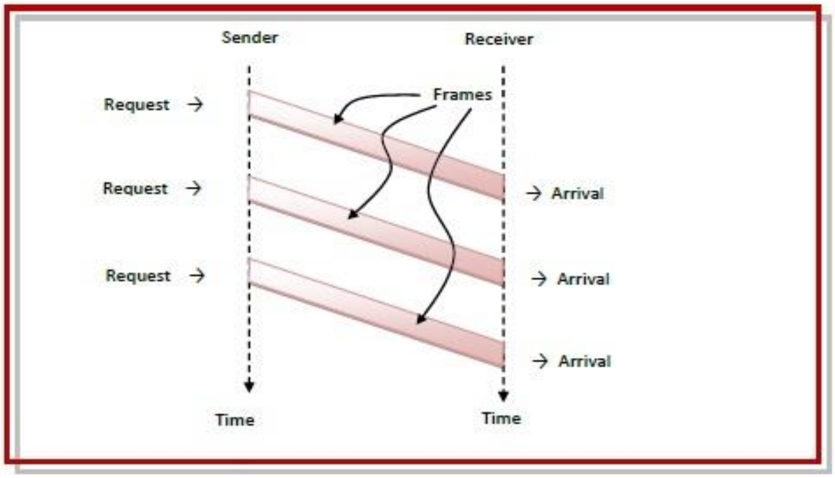
Types of Data Link Protocols:

Data link protocols can be broadly divided into two categories, depending on whether the transmission channel is noiseless (No frames lost here) or noisy (frames will lost here).



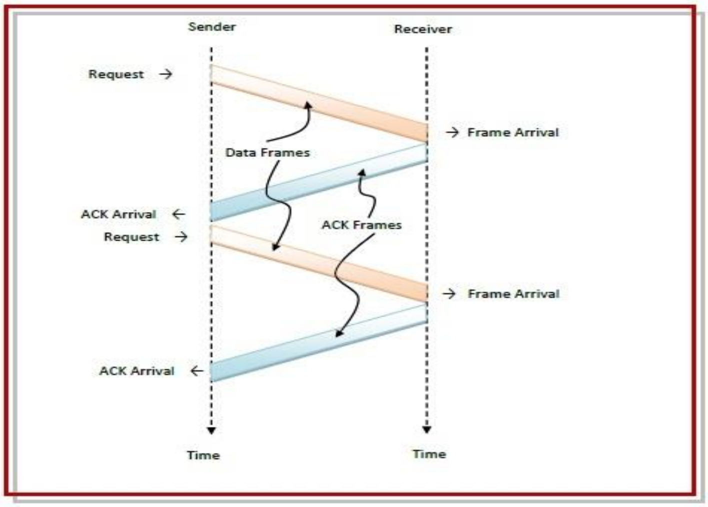
Simplex Protocol

The Simplex protocol is hypothetical protocol designed for unidirectional data transmission over an ideal channel, i.e. a channel through which transmission can never go wrong. It has distinct procedures for sender and receiver. The sender simply sends all its data available onto the channel as soon as they are available in its buffer. The receiver is assumed to process all incoming data instantly. It is hypothetical since it does not handle flow control or error control.



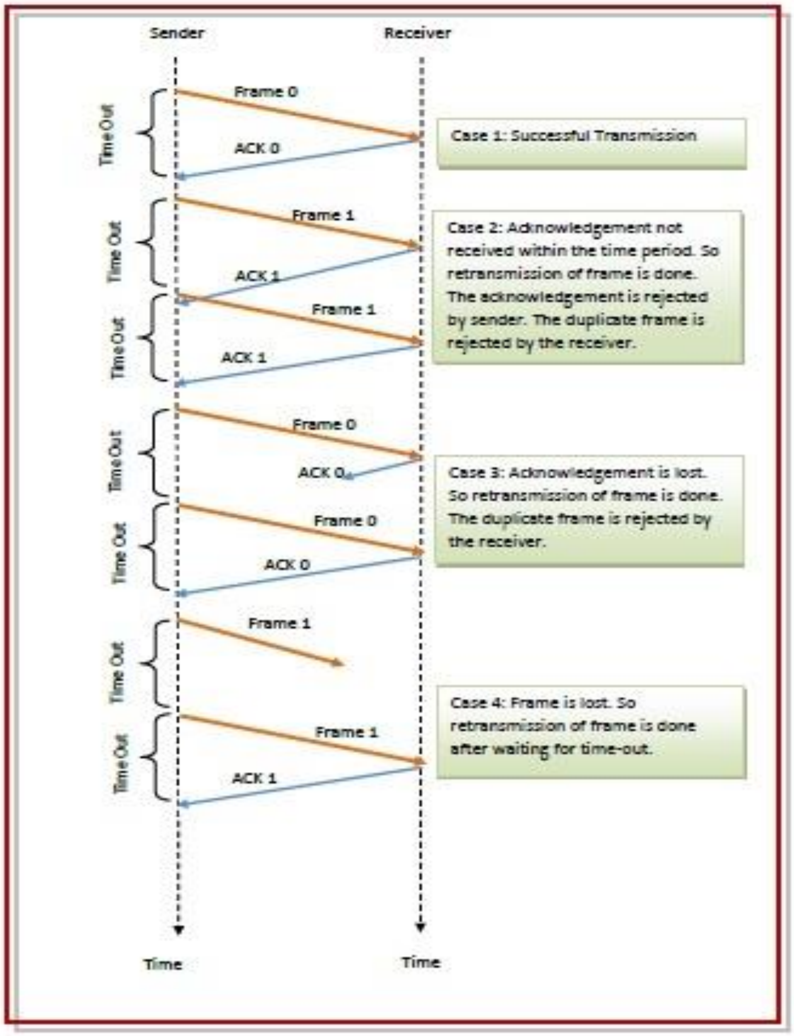
Stop – and – Wait Protocol

Stop – and – Wait protocol is for noiseless channel too. It provides unidirectional data transmission without any error control facilities. However, it provides for flow control so that a fast sender does not drown a slow receiver. The receiver has a finite buffer size with finite processing speed. The sender can send a frame only when it has received indication from the receiver that it is available for further data processing.



Stop – and – Wait ARQ

Stop – and – wait Automatic Repeat Request (Stop – and – Wait ARQ) is a variation of the above protocol with added error control mechanisms, appropriate for noisy channels. The sender keeps a copy of the sent frame. It then waits for a finite time to receive a positive acknowledgement from receiver. If the timer expires or a negative acknowledgement is received, the frame is retransmitted. If a positive acknowledgement is received then the next frame is sent.



Case 1: Successful Transmission

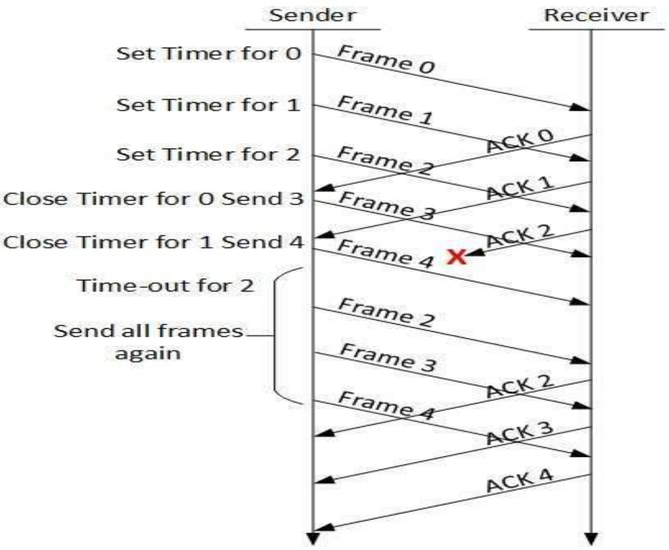
Case 2: Acknowledgement not received within the time period. So retransmission of frame is done. The acknowledgement is rejected by sender. The duplicate frame is rejected by the receiver.

Case 3: Acknowledgement is lost. So retransmission of frame is done. The duplicate frame is rejected by the receiver.

Case 4: Frame is lost. So retransmission of frame is done after waiting for time-out.

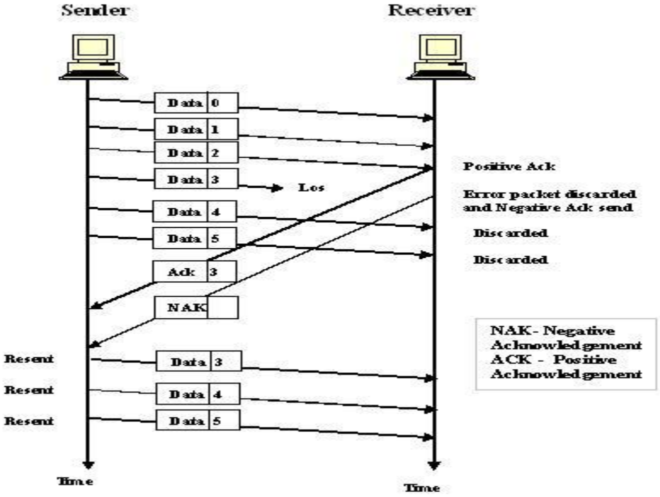
Go – Back – N ARQ

- The sender maintains a timeout counter. When a frame is sent, the sender starts the timeout counter.
- If acknowledgement of frame comes in time, the sender transmits the next frame in queue.
- If acknowledgement does not come in time, the sender assumes that either the frame or its acknowledgement is lost in transit. Sender retransmits the frame and starts the timeout counter.
- If a negative acknowledgement is received, the sender retransmits the frame.
- Sender keeps a copy of sent frame until successful delivery is ensured.
- Receiver responds with an ack when it successfully receives a frame.
- Both data and Ack frames must be numbered



Selective Repeat ARQ

This protocol also provides for sending multiple frames before receiving the acknowledgement for the first frame. However, here only the erroneous or lost frames are retransmitted, while the good frames are received and buffered.



Difference between the Go-Back-N ARQ and Selective Repeat ARQ?

Go-Back-N ARQ	Selective Repeat ARQ
If a frame is corrupted or lost in it,all subsequent frames have to be sent again.	In this, only the frame is sent again, which is corrupted or lost.
If it has a high error rate,it wastes a lot of bandwidth.	There is a loss of low bandwidth.
It is less complex.	It is more complex because it has to do sorting and searching as well. And it also requires more storage.
It does not require sorting.	In this, sorting is done to get the frames in the correct

	order.
It does not require searching.	The search operation is performed in it.
It is used more.	It is used less because it is more complex.

Error Detection & Correction:

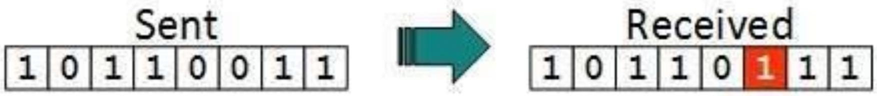
There are many reasons such as noise, cross-talk etc., which may help data to get corrupted during transmission. The upper layers work on some generalized view of network architecture and are not aware of actual hardware data processing. Hence, the upper layers expect error-free transmission between the systems. Most of the applications would not function expectedly if they receive erroneous data. Applications such as voice and video may not be that affected and with some errors they may still function well.

Data-link layer uses some error control mechanism to ensure that frames (data bit streams) are transmitted with certain level of accuracy. But to understand how errors is controlled, it is essential to know what types of errors may occur.

Types of Errors

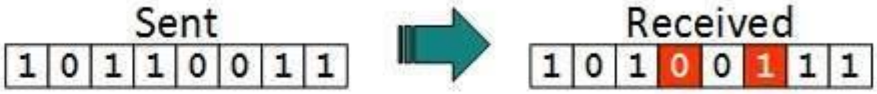
There may be three types of errors:

- **Single bit error**



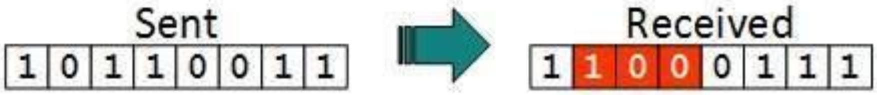
In a frame, there is only one corrupted bit received.

- **Multiple bits error**



Frame is received with more than one bit in corrupted state.

- **Burst error**



Frame contains more than1 consecutive bits corrupted.

Error control mechanism may involve two possible ways:

- Error detection
- Error correction

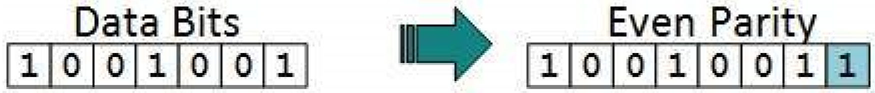
Error Detection

Errors in the received frames are detected by means of Parity Check and Cyclic Redundancy Check (CRC). In both cases, few extra bits are sent along with actual data to confirm that bits received at other end are same as they were sent. If the counter-check at receiver' end fails, the bits are considered corrupted.

Parity Check

One extra bit is sent along with the original bits to make number of 1s either even in case of even parity, or odd in case of odd parity.

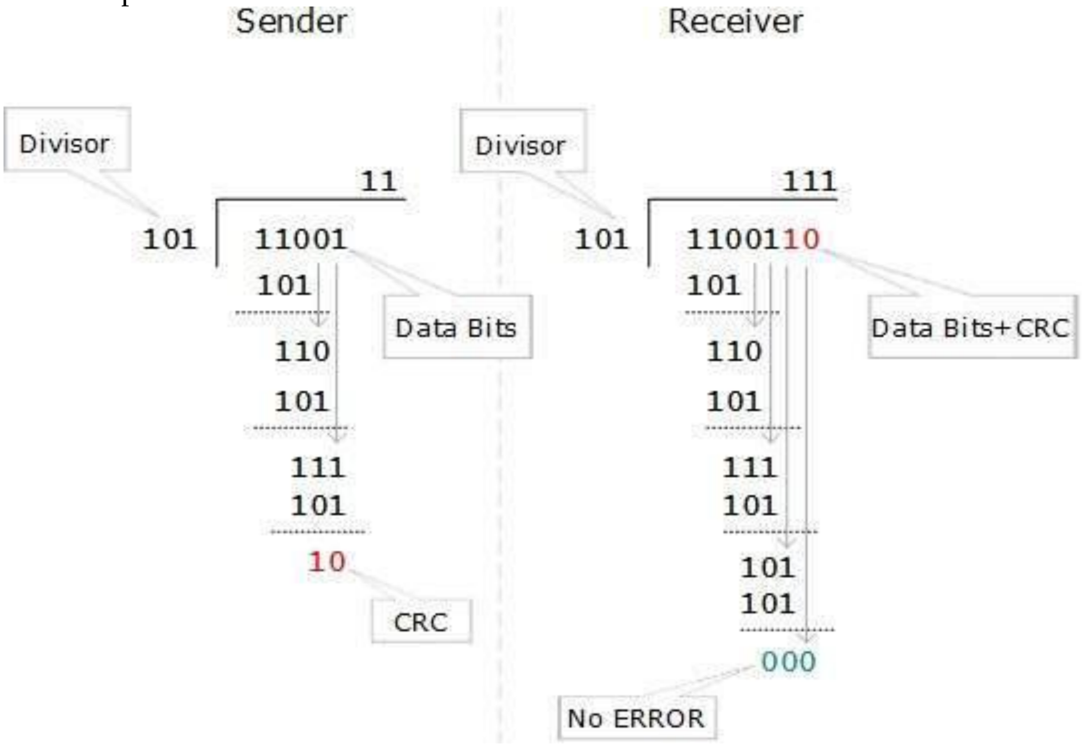
The sender while creating a frame counts the number of 1s in it. For example, where a parity of 1 is added to the block if it contains odd number of 1's, and 0 is added if it contains even number of 1's, This scheme makes the total number of 1's even, that is why it is called even parity checking.



The receiver simply counts the number of 1s in a frame. If the count of 1s is even and even parity is used, the frame is considered to be not-corrupted and is accepted. If the count of 1s is odd and odd parity is used, the frame is still not corrupted.

Cyclic Redundancy Check (CRC)

CRC is a different approach to detect if the received frame contains valid data. This technique involves binary division of the data bits being sent. The divisor is generated using polynomials. The sender performs a division operation on the bits being sent and calculates the remainder. Before sending the actual bits, the sender adds the remainder at the end of the actual bits. Actual data bits plus the remainder is called a codeword. The sender transmits data bits as codeword.



At the other end, the receiver performs division operation on codeword using the same CRC divisor. If the remainder contains all zeros the data bits are accepted, otherwise it is considered as there some data corruption occurred in transit.

Error Correction

In the digital world, error correction can be done in two ways:

- **Backward Error Correction** When the receiver detects an error in the data received, it requests back the sender to retransmit the data unit.
- **Forward Error Correction** When the receiver detects some error in the data received, it executes error-correcting code, which helps it to auto-recover and to correct some kinds of errors.

The first one, Backward Error Correction, is simple and can only be efficiently used where retransmitting is not expensive. For example, fiber optics. But in case of wireless transmission retransmitting may cost too much. In the latter case, Forward Error Correction is used.

To correct the error in data frame, the receiver must know exactly which bit in the frame is corrupted. To locate the bit in error, redundant bits are used as parity bits for error detection. For example, we take ASCII words (7 bits data), then there could be 8 kind of information we need: first seven bits to tell us which bit is error and one more bit to tell that there is no error.

For m data bits, r redundant bits are used. r bits can provide 2r combinations of information. In m+r bit codeword, there is possibility that the r bits themselves may get corrupted. So the number of r bits used must inform about m+r bit locations plus no-error information, i.e. m+r+1.

$$2^r >= m+r+1$$

Multiple access protocol- ALOHA, CSMA, CSMA/CA and CSMA/CD

Data Link Layer

The data link layer is used in a computer network to transmit the data between two devices or nodes. It divides the layer into parts such as **data link control** and the **multiple access resolution/protocol**. The upper layer has the responsibility to flow control and the error control in the data link layer, and hence it is termed as **logical of data link control**. Whereas the lower sub-layer is used to handle and reduce the collision or multiple access on a channel. Hence it is termed as **media access control**

Data Link Control

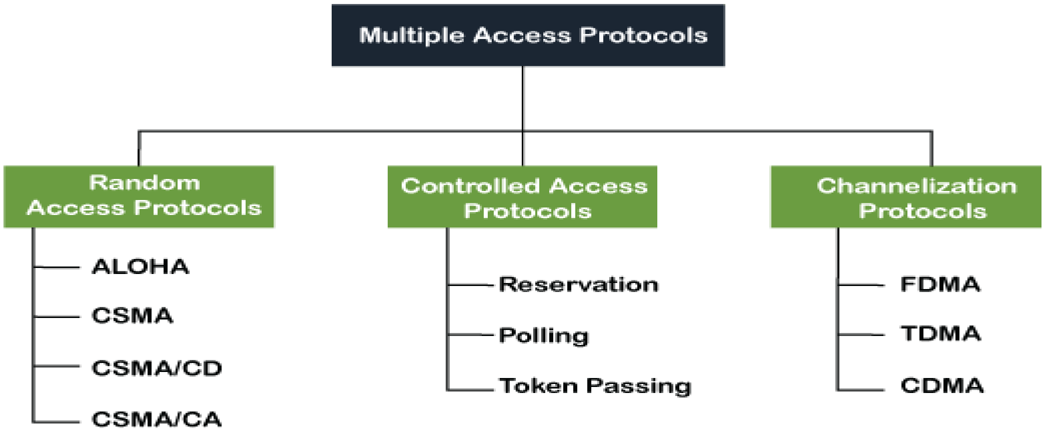
A data link control is a reliable channel for transmitting data over a dedicated link using various techniques such as framing, error control and flow control of data packets in the computer network.

What is a multiple access protocol?

When a sender and receiver have a dedicated link to transmit data packets, the data link control is enough to handle the channel. Suppose there is no dedicated path to communicate or transfer the data between two devices. In that case, multiple stations access the channel and simultaneously transmit the data over the channel. It may create collision and cross talk. Hence, the multiple access protocol is required to reduce the collision and avoid crosstalk between the channels.

For example, suppose that there is a classroom full of students. When a teacher asks a question, all the students (small channels) in the class start answering the question at the same time (transferring the data simultaneously). All the students respond at the same time due to which data is overlap or data lost. Therefore it is the responsibility of a teacher (multiple access protocol) to manage the students and make them one answer.

Following are the types of multiple access protocol that is subdivided into the different process as:



A. Random Access Protocol :

In this protocol, all the station has the equal priority to send the data over a channel. In random access protocol, one or more stations cannot depend on another station nor any station control another station. Depending on the channel's state (idle or busy), each station transmits the data frame. However, if more than one station sends the data over a channel, there may be a collision or data conflict. Due to the collision, the data frame packets may be lost or changed. And hence, it does not receive by the receiver end.

Following are the different methods of **random-access protocols** for broadcasting frames on the channel.

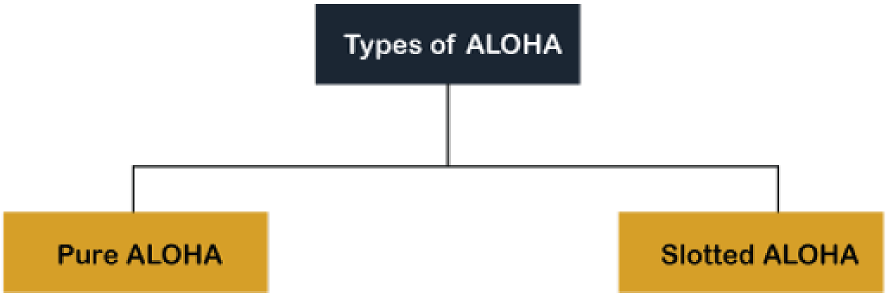
- ALOHA
- CSMA
- CSMA/CD
- CSMA/CA

ALOHA:

It is designed for wireless LAN (Local Area Network) but can also be used in a shared medium to transmit data. Using this method, any station can transmit data across a network simultaneously when a data frameset is available for transmission.

Aloha Rules

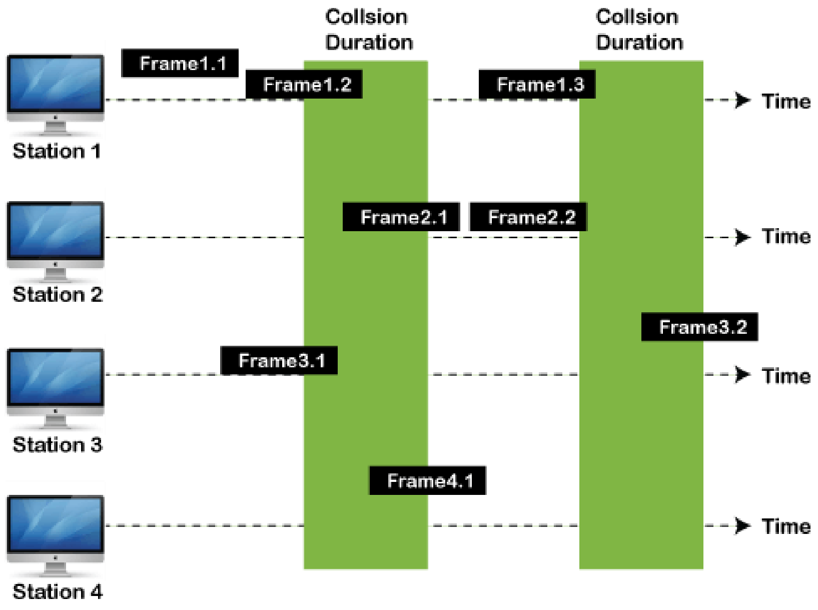
1. Any station can transmit data to a channel at any time.
2. It does not require any carrier sensing.
3. Collision and data frames may be lost during the transmission of data through multiple stations.
4. Acknowledgment of the frames exists in Aloha. Hence, there is no collision detection.
5. It requires retransmission of data after some random amount of time.



Pure Aloha

Whenever data is available for sending over a channel at stations, we use Pure Aloha. In pure Aloha, when each station transmits data to a channel without checking whether the channel is idle or not, the chances of collision may occur, and the data frame can be lost. When any station transmits the data frame to a channel, the pure Aloha waits for the receiver's acknowledgment. If it does not acknowledge the receiver end within the specified time, the station waits for a random amount of time, called the backoff time (Tb). And the station may assume the frame has been lost or destroyed. Therefore, it retransmits the frame until all the data are successfully transmitted to the receiver.

1. The total vulnerable time of pure Aloha is 2 * Tfr.
2. Maximum throughput occurs when G = 1/ 2 that is 18.4%.
3. Successful transmission of data frame is S = G * e ^ - 2 G.



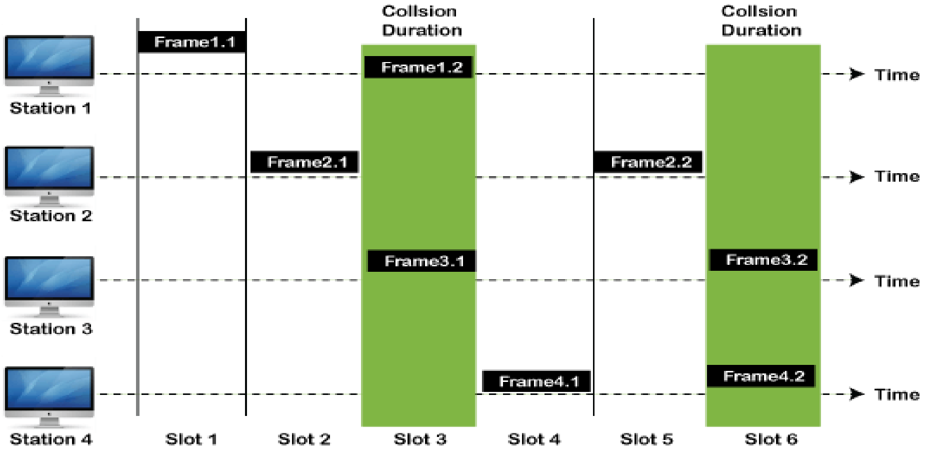
Frames in Pure ALOHA

As we can see in the figure above, there are four stations for accessing a shared channel and transmitting data frames. Some frames collide because most stations send their frames at the same time. Only two frames, frame 1.1 and frame 2.2, are successfully transmitted to the receiver end. At the same time, other frames are lost or destroyed. Whenever two frames fall on a shared channel simultaneously, collisions can occur, and both will suffer damage. If the new frame's first bit enters the channel before finishing the last bit of the second frame. Both frames are completely finished, and both stations must retransmit the data frame.

Slotted Aloha

The slotted Aloha is designed to overcome the pure Aloha's efficiency because pure Aloha has a very high possibility of frame hitting. In slotted Aloha, the shared channel is divided into a fixed time interval called **slots**. So that, if a station wants to send a frame to a shared channel, the frame can only be sent at the beginning of the slot, and only one frame is allowed to be sent to each slot. And if the stations are unable to send data to the beginning of the slot, the station will have to wait until the beginning of the slot for the next time. However, the possibility of a collision remains when trying to send a frame at the beginning of two or more station time slot.

1. Maximum throughput occurs in the slotted Aloha when G = 1 that is 37%.
2. The probability of successfully transmitting the data frame in the slotted Aloha is S = G * e ^ - 2 G.
3. The total vulnerable time required in slotted Aloha is Tfr.



Frames in Slotted ALOHA

CSMA (Carrier Sense Multiple Access)

It is a **carrier sense multiple access** based on media access protocol to sense the traffic on a channel (idle or busy) before transmitting the data. It means that if the channel is idle, the station can send data to the channel. Otherwise, it must wait until the channel becomes idle. Hence, it reduces the chances of a collision on a transmission medium.

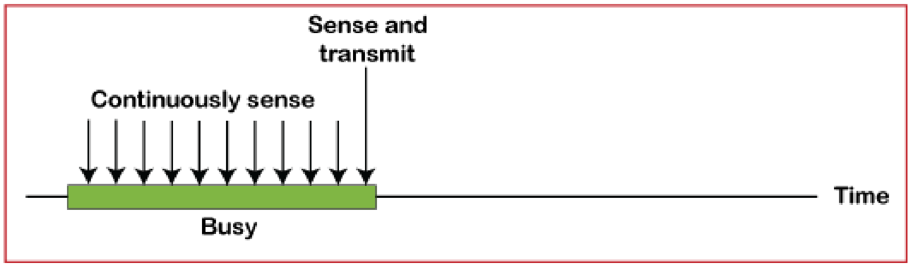
CSMA Access Modes

1-Persistent: In the 1-Persistent mode of CSMA that defines each node, first sense the shared channel and if the channel is idle, it immediately sends the data. Else it must wait and keep track of the status of the channel to be idle and broadcast the frame unconditionally as soon as the channel is idle.

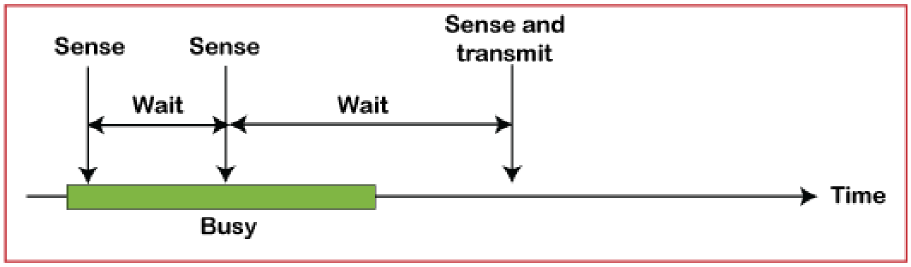
Non-Persistent: It is the access mode of CSMA that defines before transmitting the data, each node must sense the channel, and if the channel is inactive, it immediately sends the data. Otherwise, the station must wait for a random time (not continuously), and when the channel is found to be idle, it transmits the frames.

P-Persistent: It is the combination of 1-Persistent and Non-persistent modes. The P-Persistent mode defines that each node senses the channel, and if the channel is inactive, it sends a frame with a **P** probability. If the data is not transmitted, it waits for a (**q = 1-p probability**) random time and resumes the frame with the next time slot.

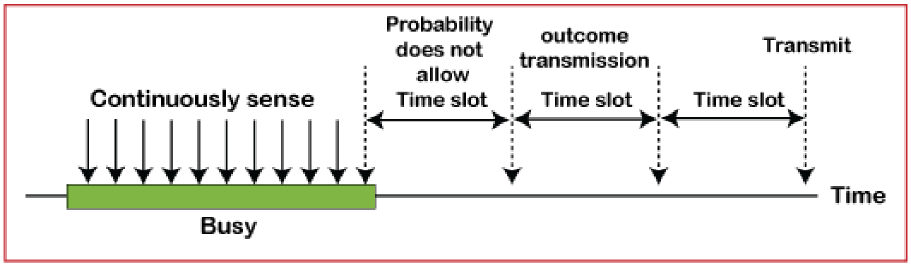
O- Persistent: It is an O-persistent method that defines the superiority of the station before the transmission of the frame on the shared channel. If it is found that the channel is inactive, each station waits for its turn to retransmit the data.



a. 1-persistent



b. Nonpersistent



c. p-persistent

CSMA/ CD

It is a **carrier sense multiple access/ collision detection** network protocol to transmit data frames. The CSMA/CD protocol works with a medium access control layer. Therefore, it first senses the shared channel before broadcasting the frames, and if the channel is idle, it transmits a frame to check whether the transmission was successful. If the frame is successfully received, the station sends another frame. If any collision is detected in the CSMA/CD, the station sends a jam/ stop signal to the shared channel to terminate data transmission. After that, it waits for a random time before sending a frame to a channel.

CSMA/ CA

It is a **carrier sense multiple access/collision avoidance** network protocol for carrier transmission of data frames. It is a protocol that works with a medium access control layer. When a data frame is sent to a channel, it receives an acknowledgment to check whether the channel is clear. If the station receives only a single (own) acknowledgments, that means the data frame has been successfully transmitted to the receiver. But if it gets two signals (its own and one more in which the collision of frames), a collision of the frame occurs in the shared channel. Detects the collision of the frame when a sender receives an acknowledgment signal.

Following are the methods used in the CSMA/CA to avoid the collision:

Interframe space: In this method, the station waits for the channel to become idle, and if it gets the channel is idle, it does not immediately send the data. Instead of this, it waits for some time, and this time period is called the **Interframe** space or IFS. However, the IFS time is often used to define the priority of the station.

Contention window: In the Contention window, the total time is divided into different slots. When the station/ sender is ready to transmit the data frame, it chooses a random slot number of slots as **wait time**. If the channel is still busy, it does not restart the entire process, except that it restarts the timer only to send data packets when the channel is inactive.

Acknowledgment: In the acknowledgment method, the sender station sends the data frame to the shared channel if the acknowledgment is not received ahead of time.

B. <u>Controlled Access Protocol :</u>

It is a method of reducing data frame collision on a shared channel. In the controlled access method, each station interacts and decides to send a data frame by a particular station approved by all other stations. It means that a single station cannot send the data frames unless all other stations are not approved. It has three types of controlled access: **Reservation, Polling**, and **Token Passing**.

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.

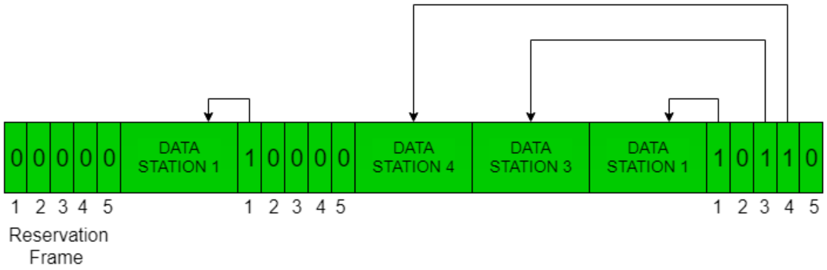
The three controlled-access methods are:

1. Reservation
2. Polling
3. Token Passing

Reservation

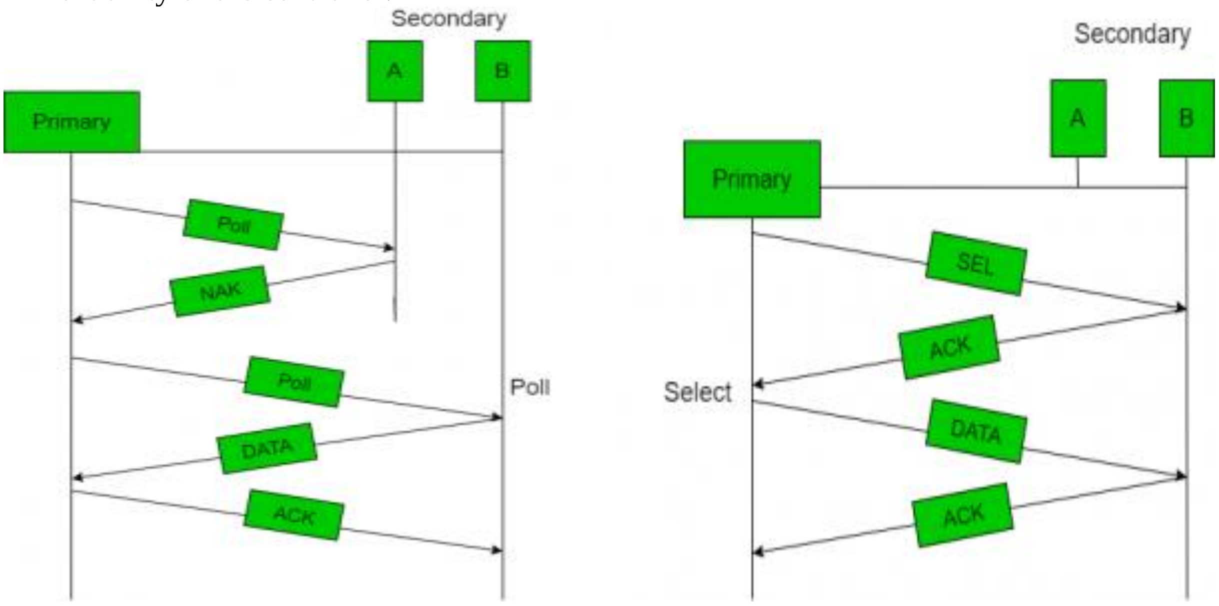
- In the reservation method, a station needs to make a reservation before sending data.
- The time line has two kinds of periods:
    1. Reservation interval of fixed time length
    2. Data transmission period of variable frames.
- If there are M stations, the reservation interval is divided into M 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.



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

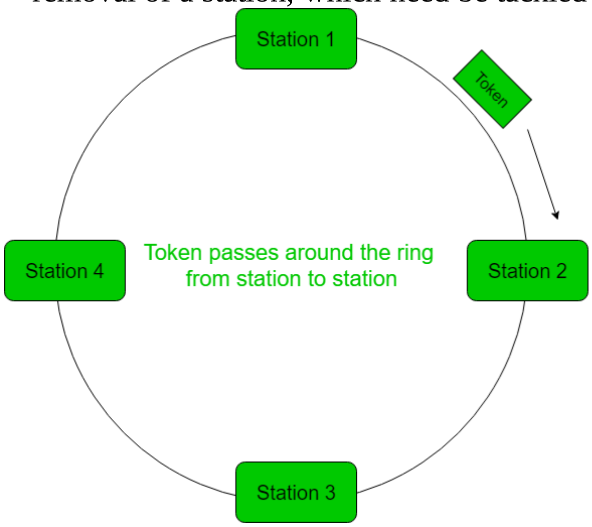
Let T_{poll} be the time for polling and T_t be the time required for transmission of data. Then,

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 exists 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.



1

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)

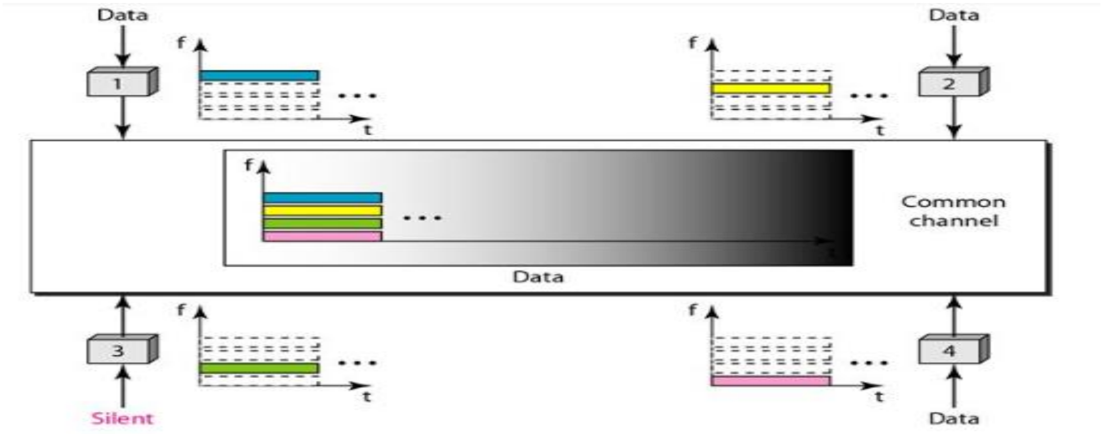
Channelization:

It is a multiple-access method in which the available bandwidth of a link is shared in time, frequency, or through code, between different stations. The three channelization protocols are FDMA, TDMA, and CDMA.

Frequency-Division Multiple Accesses (FDMA):

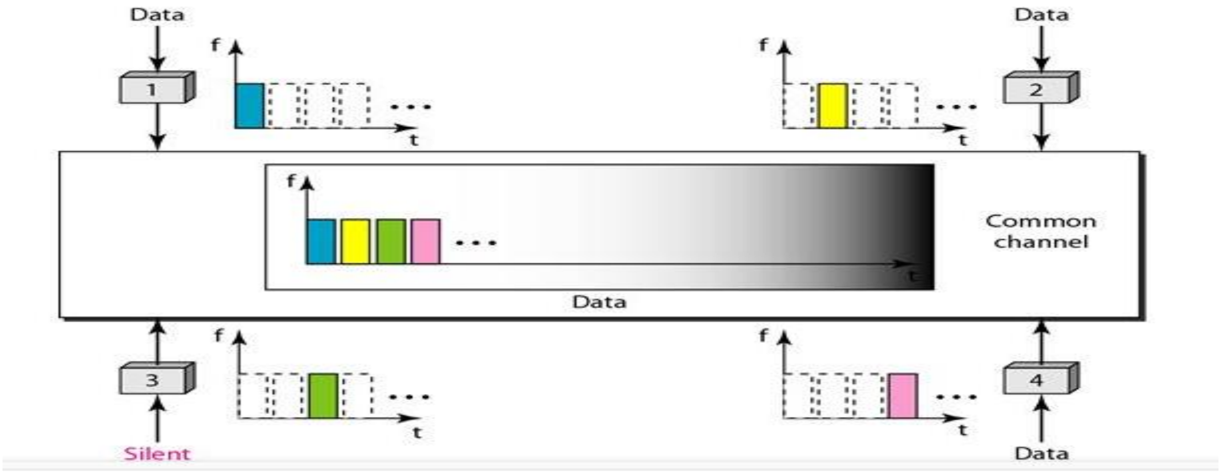
In frequency-division multiple access (FDMA), the available bandwidth is divided into frequency bands. Each station is allocated a band to send its data. In other words, each band is reserved for a specific station, and it belongs to the station all the time. Each station also uses a band pass filter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

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



Time-Division Multiple Accesses (TDMA):

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



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

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

Code-Division Multiple Accesses (CDMA):

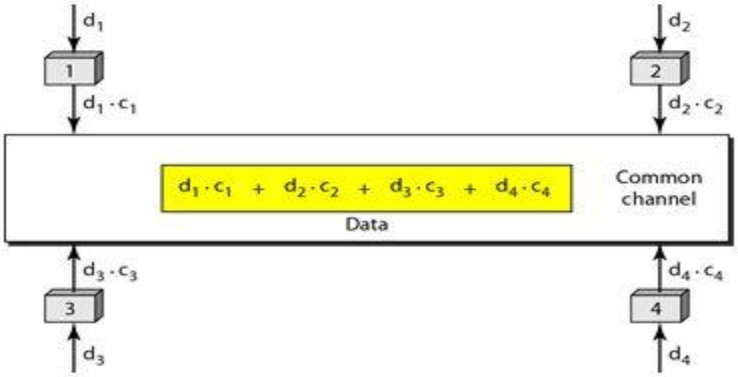
CDMA simply means communication with different codes. CDMA differs from FDMA because only one channel occupies the entire bandwidth of the link. It differs from TDMA because all stations can send data simultaneously; there is no timesharing.

Implementation:

Let us assume we have four stations 1, 2, 3, and 4 connected to the same channel. The data from station 1 are d1 , from station 2 are d2, and so on. The code assigned to the first station is c1, to the second is c2, and so on. We assume that the assigned codes have two properties.

1. If we multiply each code by another, we get 0.

2. If we multiply each code by itself, we get 4 (the number of stations).
With these two properties in mind, how the above four stations can send data using the same common channel, as shown in the following figure.



Station 1 multiplies (a special kind of multiplication, as we will see) its data by its code to get d1.c1. Station 2 multiplies its data by its code to get d2.c2. And so on. The data that go on the channel are the sum of all these terms, as shown in the box.

Any station that wants to receive data from one of the other three multiplies the data on the channel by the code of the sender. For example, suppose stations 1 and 2 are talking to each other. Station 2 wants to hear what station 1 is saying. It multiplies the data on the channel by c1 the code of station1.
Because (c1.c1) is 4, but (c2 . c1), (c3. c1), and (c4 .c1) are all 0s, station 2 divides the result by 4 to get the data from station1.

data =(d1.c1+d2.c2+d3.c3+d4.c4).c1

= c1. d1. c1+ c1. d2. c2+ c1. d3. c3+ c1. d4. c4= 4d1

Wired LANs:

A local area network (LAN) is a computer network that is designed for a limited geographic area such as a building or a campus. Although a LAN can be used as an isolated network to connect computers in an organization for the sole purpose of sharing resources, most LANs today are also linked to a wide area network (WAN) or the Internet. The LAN market has seen several technologies such as Ethernet, Token Ring, Token Bus and ATM LAN. Some of these technologies survived for a while, but Ethernet is by far the dominant technology.

1. IEEE STANDARDS

In **1985**, the Computer Society of the **IEEE** started a project, called **Project 802**, to set standards to enable intercommunication among equipment from a variety of manufacturers. Project 802 does not seek to replace any part of the OSI or the Internet model. Instead, it is a way of specifying functions of the physical layer and the data link layer of major LAN protocols.

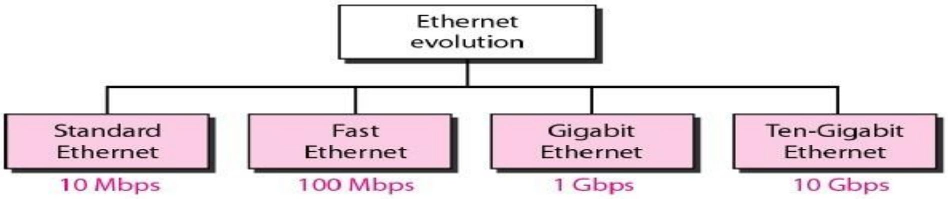


Figure 13.1 Ethernet evolutions through four generations

Standard Ethernet (IEEE 802.3):

The original Ethernet was created in 1976 at Xerox's Palo Alto Research Center (PARC). Since then, it has gone through four generations:

- **a.** Standard Ethernet (10 Mbps),
- **b.** Fast Ethernet (100 Mbps),
- **c.** Gigabit Ethernet (1 Gbps), and
- **d.** Ten-Gigabit Ethernet (10 Gbps), as shown in Figure 13.1.

Standard Ethernet also known as IEEE 802.3 was the LAN standard proposed by IEEE. Data rate for standard Ethernet is 10 Mbps.

MAC Sub layer

In Standard Ethernet, the MAC sub layer governs the operation of the access method. It also frames data received from the upper layer and passes them to the physical layer.

- **Frame Format**

The Ethernet frame contains seven fields: preamble, SFD, DA, SA, length or type of protocol data unit (PDU), upper-layer data, and the CRC. Ethernet does not provide any mechanism for acknowledging received frames, making it what is known as an unreliable medium. Acknowledgments must be implemented at the higher layers. The format of the MAC frame is shown in Figure 13.2.

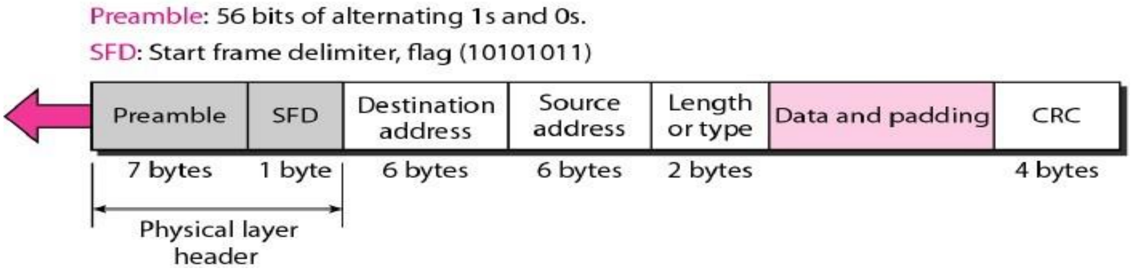


Figure 13.2 802.3 MAC frame

i. **Preamble.** The first field of the 802.3 frame contains 7 bytes (56 bits) of alternating 0s and 1s that alerts the receiving system to the coming frame and enables it to synchronize its input timing. The pattern provides only an alert and a timing pulse. The 56-bit pattern allows the stations to miss some bits at the beginning of the frame. The preamble is actually added at the physical layer and is not (formally) part of the frame.

ii. **Start frame delimiter (SFD).** The second field (1 byte: 10101011) signals the beginning of the frame. The SFD warns the station or stations that this is the last chance for synchronization. The last 2 bits is 11 and alerts the receiver that the next field is the destination address.

iii. **Destination address (DA).** The DA field is 6 bytes and contains the physical address of the destination station or stations to receive the packet.

iv. **Source address (SA).** The SA field is also 6 bytes and contains the physical address of the sender of the packet.

v. **Length or type.** This field is defined as a type field or length field. The original Ethernet used this field as the type field to define the upper-layer protocol using the MAC frame. The IEEE standard used it as the length field to define the number of bytes in the data field. Both uses are common today.

vi. **Data.** This field carries data encapsulated from the upper-layer protocols. It is a minimum of 46 and a maximum of 1500 bytes.

vii. **CRC.** The last field contains error detection information, in this case a CRC-32.

viii. **Frame Length** Ethernet has imposed restrictions on both the minimum and maximum lengths of a frame, as shown in Figure 13.3.

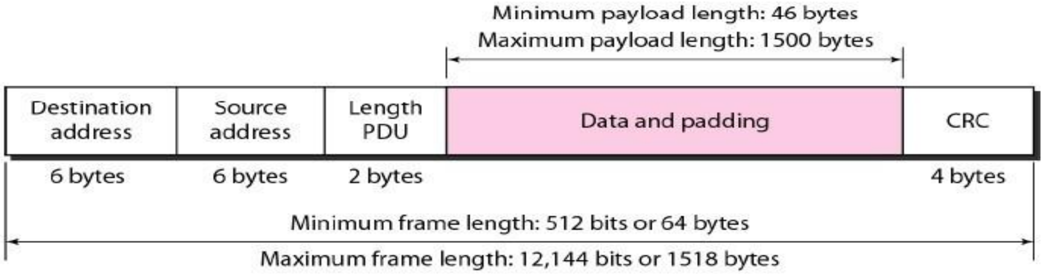

Minimum payload length: 46 bytes
Maximum payload length: 1500 bytes
Destination address
Source address
Length PDU
Data and padding
CRC
6 bytes
6 bytes
2 bytes
4 bytes
Minimum frame length: 512 bits or 64 bytes
Maximum frame length: 12,144 bits or 1518 bytes

Figure 13.3 Minimum and maximum lengths

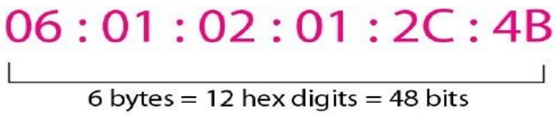
The minimum length restriction is required for the correct operation of *CSMA/CD* as we will see shortly. An Ethernet frame needs to have a minimum length of 512 bits or 64 bytes. Part of this length is the header and the trailer. If we count 18 bytes of header and trailer (6 bytes of source address, 6 bytes of destination address, 2 bytes of length or type, and 4 bytes of CRC), then the minimum length of data from the upper layer is 64 - 18 = 46 bytes. If the upper-layer packet is less than 46 bytes, padding is added to make up the difference.

The standard defines the maximum length of a frame (without preamble and SFD field) as 1518 bytes. If we subtract the 18 bytes of header and trailer, the maximum length of the payload is 1500 bytes. The maximum length restriction has two historical reasons. First, memory was very expensive when Ethernet was designed: a maximum length restriction helped to reduce the size of the buffer. Second, the maximum length restriction prevents one station from monopolizing the shared medium, blocking other stations that have data to send.

MAC Addressing

Each station on an Ethernet network (such as a PC, workstation, or printer) has its own network interface card (NIC). The NIC fits inside the station and provides the station with a 6-byte physical (MAC) address. As shown in Figure 13.4, the Ethernet address is 6 bytes (48 bits), normally written in hexadecimal notation, with a colon between the bytes.

Figure 13.4 Example of an Ethernet address in hexadecimal notation


06 : 01 : 02 : 01 : 2C : 4B
6 bytes = 12 hex digits = 48 bits

Unicast, Multicast, and Broadcast Addresses

Data is transmitted over a network by three simple methods i.e. Unicast, Broadcast, and Multicast Figure 13.5. So let’s begin to summarize the difference between these three**:**

- **Unicast**: from one source to one destination i.e. One-to-One
- **Broadcast**: from one source to all possible destinations i.e. One-to-All
- **Multicast**: from one source to multiple destinations stating an interest in receiving the traffic i.e. One-to-Many

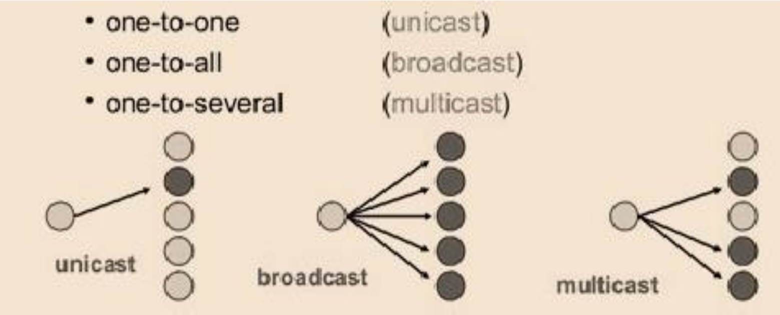
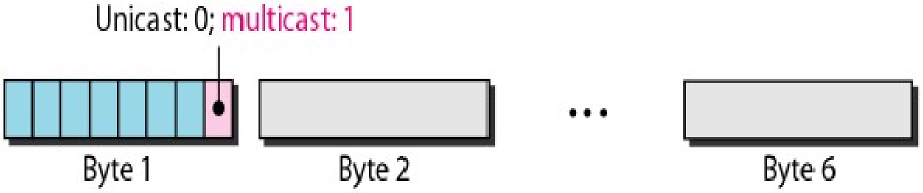


Figure 13.5 Unicasting, Multicasting and Broadcasting

- A source address is always a unicast address as the frame comes from only one station.
- The destination address, however, can be unicast, multicast, or broadcast.
- Figure 13.6 shows how to distinguish a unicast address from a multicast address. If the least significant bit of the first byte in a destination address is 0, the address is unicast; otherwise, it is multicast.

Figure 13.6 Unicast and multicast MAC addresses



A unicast destination address defines only one recipient; the relationship between the sender and the receiver is one-to-one.

A multicast destination Address defines a group of addresses; the Relationship

Between the sender and the receivers is one-to-many.
The broadcast address is a special case of the multicast address; the recipients are all the stations on the LAN. A broadcast destination address is forty-eight 1s.

Example 13.1

Define the type of the following destination addresses:
a. 4A:30:10:21:10:1A
b. 47:20:1B:2E:07:EE
c. FF:FF:FF:FF:FF:FF

Solution

To find the type of the address, we need to look at the second hexadecimal digit from the left. If it is even, the address is unicast. If it is odd, the address is multicast. If all digits are F's, the address is broadcast. Therefore, we have the following:

a. This is a unicast address because 10(A) in binary is 1010 (even).
b. This is a multicast address because 7 in binary is 0111 (odd).
c. This is a broadcast address because all digits are F's.

The way the addresses are sent out on line is different from the way they are written in hexadecimal notation. The transmission is left-to-right, byte by byte; however, for each byte, the least significant bit is sent first and the most significant bit is sent last. This means that the bit that defines an address as unicast or multicast arrives first at the receiver.

Categories of Standard Ethernet

The Standard Ethernet defines several physical layer implementations; four of the most common, are shown in Figure 13.7.

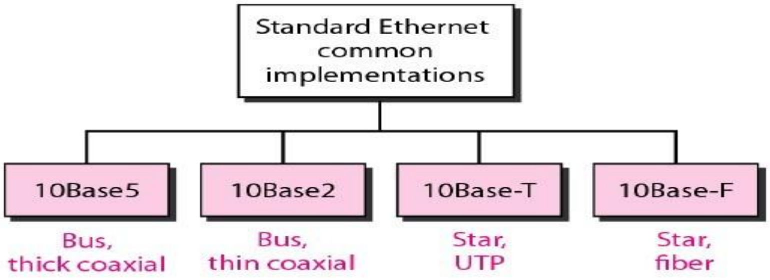


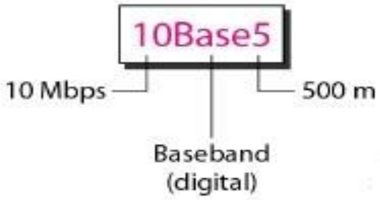
Figure 13.7 Categories of Standard Ethernet

Encoding and Decoding

All standard implementations use digital signaling (baseband) at 10 Mbps. At the sender, data are converted to a digital signal using the Manchester scheme; at the receiver, the received signal is interpreted as Manchester and decoded into data.

- ***10Base5: Thick Ethernet***

The first implementation is called **10Base5, thick Ethernet, or Thick net.** The nickname derives from the size of the cable, which is roughly the size of a garden hose



and too stiff to bend with your hands. 10Base5 was the first Ethernet specification to use a bus topology with an external **transceiver** (transmitter/receiver) connected via a tap to a thick coaxial cable.

The transceiver is responsible for transmitting, receiving, and detecting collisions. The transceiver is connected to the station via a transceiver cable that provides separate paths for sending and receiving. This means that collision can only happen in the coaxial cable. The maximum length of the coaxial cable must not exceed *500* m, otherwise, there is excessive degradation of the signal. If a length of more than *500* m is needed, up to five segments, each a maximum of 500-meter, can be connected using repeaters.

- ***10Base2: Thin Ethernet***

The second implementation is called 10Base2, **thin** Ethernet, or Cheaper net. 10Base2 also uses a bus topology, but the cable is much thinner and more flexible. The cable can be bent to pass very close to the stations. In this case, the transceiver is normally part of the network interface card (NIC), which is installed inside the station.

Note that the collision here occurs in the thin coaxial cable. This implementation is more cost effective than 10Base5 because thin coaxial cable is less expensive than thick coaxial and the tee connections are much cheaper than taps. Installation is simpler because the thin coaxial cable is very flexible. However, the length of each segment cannot exceed *185* m (close to 200 m) due to the high level of attenuation in thin coaxial cable.

- ***10Base-T: Twisted-Pair Ethernet***

The third implementation is called 10Base-T or twisted-pair Ethernet. 10Base-T uses a physical star topology. The stations are connected to a hub via two pairs of twisted cable. Note that two pairs of twisted cable create two paths (one for sending and one for receiving) between the station and the hub. Any collision here happens in the hub. Compared to 10Base5 or 10Base2, we can see that the hub actually replaces the coaxial cable as far as a collision is concerned. The maximum length of the twisted cable here is defined as 100 m, to minimize the effect of attenuation in the twisted cable.

- ***10Base-F: Fiber Ethernet***

Although there are several types of optical fiber 10-Mbps Ethernet, the most common is called 10Base-F. 10Base-F uses a star topology to connect stations to a hub. The stations are connected to the hub using two fiber-optic cables.

Summary

Table 13.1 shows a summary of Standard Ethernet implementations.

Characteristics	***10Base5***	***10Base2***	***10Base-T***	***10Base-F***
Media	Thick Coaxial Cable	Thin Coaxial Cable	2UTP	2Fiber
Maximum length	500m	185m	100m	2000m
Line encoding	Manchester	Manchester	Manchester	Manchester

Changes in the Standard

The 10-Mbps Standard Ethernet has gone through several changes before moving to the higher data rates. These changes actually opened the road to the evolution of the Ethernet to become compatible with other high-data-rate LANs. We discuss some of these changes in this section.

Bridged Ethernet

The first step in the Ethernet evolution was the division of a LAN by bridges. A Bridge is a two port switch used to connect two segments of a LAN. Bridges have two effects on an Ethernet LAN:

- They **raise the bandwidth** and
- They **separate collision domains**.

Raising the Bandwidth

In an unbridged Ethernet network, the total capacity (10 Mbps) is shared among all stations with a frame to send; the stations share the bandwidth of the network. If only one station has frames to send, it benefits from the total capacity (10 Mbps). But if more than one station needs to use the network, the capacity is shared. For example, if two stations have a lot of frames to send, they probably alternate in usage. When one station is sending, the other one refrains from sending. We can say that, in this case, each station on average, sends at a rate of 5 Mbps. Figure 13.8 shows the situation.

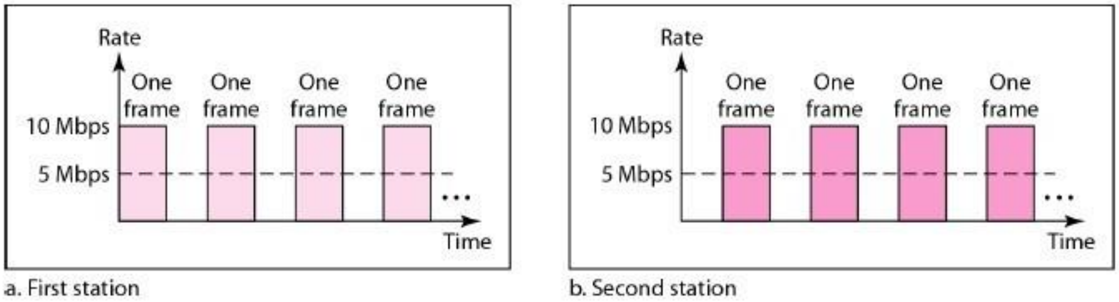


Figure 13.8 Sharing bandwidth

A bridge divides the network into two or more networks. Bandwidth-wise, each network is independent. For example, in Figure 13.9, a network with 12 stations is divided into two networks, each with 6 stations. Now each network has a capacity of 10 Mbps. The 10-Mbps capacity in each segment is now shared between 6 stations (actually 7 because the bridge acts as a station in each segment), not 12 stations. In a network with a heavy load, each station theoretically is offered 10/6 Mbps instead of 10/12 Mbps, assuming that the traffic is not going through the bridge. It is obvious that if we further divide the network, we can gain more bandwidth for each segment. For example, if we use a four-port bridge, each station is now offered 10/3 Mbps, which is 4 times more than an unbridged network.

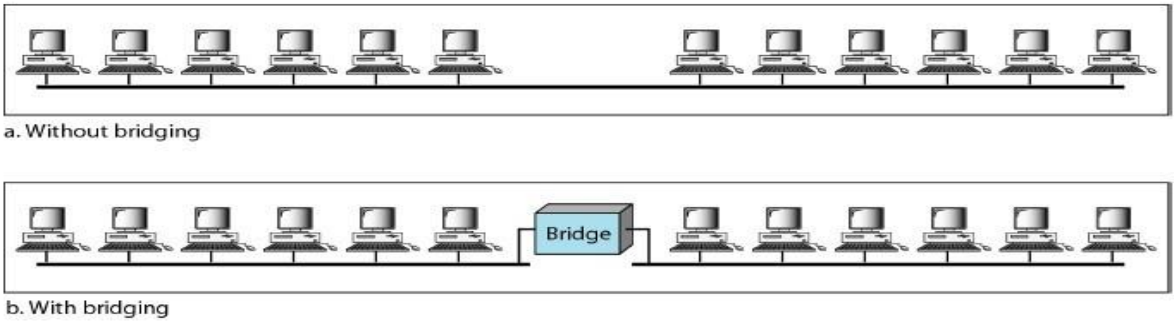


Figure 13.9 A network with and without a bridge

Separating Collision Domains

Another advantage of a bridge is the separation of the collision domain. Figure 13.10 shows the collision domains for an unbridged and a bridged network. You can see that the collision domain becomes much smaller and the probability of collision is reduced tremendously. Without bridging, 12 stations contend for access to the medium; with bridging only 3 stations contend for access to the medium.

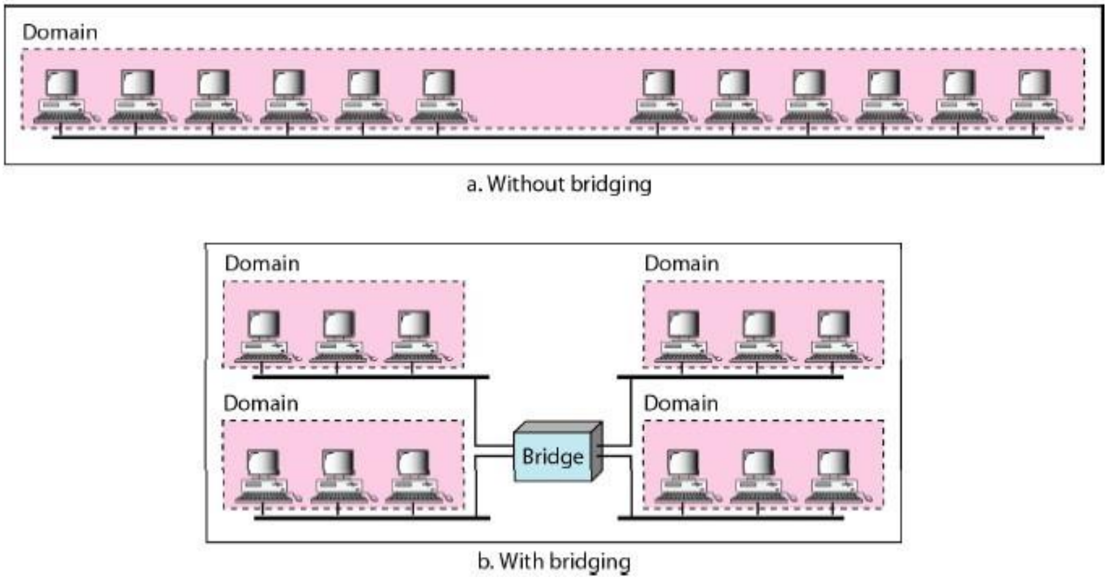


Figure 13.10 Collision domains in an unbridged network and a bridged network

Switched Ethernet

The idea of a bridged LAN can be extended to a switched LAN. Instead of having two to four networks, why not have N networks, where N is the number of stations on the LAN? In other words, if we can have a multiple-port bridge, why not have an N-port switch? In this way, the bandwidth is shared only between the station and the switch (5 Mbps each). In addition, the collision domain is divided into N domains.

A layer 2 switch is an N-port bridge with additional sophistication that allows faster handling of the packets. Evolution from a bridged Ethernet to a switched Ethernet was a big step that opened the way to an even faster Ethernet, as we will see. Figure 13.11 shows a switched LAN.

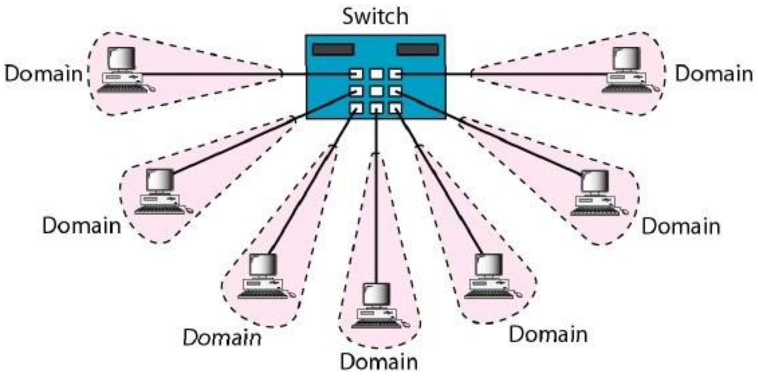


Figure 13.11 Switched Ethernet

Full-Duplex Ethernet

One of the limitations of 10Base5 and 10Base2 is that communication is half-duplex (10Base-T is always full-duplex); a station can either send or receive, but may not do both at the same time. The next step in the evolution was to move from switched Ethernet to full- duplex switched Ethernet. The full-duplex mode increases the capacity of each domain from 10 to 20 Mbps. Figure 13.12 shows a switched Ethernet in full-duplex mode. Note that instead of using one link between the station and the switch, the configuration uses two links: one to transmit and one to receive.

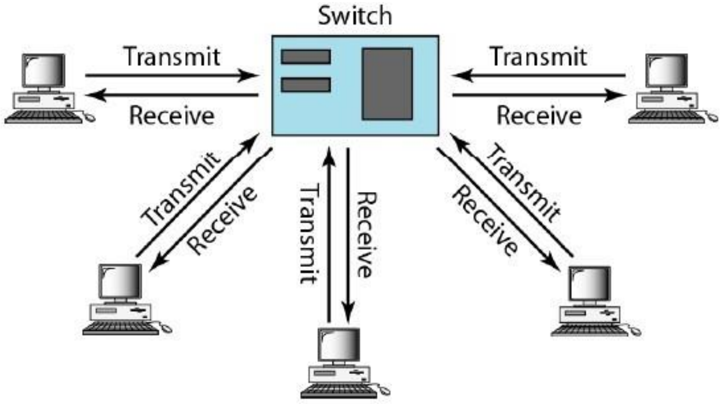


Figure 13.12 Full-duplex switched Ethernet

No Need for CSMA/CD

In full-duplex switched Ethernet, there is no need for the *CSMAICD* method. In a full duplex switched Ethernet, each station is connected to the switch via two separate links. Each station or switch can send and receive independently without worrying about collision. Each link is a point-to-point dedicated path between the station and the switch. There is no longer a need for carrier sensing; there is no longer a need for collision detection. The job of the MAC layer becomes much easier. The carrier sensing and collision detection functionalities of the MAC sub-layer can be turned off.

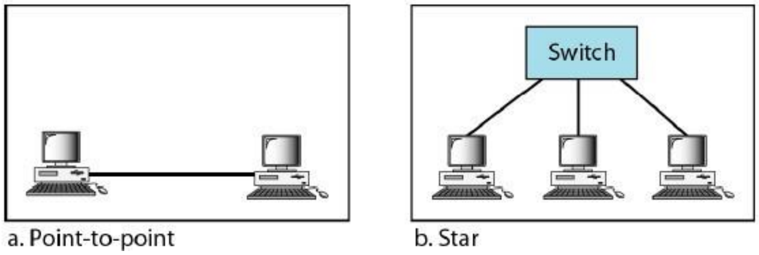
Fast Ethernet(IEEE 802.3u)

Fast Ethernet was designed to compete with LAN protocols such as FDDI or Fiber Channel (or Fibre Channel, as it is sometimes spelled). IEEE created Fast Ethernet under the name 802.3u. Fast Ethernet is backward-compatible with Standard Ethernet, but it can transmit data 10 times faster at a rate of 100 Mbps. The goals of Fast Ethernet can be summarized as follows:

a. Upgrade the data rate to 100 Mbps.
b. Make it compatible with Standard Ethernet.
c. Keep the same 48-bit address.
d. Keep the same frame format.
e. Keep the same minimum and maximum frame lengths.

Topology

Fast Ethernet is designed to connect two or more stations together. If there are only two stations, they can be connected point-to-point. Three or more stations need to be connected in a star topology with a hub or a switch at the center, as shown in Figure 13.13.



a. Point-to-point

b. Star

Figure 13.13 Fast Ethernet topology

Implementation

Fast Ethernet implementation at the physical layer can be categorized as either two-wire or four-wire. The two-wire implementation can be either category 5 UTP (100Base-TX) or fiber- optic cable (100Base-FX). The four-wire implementation is designed only for category 3 UTP (100Base-T4). See Figure 13.14.

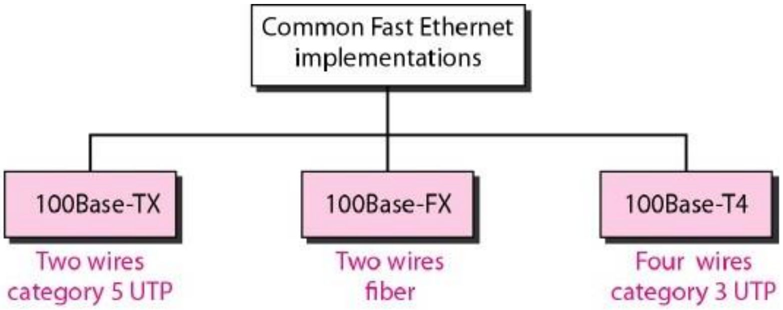


Figure 13.14 Fast Ethernet implementations

Table 13.2 Summary of Fast Ethernet implementations

Characteristics	**100Base-TX**	**100Base-FX**	**100Base-T4**
Media	Cat 5 UTP or STP	Fiber	Cat 4 UTP
Number of wires	2	2	4
Maximum length	100m	100m	100m
Line encoding	MLT-3	NRZ-I	8B/6T

Gigabit Ethernet(IEEE 802.3z)

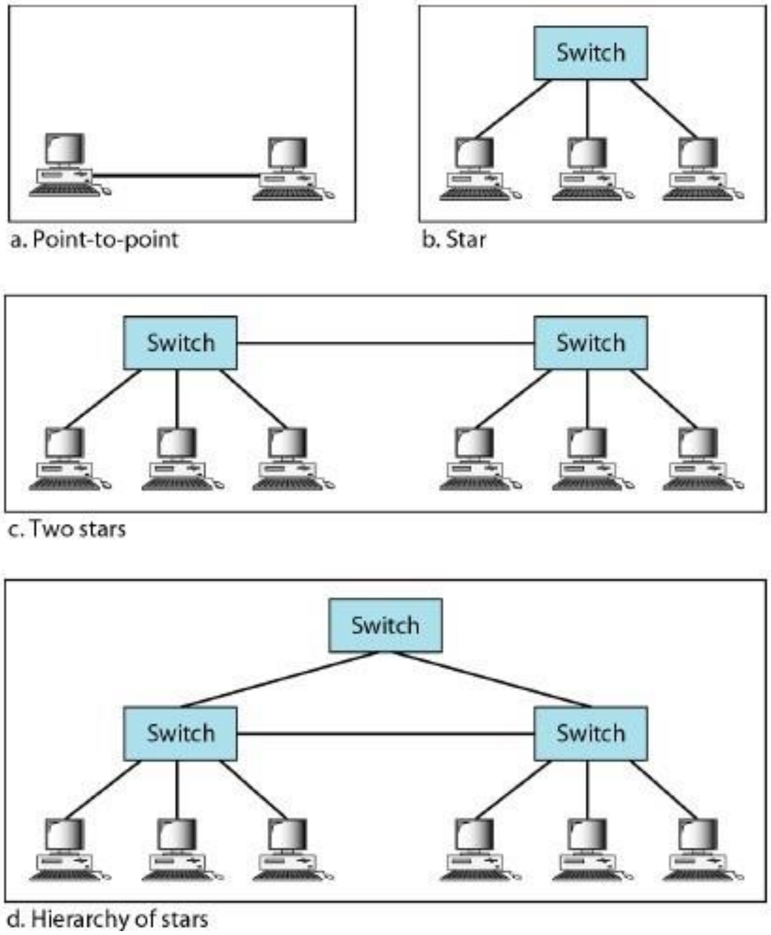
The need for an even higher data rate resulted in the design of the Gigabit Ethernet protocol (1000 Mbps). The IEEE committee calls the Standard 802.3z. The goals of the Gigabit Ethernet design can be summarized as follows:

a. Upgrade the data rate to 1 Gbps.
b. Make it compatible with Standard or Fast Ethernet.
c. Use the same 48-bit address.
d. Use the same frame format.
e. Keep the same minimum and maximum frame lengths.
f. To support auto negotiation as defined in Fast Ethernet.

Gigabit Ethernet has two distinctive approaches for medium access: half-duplex and full- duplex. Almost all implementations of Gigabit Ethernet follow the full-duplex approach.

Topology

Gigabit Ethernet is designed to connect two or more stations. If there are only two stations, they can be connected point-to-point. Three or more stations need to be connected in a star topology with a hub or a switch at the center. Another possible configuration is to connect several star topologies or let a star topology be part of another as shown in Figure 13.15.



a. Point-to-point

b. Star

c. Two stars

d. Hierarchy of stars

Figure 13.15 Topologies of Gigabit Ethernet

Implementation

Gigabit Ethernet can be categorized as either a two-wire or a four-wire implementation. The two-wire implementations use fiber-optic cable (1000Base-SX, short-wave, or 1000Base-LX, long-wave), or STP (1000Base-CX). The four-wire version uses category 5 twisted-pair cable (1000Base-T). In other words, we have four implementations, as shown in Figure 13.16.

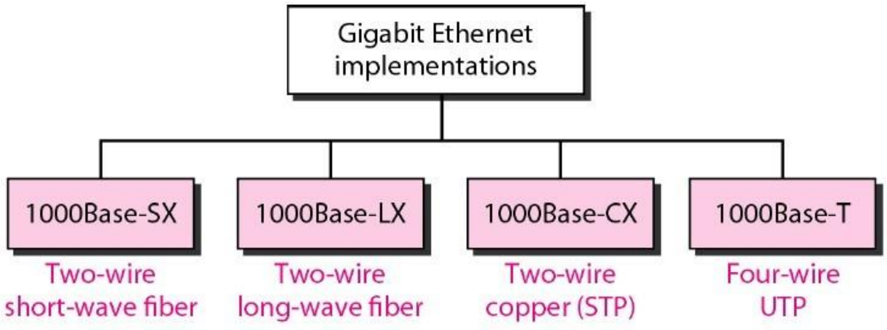


Figure 13.16 Gigabit Ethernet implementations

Summary

Table 13.3 is a summary of the Gigabit Ethernet implementations.

Table 13.3 Summary of Gigabit Ethernet implementations

Characteristics	**1000Base-SX**	**1000Base-LX**	**1000Base-CX**	**1000Base-T**
Media	Fiber Short wave	Fiber Long wave	STP	CAT 5 UTP
Number of wires	2	2	2	4
Maximum length	550m	5000m	25m	100m
Line encoding	NRZ	NRZ	NRZ	4D-PAM5

Ten-Gigabit Ethernet(IEEE 802.3ae)

The IEEE committee created Ten-Gigabit Ethernet and called it Standard 802.3ae. The goals of the Ten-Gigabit Ethernet design can be summarized as follows:

a. Upgrade the data rate to 10 Gbps.
b. Make it compatible with Standard, Fast, and Gigabit Ethernet.
c. Use the same 48-bit address.
d. Use the same frame format.
e. *S.* Keep the same minimum and maximum frame lengths.
f. Allow the interconnection of existing LANs into a metropolitan area network (MAN) or a wide area network (WAN).
g. Make Ethernet compatible with technologies such as Frame Relay and ATM.

Ten-Gigabit Ethernet operates only in **full duplex mode** which means there is no need for contention; *CSMA/CD* is not used in Ten-Gigabit Ethernet.

Implementation

Ten-Gigabit Ethernet is designed for using fiber-optic cable over long distances. Three implementations are the most common: 10GBase-S, 10GBase-L, and 10GBase-E. Table 13.4 shows a summary of the Ten-Gigabit Ethernet implementations:

Table 13.4 Summary of Ten-Gigabit Ethernet implementations

Characteristics	**10GBase-S**	**10GBase-L**	**10GBase-E**
Media	Short-wave 850-nm multimode	Long-wave 1310-nm Single mode	Extended 1550-mm Single mode
Maximum Length	300m	10km	40km

Introduction to Wireless LAN:

Wireless LAN stands for **Wireless Local Area Network**. It is also called LAWN (**Local Area Wireless Network**). WLAN is one in which a mobile user can connect to a Local Area Network (LAN) through a wireless connection.

The IEEE 802.11 group of standards defines the technologies for wireless LANs. For path sharing, 802.11 standard uses the Ethernet protocol and CSMA/CA (carrier sense multiple access with collision avoidance). It also uses an encryption method i.e. wired equivalent privacy algorithm.

Wireless LANs provide high speed data communication in small areas such as building or an office. WLANs allow users to move around in a confined area while they are still connected to the network.

In some instance wireless LAN technology is used to save costs and avoid laying cable, while in other cases, it is the only option for providing high-speed internet access to the public. Whatever the reason, wireless solutions are popping up everywhere.

Examples of WLANs that are available today are NCR's waveLAN and Motorola's ALTAIR.

Advantages of WLANs

- **Flexibility:** Within radio coverage, nodes can communicate without further restriction. Radio waves can penetrate walls, senders and receivers can be placed anywhere (also non-visible, e.g., within devices, in walls etc.).
- **Planning:** Only wireless ad-hoc networks allow for communication without previous planning, any wired network needs wiring plans.
- **Design:** Wireless networks allow for the design of independent, small devices which can for example be put into a pocket. Cables not only restrict users but also designers of small notepads, PDAs, etc.
- **Robustness:** Wireless networks can handle disasters, e.g., earthquakes, flood etc. whereas, networks requiring a wired infrastructure will usually break down completely in disasters.
- **Cost:** The cost of installing and maintaining a wireless LAN is on average lower than the cost of installing and maintaining a traditional wired LAN, for two reasons. First, after providing wireless access to the wireless network via an access point for the first user, adding additional users to a network will not increase the cost. And second, wireless

LAN eliminates the direct costs of cabling and the labor associated with installing and repairing it.

- **Ease of Use:** Wireless LAN is easy to use and the users need very little new information to take advantage of WLANs.

Disadvantages of WLANs

- **Quality of Services:** Quality of wireless LAN is typically lower than wired networks. The main reason for this is the lower bandwidth due to limitations is radio transmission, higher error rates due to interference and higher delay/delay variation due to extensive error correction and detection mechanisms.
- **Proprietary Solutions:** Due to slow standardization procedures, many companies have come up with proprietary solutions offering standardization functionality plus many enhanced features. Most components today adhere to the basic standards IEEE 802.11a or 802.11b.
- **Restrictions:** Several govt. and non-govt. institutions world-wide regulate the operation and restrict frequencies to minimize interference.
- **Global operation:** Wireless LAN products are sold in all countries so, national and international frequency regulations have to be considered.
- **Low Power:** Devices communicating via a wireless LAN are typically power consuming, also wireless devices running on battery power. Whereas the LAN design should take this into account and implement special power saving modes and power management functions.
- **License free operation:** LAN operators don't want to apply for a special license to be able to use the product. The equipment must operate in a license free band, such as the 2.4 GHz ISM band.
- **Robust transmission technology:** If wireless LAN uses radio transmission, many other electrical devices can interfere with them (such as vacuum cleaner, train engines, hair dryers, etc.).Wireless LAN transceivers cannot be adjusted for perfect transmission is a standard office or production environment.

Asynchronous Transfer Mode (ATM):

Why ATM networks?

1. Driven by the integration of services and performance requirements of both telephony and data networking: “broadband integrated service vision” (B-ISON).

2. Telephone networks support a single quality of service and are expensive to boot.

3. Internet supports no quality of service but is flexible and cheap.

4. ATM networks were meant to support a range of service qualities at a reasonable cost- intended to subsume both the telephone network and the Internet.

Asynchronous Transfer Mode (ATM):

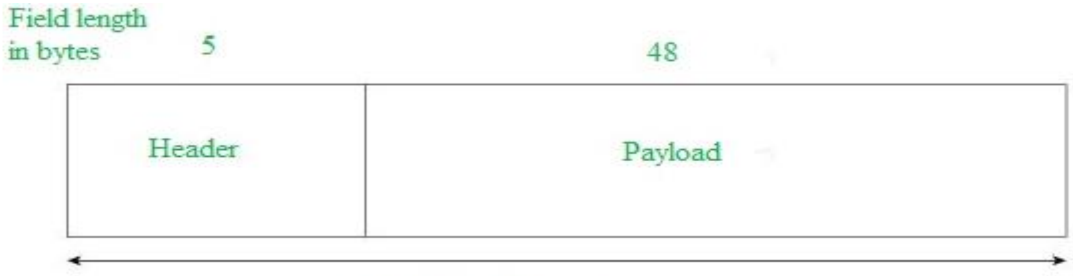
It is an International Telecommunication Union- Telecommunications Standards Section (ITU-T) efficient for call relay and it transmits all information including multiple service types such as data, video, or voice which is conveyed in small fixed-size packets called cells. Cells are transmitted asynchronously and the network is connection-oriented.

ATM is a technology that has some event in the development of broadband ISDN in the 1970s and 1980s, which can be considered an evolution of packet switching. *Each cell is 53 bytes long* – 5 bytes header and 48 bytes payload. Making an ATM call requires first sending a message to set up a connection.

Subsequently, all cells follow the same path to the destination. It can handle both constant rate traffic and variable rate traffic. Thus it can carry multiple types of traffic with **end-to-end** quality of service. ATM is independent of a transmission medium, they may be sent on a wire or fiber by themselves or they may also be packaged inside the payload of other carrier systems. ATM networks use "Packet" or "cell" Switching with virtual circuits. Its design helps in the implementation of high-performance multimedia networking.

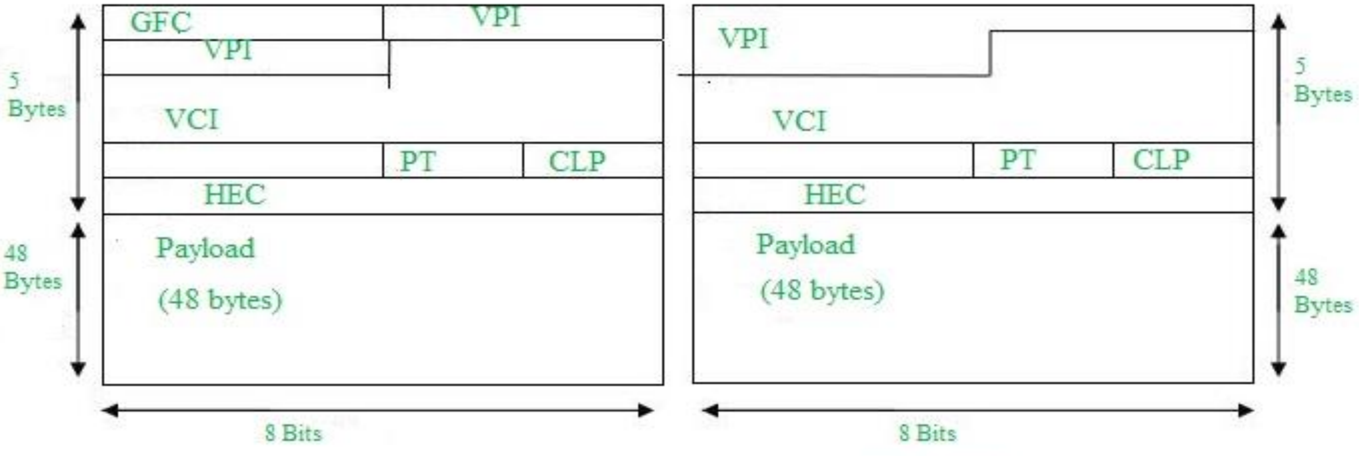
ATM Cell Format –

As information is transmitted in ATM in the form of fixed-size units called **cells**. As known already each cell is 53 bytes long which consists of a 5 bytes header and 48 bytes payload.



ATM Cell Format

Asynchronous Transfer Mode can be of two format types which are as follows:



UNI Cell Format

NNI Cell Format

1. **UNI Header:** This is used within private networks of ATMs for communication between ATM endpoints and ATM switches. It includes the Generic Flow Control (GFC) field.

2. **NNI Header:** is used for communication between ATM switches, and it does not include the Generic Flow Control (GFC) instead it includes a Virtual Path Identifier (VPI) which occupies the first 12 bits.

Working of ATM:
ATM standard uses two types of connections. i.e., Virtual path connections (VPCs) which consist of Virtual channel connections (VCCs) bundled together which is a basic unit carrying a single stream of cells from user to user. A virtual path can be created end-to-end across an ATM network, as it does not route the cells to a particular virtual circuit. In case of major failure, all cells belonging to a particular virtual path are routed the same way through the ATM network, thus helping in faster recovery.
Switches connected to subscribers use both VPIs and VCIs to switch the cells which are Virtual Path and Virtual Connection switches that can have different virtual channel connections between them, serving the purpose of creating a *virtual trunk* between the switches which can be handled as a single entity. Its basic operation is straightforward by looking up the connection value in the local translation table determining the outgoing port of the connection and the new VPI/VCI value of connection on that link.

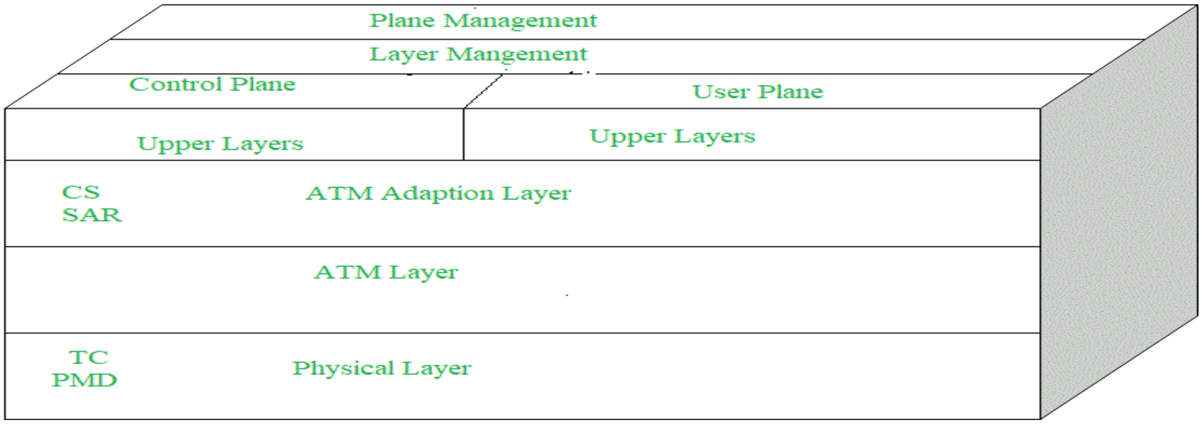
ATM vs DATA Networks (Internet)

- ATM is a "virtual circuit" based: the path is reserved before transmission. While Internet Protocol (IP) is connectionless and end-to-end resource reservations are not possible. RSVP is a new signaling protocol on the internet.

- ATM Cells: Fixed or small size and Tradeoff is between voice or data. While IP packets are of variable size.

- Addressing: ATM uses 20-byte global NSAP addresses for signaling and 32-bit locally assigned labels in cells. While IP uses 32-bit global addresses in all packets.

ATM Layers:



1. **ATM Adaption Layer (AAL) –**
   It is meant for isolating higher-layer protocols from details of ATM processes and prepares for conversion of user data into cells and segments it into 48-byte cell payloads. AAL protocol accepts transmission from upper-layer services and helps them in mapping applications, e.g., voice, data to ATM cells.
2. **ATM Layer –**
   It handles transmission, switching, congestion control, cell header processing, sequential delivery, etc., and is responsible for simultaneously sharing the virtual circuits over the physical link known as cell multiplexing and passing cells through an ATM network known as cell relay making use of the VPI and VCI information in the cell header.
3. **Physical Layer –**
   It manages the medium-dependent transmission and is divided into two parts physical medium-dependent sub layer and transmission convergence sub layer. The main functions are as follows:
   - It converts cells into a bit stream.
   - It controls the transmission and receipt of bits in the physical medium.
   - It can track the ATM cell boundaries.
   - Look for the packaging of cells into the appropriate type of frames.

ATM Applications:

1. **ATM WANs –**
   It can be used as a WAN to send cells over long distances, a router serving as an end-point between ATM network and other networks, which has two stacks of the protocol.

2. **Multimedia virtual private networks and managed services –**
   It helps in managing ATM, LAN, voice, and video services and is capable of full-service virtual private networking, which includes integrated access to multimedia.

3. **Frame relay backbone –**
   Frame relay services are used as a networking infrastructure for a range of data services and enabling frame-relay ATM service to Internetworking services.

4. **Residential broadband networks –**
   ATM is by choice provides the networking infrastructure for the establishment of residential broadband services in the search of highly scalable solutions.

5. **Carrier infrastructure for telephone and private line networks –**
   To make more effective use of SONET/SDH fiber infrastructures by building the ATM infrastructure for carrying the telephonic and private-line traffic.