BHARAT INSTITUTE OF ENGINEERING & TECHNOLOGY

BHARAT INSTITUTE OF ENGINEERING AND TECHNOLOGY

POLYTECHNIC

MOHADA,BERHAMPUR,GANJAM

LECTURE NOTES ON

DATA COMMUNICATION AND COMPUTER NETWORKS

2nd Year, 4th Semester

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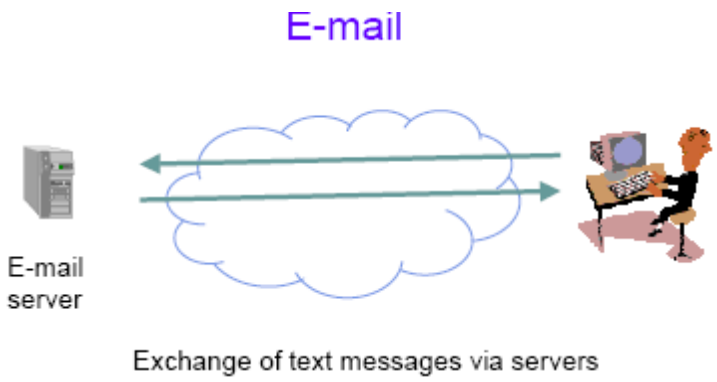
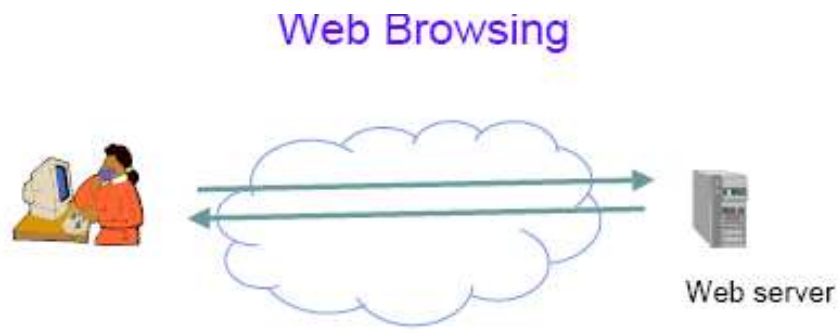
MODULE- I

Overview of Data Communication & Networking:

Data Communication:
The information is shared when we communicate. This sharing can be local or over long distance. Data refers to information presented in whatever form is agreed upon by the parties creating and using it. Data communications are the exchange of data between two devices via some form of transmission medium such as a wire cable. A communication service enables the exchange of information between users at different locations. The communicating devices must be a part of a communication system made up of a combination of hardware (physical equipment) and software (programs).
Communication services & applications are everywhere. Some examples are given below:



Exchange of text messages via servers



Retrieval of information from web servers

Characteristics of data Communication:

The effectiveness of a data communication system depends on Four fundamental characteristics:
1. Delivery
2. Accuracy
3. Timeliness
4. Jitter

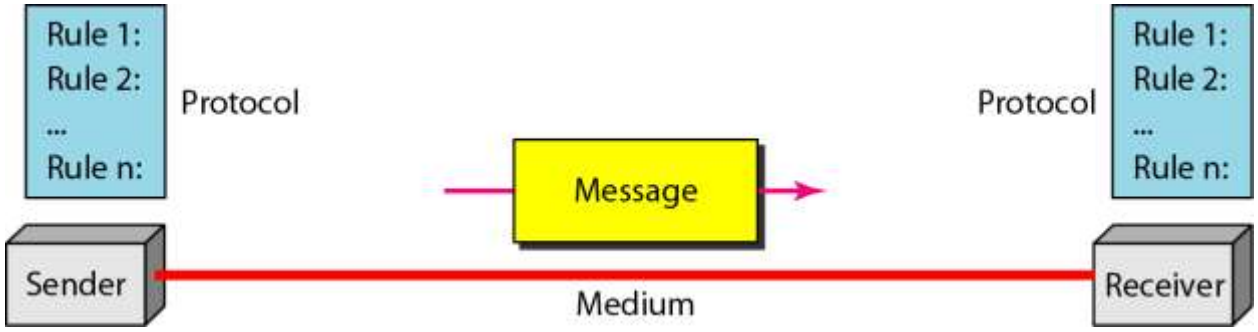
Delivery: The system must deliver data to correct destination.
Accuracy: The system must deliver data accurately.
Timeliness: The system must deliver data in a timely manner. Timely delivery means delivering data as they are produced, in the same order that they are produced and without significant delay. This kind of delivery is called real –time transmission.
Jitter: Jitter refers to the variation in the packet arrival time.

Simply we can say that a data communication system must deliver data to the correct destination in an accurate and timely manner.

Components:

The essential components of a data communication system are:

Message
Sender
Receiver
Medium
Protocol



Message: The information to be communicated. It can consist of text, pictures, numbers, sound, video or audio.
Sender: The sender is the device that sends the data message. It can be a computer or workstation telephone handset, video camera and so on.
Receiver: The receiver is the device that receives the message. It can be a computer or workstation telephone handset, video camera and so on.
Medium: The transmission medium is the physical path connecting both the sender as well as the receiver by which a message travels from sender to receiver. It could be a twisted pair wire, coaxial cable, fiber optic cable, or radio waves.
Protocol: A protocol is a set of rules that governs data communications. It represents an agreement between the communicating devices.

Data representation:

Information can be in any form such as text, numbers, images, audio and video.

Text

Text is represented as a bit pattern
The number of bits in a pattern depends on the number of symbols in that language.
Code is the set of bit patterns designed to represent text symbols.

ASCII

The American National Standards Institute developed a code called the American Standard code for Information Interchange (ASCII) .This code uses 7 bits for each symbol.

Extended ASCII

To make the size of each pattern 1 byte (8 bits), an extra 0 is augmented at the left the ASCII bit patterns which doesn't change the value of the pattern.

Unicode

To represent a symbol or code in any language Unicode is used. It uses 32 bits to represent.

ISO

The international organization for standardization known as ISO has designed a code using a 32 – bit pattern. This code can represent up to 4,294,967,296 symbols.

Numbers

Numbers are also represented by using bit patterns. Instead of using ASCII to represent numbers, the number is directly converted to a binary number.

Images

Images are also represented by bit patterns. An image is divided into a matrix of pixels (The smallest element of an image) where each pixel is a small dot having dimension. Each pixel is assigned a bit pattern. The size and value of the pattern depends on the image.

Audio

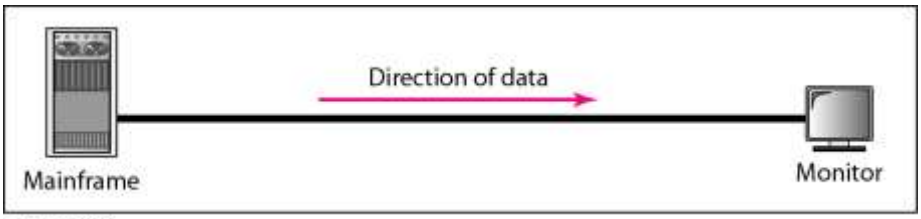
Audio is the recording or broadcasting of sound or music. Audio is by nature different from text, numbers or images. It is continuous not discrete.

Video

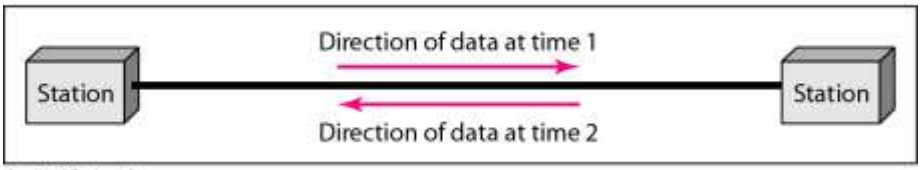
Video is the recording or broadcasting of picture or movie. Video can be produced either a continuous entity or it can be a combination of images.

Direction of data flow

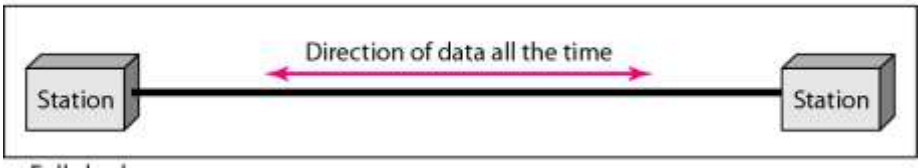
Two devices can communicate in simplex, half-duplex or full-duplex mode.



a. Simplex



b. Half-duplex



c. Full-duplex

Simplex:

In simplex mode, the communication is unidirectional. Only one of the devices on a link can transmit; the other can only receive.

Ex. Keyboard and monitor

Half-duplex

In half-duplex mode, each station can both transmit and receive but not at the same time. When one device is sending, the other can only receive.

Ex. Walkie-talkies and CB (citizen band radios)

Full-duplex

In full-duplex mode, both stations can transmit and receive simultaneously.

Ex. Telephone network

When two people are communicating by a telephone line, both can listen and talk at the same time.

Network:

Definition:

- A network is set of devices (nodes) connected by communication links (media)
- A node can be a computer, printer or other device capable of sending and/or receiving data
- Link connecting the devices are often called communication channels
- Most network use distributed processing.

Distributed Processing

Networks use distributed processing in which a task divided among multiple computers. Separate computers handle a subset instead of a single machine responsible for all aspects of a process.

Performance

Performance can be measured in terms of transit time, response time, number of users, type of transmission medium, and capabilities of the connected hardware and the efficiency of the software.

Transit time

The time required for a message to travel from one device to another.

Response time

The time spent between an inquiry and a response

Reliability

It is measured by the frequency of failure and time required to recover from a failure.

Security

Network security is protecting data from unauthorized access.

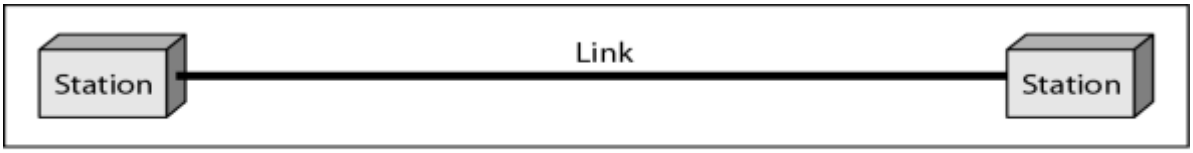
Type of connection

Two types of connections

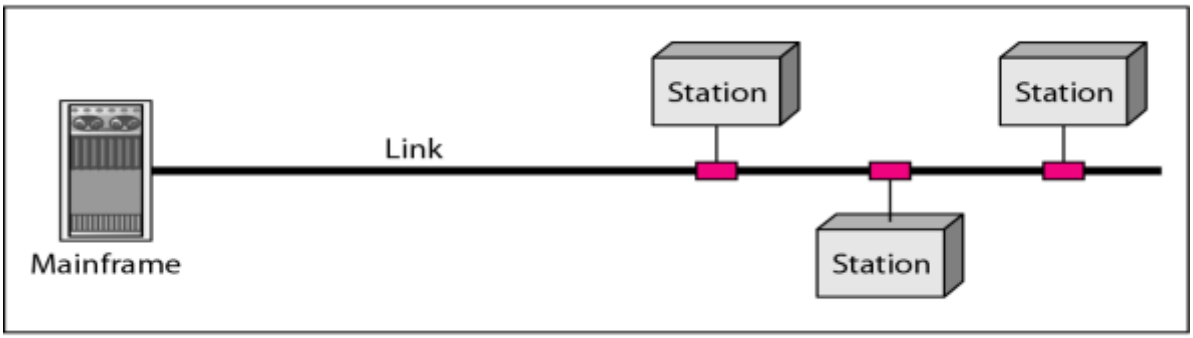
a. Point-to-point
b. Multipoint

In point-to-point connection the two devices are connected by a dedicated link. The entire capacity of the link is reserved for transmission between those two devices.

A multipoint (also known as multidrop) connection is one in which more than two specific devices share a single link. The capacity of the channel is shared either spatially or temporally.



a. Point-to-point



b. Multipoint

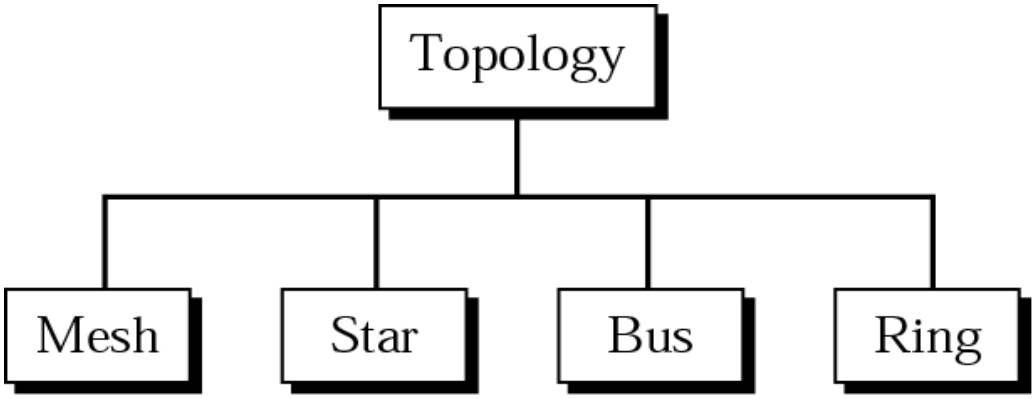
Physical Topology

Physical Topology refers to the way in which network is laid out physically. The topology of a network is the geometric representation of the relationship of all the links and the linking devices. The physical or logical arrangement of a network is also topology.

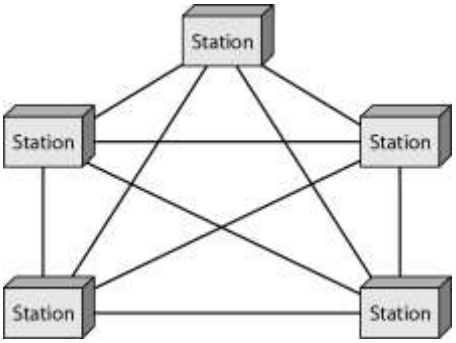
The basic topologies are

- Mesh

- Star
- Bus
- Ring



Mesh



- Dedicated point-to-point links to every other device
- Has *n*(*n*-1)/2 physical channel to link *n* devices
- Devices have *n*-1 I/O

Advantages:

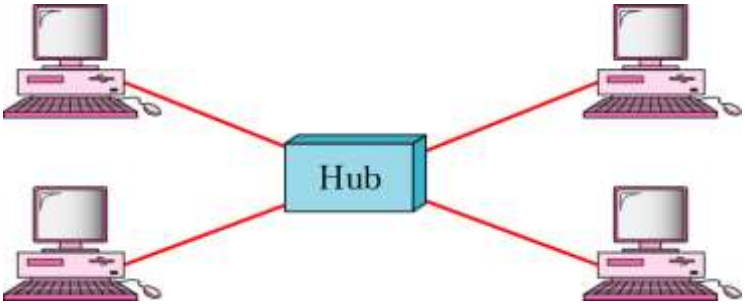
- Dedicated link guarantees that each connection can carry its own data load and thus eliminates the traffic problems that occur when links shared by multiple devices.
- If one link becomes unusable, it does not incapacitate the entire system.
- As every message travels along a dedicated line only the intended recipient, so it is secure.

Disadvantages

- More amount of cabling and the I/O ports required
- Installation and reconnection are difficult
- The hardware required to connect each link can be prohibitively expensive.

Star

- Dedicated point-to-point links to central controller (hub)
- Controller acts as exchange



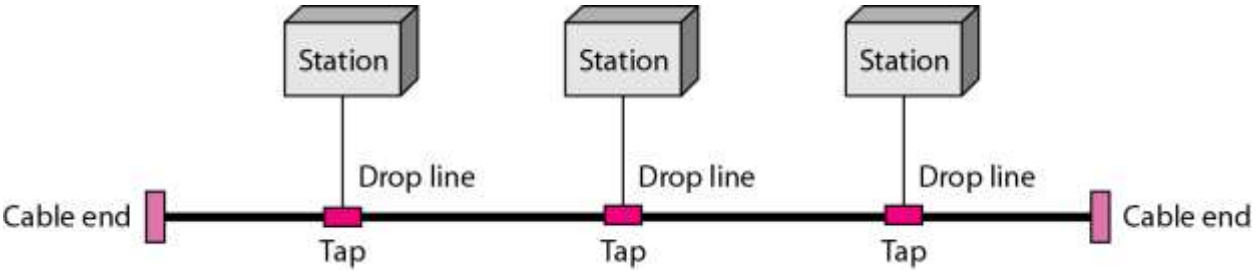
Advantages

- Less expensive than a mesh topology. Each device needs only one link and I/O port to connect with Hub
- Installation and reconfigure is easier.
- If one link fails only that link is affected.
- Requires less cable than a mesh.

Disadvantages

- Yet requires more cable compared to bus and ring topologies.

Bus



- Multipoint configuration
- One long cable acts as a backbone to link all devices.
- Stations are connected through tap and drop lines.

Advantages

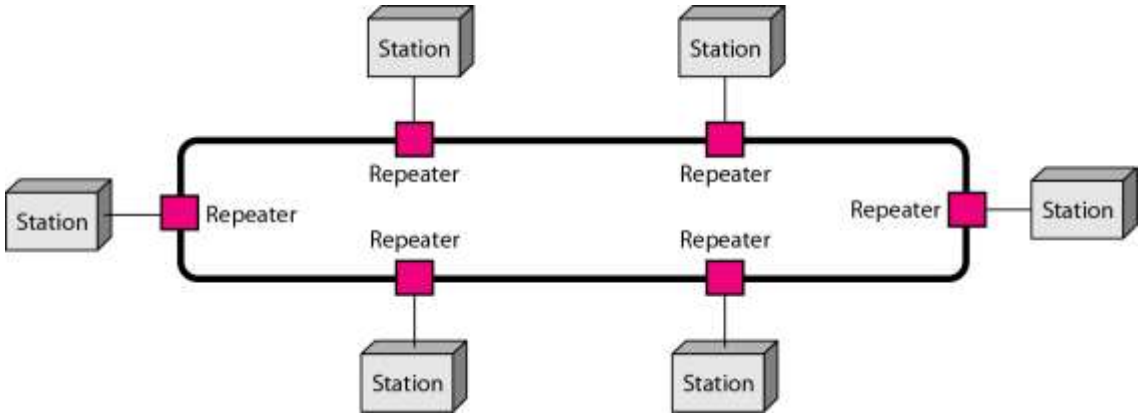
- Ease of installation.
- Uses less cabling than mesh or star topologies.

Disadvantages

- Difficult reconnection and isolation.
- Signal reflection at the taps can cause degradation in quality.
- A fault or break in the bus cable stops all transmission.

Ring

- Dedicated point-to-point configuration to neighbors
- Signal passes from device to device until it reaches destination
- Each device functions as a repeater



Advantages

- Easy to install and reconfigure.
- Only two connections are to be changed to add or delete a device.
- If one device does not receive the signal within a specified period, it issue an alarm that alerts the network operator to the problem and its location

Disadvantages

- A break in the ring breaks the entire network.

Categories of Network

Three primary categories of network

- Local Area Network (LAN)
- Metropolitan Area Network (MAN)
- Wide Area Network (WAN)

The category into which a network fall is determined by its size, ownership, the distance it covers and its physical architecture.

LAN

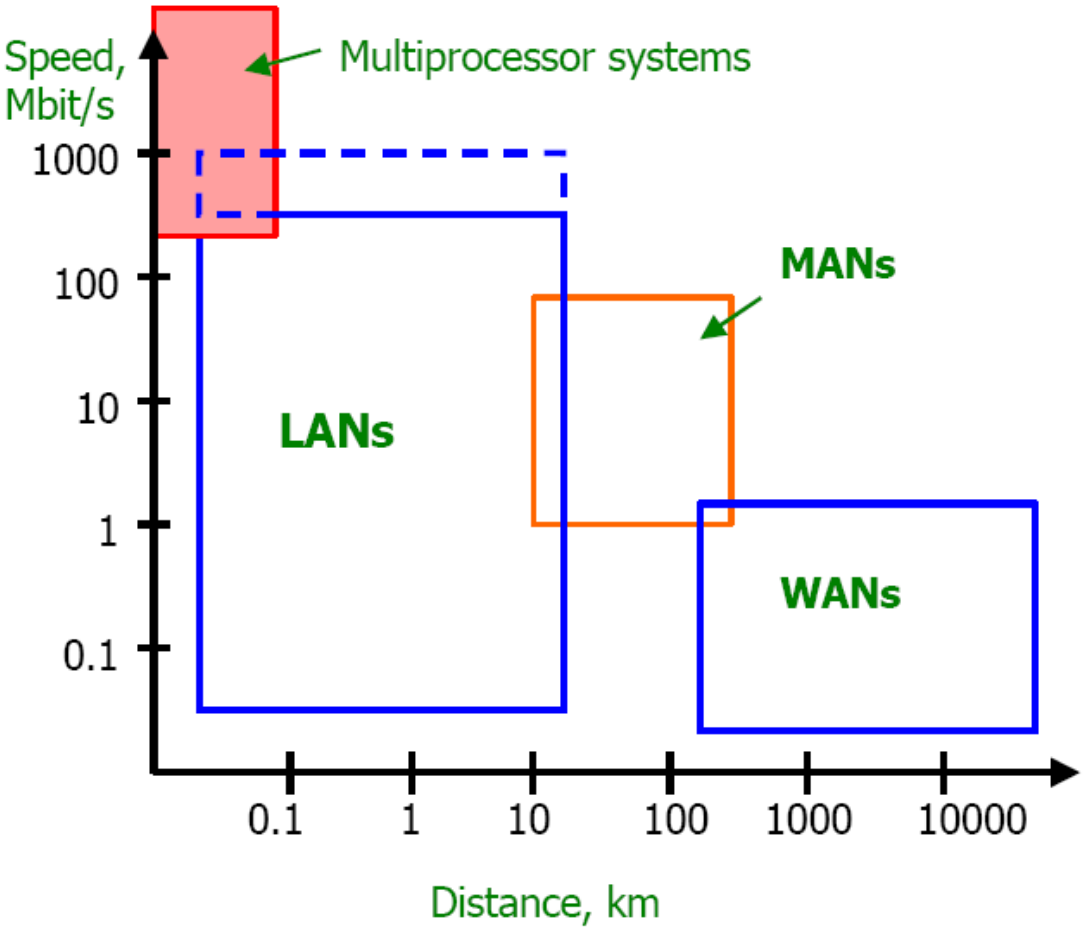
- Usually privately owned and links the devices in a single office, building, or campus
- LAN size is limited to a few kilometers.
- LANs are designed to allow resources to be shared (hardware , software and data)
- Today LANs to have data rates of 100 Mbps to 10Gbps
- Backbone Networks (BN), have a scale of a few hundred meters to a few kilometers. Include a high speed backbone linking the LANs at various locations.

MAN

- A MAN is designed to cover an entire city.
- May be a single network such as cable TV network
- May be a means of connecting a number of LANs into a larger network
- MANs have data rates of 1 Mbps to 100 Mbps
- Resources may be shared LAN to LAN as well as device to device
- A company can use a MAN to connect the LANs in all its offices throughout a city.
- A MAN can be owned by a private company or it may be a service provided by a public company ,such as local telephone company
- Telephone companies provide a popular MAN service called (SMDS) Switched Multi-megabit Data Services.

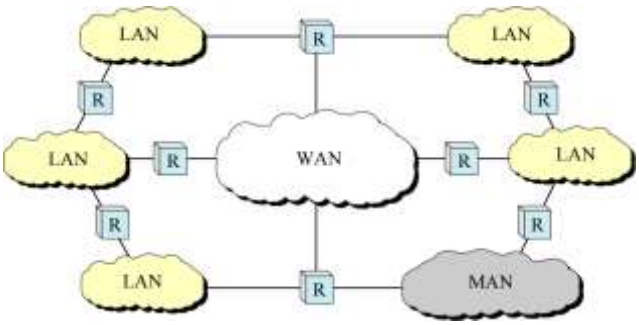
WAN

- WAN provides long distance transmission of data, voice, image, and video information over large geographical areas
- Comprise a country, a continent, or even the whole world (Interlink age of many LANs and MANs)
- Low data transmission rate (below 1 Mbps)
- Unlimited number of miles example: Internet Network



Internetwork

- Connection of two or more networks by the use of internetworking devices which include routers and gateways
- Internet is a generic term used to mean an interconnection of networks
- The Internet is the name of a specific worldwide network.



Protocols

- A protocol is a set of rules that governs data communication; the key elements of a protocol are

- ➡ Syntax – data formats and Signal levels
- ➡ Semantics – control information and error handling
- ➡ Timing – speed matching and sequencing

Standards are necessary to ensure that products from different manufacturers can work together as expected.

Standards

Why do we need standards?

- To create and maintain an open and competitive market for equipment manufacturers
- To guarantee national and international interoperability of data, telecommunication technology and process
- To give a fixed quality and product to the customer
- To allow the same product to be re used again elsewhere
- To aid the design and implementation of ideas
- To provide guidelines to manufacturers, vendors, government agencies and other service providers to ensure kind of interconnectivity.

Data communication standards are divided into two types

De facto (from the fact):

- Standards that have not been approved by an organized body.
- It has been adopted as standards through widespread use.
- This is often established originally by manufacturers to define the functionality of a new product or technology.

De jure (by law):

- Those that have been legislated by an officially recognized body.

Standards organizations

Standards are developed through the cooperation of standards creation committees, forums, and government regulatory agencies.

Standards Creation Committees

ITU, International Telecommunications Union formerly the (CCITT):

- It a standard for telecommunication in general and data systems in particular.

ISO, International Standards Organization:

- It is active in developing cooperation in the realms of scientific, technological and economic activity.

ANSI, American National Standards Institute:

- It is a private nonprofit corporation and affiliated with the U.S federal government.

IEEE, Institute of Electrical and Electronics Engineers:

- It aims to advance theory, creativity, and product quality in the fields of electrical engineering, electronics radio and in all related branches of Engineering.
- It oversees the development and adoption of international standards for computing and communications. See http://standards.ieee.org/

EIA, Electronic Industries Association:

- It is a nonprofit organization devoted to the promotion of electronics manufacturing concerns.
- Its activities include public awareness education and lobbying efforts in addition to standards development.
- It also made significant contributions by defining physical connection interfaces and electronic signaling specifications for data communication.

Forums

- It works with universities and users to test, evaluate and standardize new technologies.
- The forums are able to speed acceptance and use of those technologies in the telecommunications community.

- It presents their conclusions to standard bodies.

Regulatory Agencies:

- Its purpose is to protect the public interest by regulating radio, television and wire cable communications.
- It has authority over interstate and international commerce as it relates to communication.

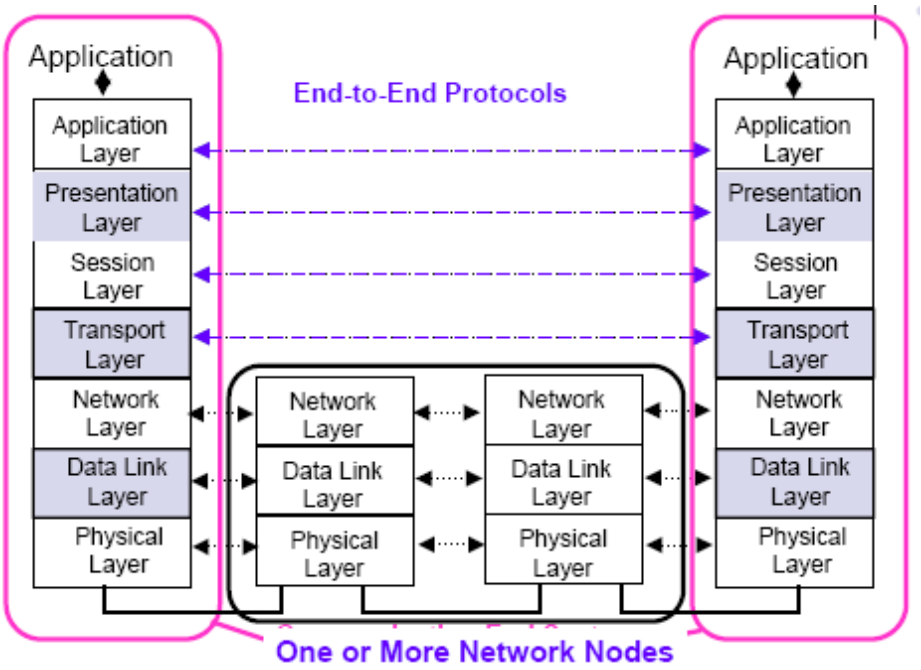
Internet Standards

- It is a thoroughly tested specification that is useful to and adhered to by those who work with the internet.
- It is a formalized regulation that must be followed.
- A specification begins as an internet draft and attains Internet standard status.
- An Internet draft is a working document and it may be published as Request for Comment (RFC).RFC is edited, assigned a number, and made available to all interested parties.

OSI Reference Model

Describes a seven-layer abstract reference model for a network architecture

Purpose of the reference model was to provide a framework for the development of protocols



Physical Layer

- It coordinates the functions required to transmit a bit stream over a physical medium.
- It deals with the mechanical and electrical specifications of the interface and transmission media.
  Mechanical: cable, plugs, pins...
  Electrical/optical: modulation, signal strength, voltage levels,
  bit times,
- It also defines the procedures and functions that physical devices and interfaces have to perform for transmission to occur

Major responsibilities of Physical layer are

- **Physical characteristics of interfaces and media:**
  It defines the characteristics of the interface between the devices and the transmission media. Also defines the type of transmission medium.
- **Representation of bits**:
  To transmit the bits, it must be encoded into electrical or optical signals. It defines the type of representation how 0s and 1s are changed to signals.
- **Data rate:**
  The number of bits sent each second is also defined by the physical layer.
- **Synchronization of bits:**
  Sender and the receiver must be synchronized at the bit level .i.e the sender and the receiver clocks must be synchronized.

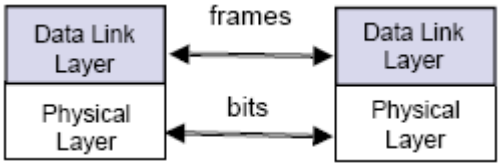
Data link layer

The data link layer is responsible for hop-to-hop (node-to-node) delivery. It transforms the physical layer a raw transmission facility to a reliable link. It makes physical layer appear error free to the network layer. The duties of the data link layer are

- Framing: The data link layer divides the stream of bits received from the network layer into manageable data units called frames.
- Physical Addressing: If the frames are to be distributed to different systems on the network the data link layer adds a header to the frame to define the receiver or sender of the frame. If the frame is intended

for a system located outside the senders' network then the receiver address is the address of the connecting device that connects the network to the next one.

- Flow Control: If the rate at which the data absorbed by the receiver is less than the rate produced in the sender, the data link layer imposes a flow control mechanism to overwhelming the receiver.
- Error control: Reliability is added to the physical layer by data link layer to detect and retransmit loss or damaged frames. and also to prevent duplication of frames. This is achieved through a trailer added to the end of the frame
- Access control: when two or more devices are connected to the same link it determines which device has control over the link at any given time.



Network Layer

The network layer is responsible for source-to-destination delivery of a packet across multiple networks. It ensures that each packet gets from its point of origin to its final destination .It does not recognize any relationship between those packets. It treats each one independently as though each belong to separate message.

The functions of the network layer are

- Logical Addressing If a packet has to cross the network boundary then the header contains information of the logical addresses of the sender and the receiver.

Networking When independent networks or links are connected to create an internetwork or a large network the connective devices route the packet to the final destination.

Transport Layer

The network layer is responsible for process-to-process delivery that is source to destination delivery of the entire message.

The responsibilities of Transport layer are

- Service-point (port) addressing: Computers run several programs at the same time. Source-to-destination delivery means delivery from a specific process on one computer to a specific process on the other. The transport layer header therefore includes a type of address called a service – point address.
- Segmentation and reassembly A message is divided into segments and each segment contains a sequence number. These numbers enable the Transport layer to reassemble the message correctly upon arriving at the destination. The packets lost in the transmission is identified and replaced.
- Connection control: The transport layer can be either connectionless or connection-oriented. A connectionless transport layer treats segment as an independent packet and delivers it to the transport layer. A connection-oriented transport layer makes a connection with the transport layer at the destination machine and delivers the packets. After all the data are transferred the connection is terminated.
- Flow control: Flow control at this layer is performed end to end.
- Error Control: Error control is performed end to end. At the sending side, the transport layer makes sure that the entire message arrives at the receiving transport layer with out error. Error correction is achieved through retransmission.

Session Layer: Session layer is the network dialog controller. It establishes, maintains, and synchronizes the interaction between communicating systems. Specific responsibilities of the layer are

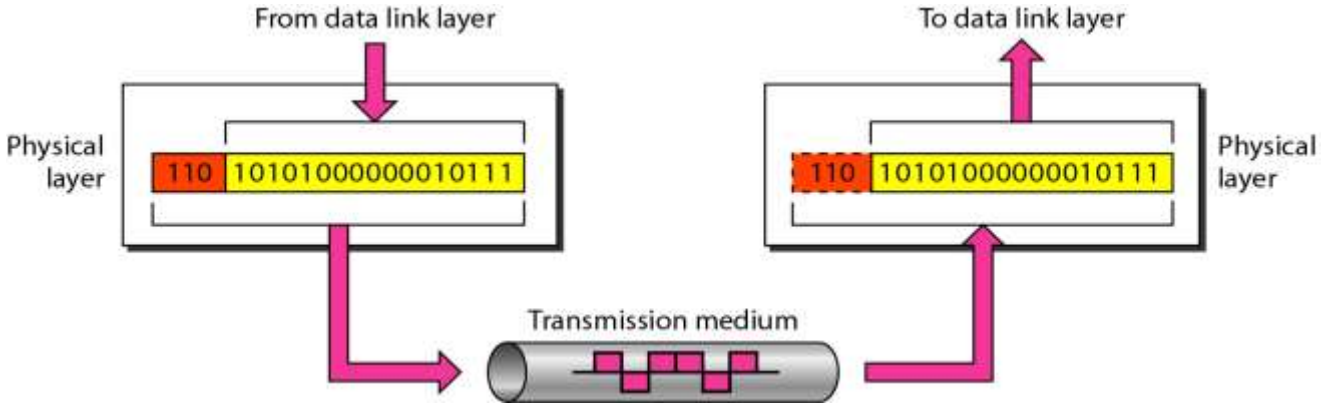
- Dialog Control: Session layer allows two systems to enter in to a dialog. Communication between two processes takes place either in half-duplex or full-duplex. Example: the dialog between a terminal connected to a mainframe. Can be half-duplex.

- Synchronization. The session layer allows a process to add checkpoints into a stream of data. Example If a system is sending a file of 2000 pages, check points may be inserted after every 100 pages to ensure that each 100 page unit is advised and acknowledged independently. So if a crash happens during the transmission of page 523, retransmission begins at page 501, pages 1 to 500 need not be retransmitted.

  Presentation layer: It is concerned with the syntax and semantics of the information exchanged between two systems. Responsibilities of the presentation layer are
- Translation .The processes in two systems are usually exchanging information in the form of character strings, numbers, and so on. The Since different computers use different encoding systems, the presentation layer is responsible for interoperability between these different encoding methods. At the sender, the presentation layer changes the information from its sender-dependent format into a common format. The presentation layer at the receiving machine changes the common format into its receiver dependent format.
- Encryption. The sender transforms the original information from to another form and sends the resulting message over the entire network. Decryption reverses the original process to transform the message back to its original form.
- Compression. It reduces the number of bits to be transmitted. It is important in the transmission of text, audio and video.

  Application Layer: It enables the user (human/software) to access the network. It provides user interfaces and support for services such as electronic mail, remote file access and transfer, shared database management and other types of distributed information services. Services provided by the application layer are
- Network Virtual terminal. A network virtual terminal is a software version of a physical terminal and allows a user to log on to a remote host.
- File transfer, access and management. This application allows a user to access files in a remote computer, to retrieve files from a remote computer and to manage or control files in a remote computer.
- Mail services. This application provides the basis for e-mail forwarding and storage.
- Directory services. It provides distributed database sources and access for global information about various objects and services.

PHYSICAL LAYER



To be transmitted, data must be transformed to electromagnetic signals.

ANALOG AND DIGITAL

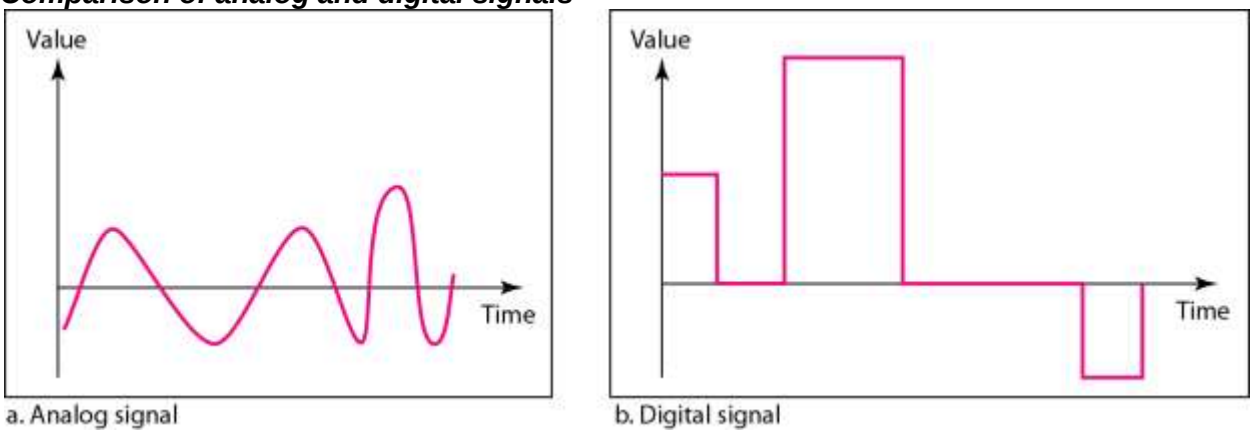
Data can be analog or digital. The term analog data refers to information that is continuous; digital data refers to information that has discrete states. Analog data take on continuous values. Digital data take on discrete values.

- Analog data refers to information that is continuous
- Analog data take on continuous values
- Digital data refers to information that has discrete states
- Digital data take on discrete values

Like data signals can be analog or digital. Analog signals can have an infinite number of values in a range; digital signals can have only a limited number of values.

In data communications, we commonly use periodic analog signals and nonperiodic digital signals.

Comparison of analog and digital signals

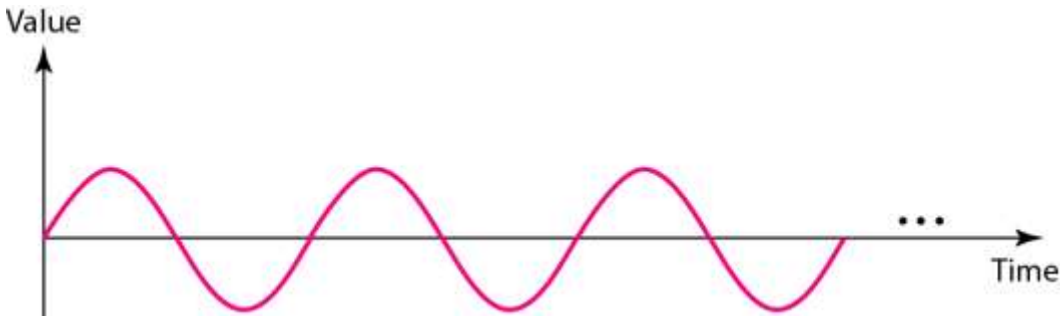


a. Analog signal

b. Digital signal

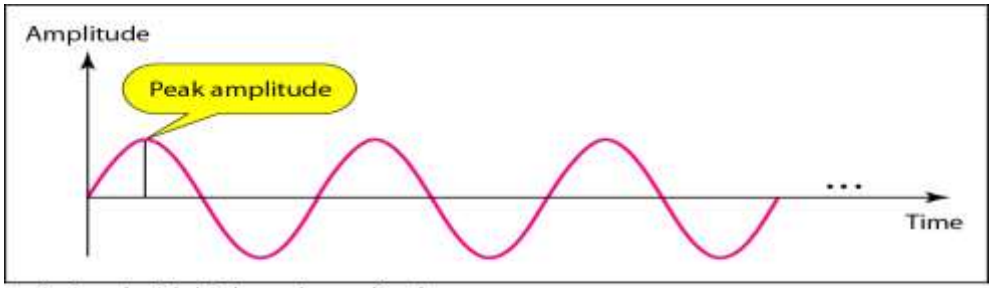
PERIODIC ANALOG SIGNALS

Periodic analog signals can be classified as simple or composite. A simple periodic analog signal, a sine wave, cannot be decomposed into simpler signals. A composite periodic analog signal is composed of multiple sine waves.

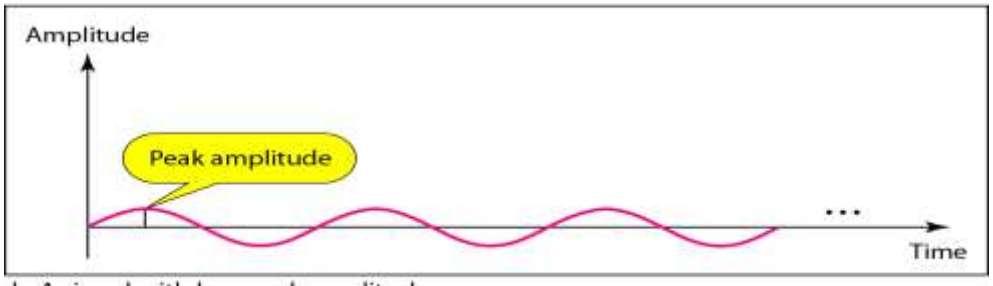


Signal amplitude
Peak Amplitude is the absolute value of its highest intensity proportional to the energy it carries. The unit is either Amp or volt.

Figure *Two signals with the same phase and frequency, but different amplitudes*



a. A signal with high peak amplitude



b. A signal with low peak amplitude

Frequency

Frequency is the rate of change with respect to time.

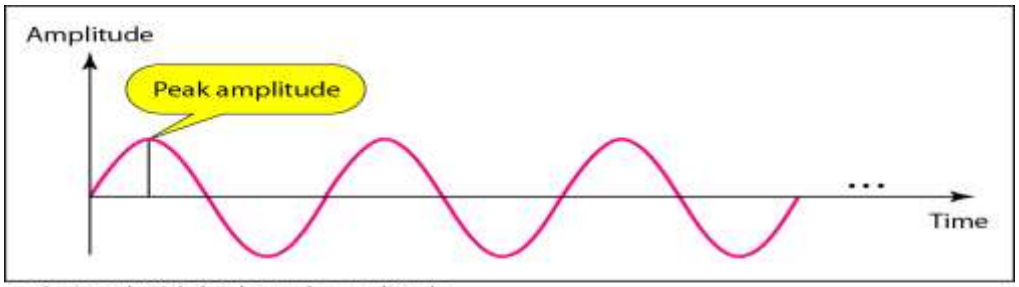
- ❑ Change in a short span of time means high frequency.
- ❑ Change over a long span of time means low frequency.
- ❑ If a signal does not change at all, its frequency is zero
- ❑ If a signal changes instantaneously, its frequency is infinite.

Frequency and Period
Frequency and period are the inverse of each other.

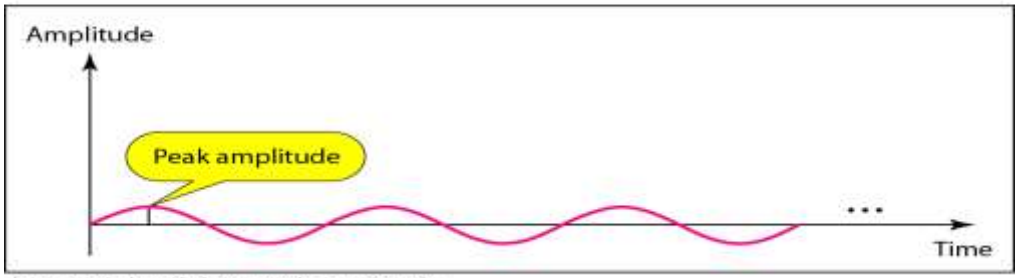
$$f = \frac{1}{T} \quad \text{and} \quad T = \frac{1}{f}$$

The units of period and frequency are sec and Hz.

Figure Two signals with the same phase and frequency, but different amplitudes



a. A signal with high peak amplitude



b. A signal with low peak amplitude

Examples:

The power we use at home has a frequency of 60 Hz. What is the period of this sine wave?

$$T = \frac{1}{f} = \frac{1}{60} = 0.0166 \text{ s} = 0.0166 \times 10^3 \text{ ms} = 16.6 \text{ ms}$$

The period of a signal is 100 ms. What is its frequency in kilohertz?

Solution

First we change 100 ms to seconds, and then we calculate the frequency from the period (1 Hz = 10^{-3} kHz).

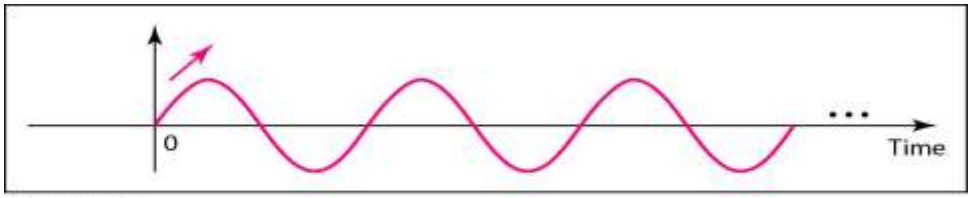
$$100 \text{ ms} = 100 \times 10^{-3} \text{ s} = 10^{-1} \text{ s}$$

$$f = \frac{1}{T} = \frac{1}{10^{-1}} \text{ Hz} = 10 \text{ Hz} = 10 \times 10^{-3} \text{ kHz} = 10^{-2} \text{ kHz}$$

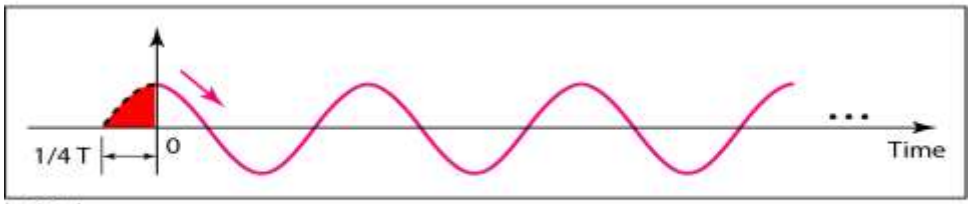
Phase

Phase describes the position of the waveform relative to time 0.

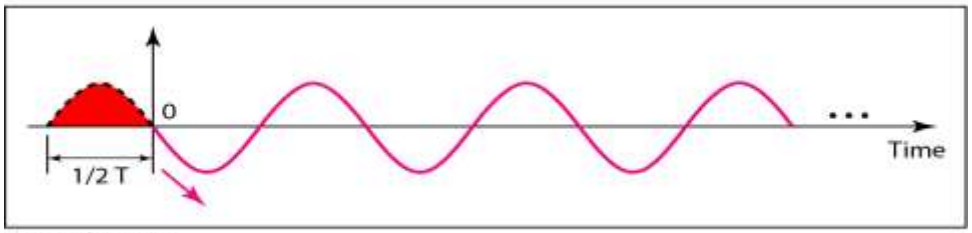
Figure Three sine waves with the same amplitude and frequency, but different phases



a. 0 degrees



b. 90 degrees



c. 180 degrees

Example

A sine wave is offset 1/6 cycle with respect to time 0. What is its phase in degrees and radians?

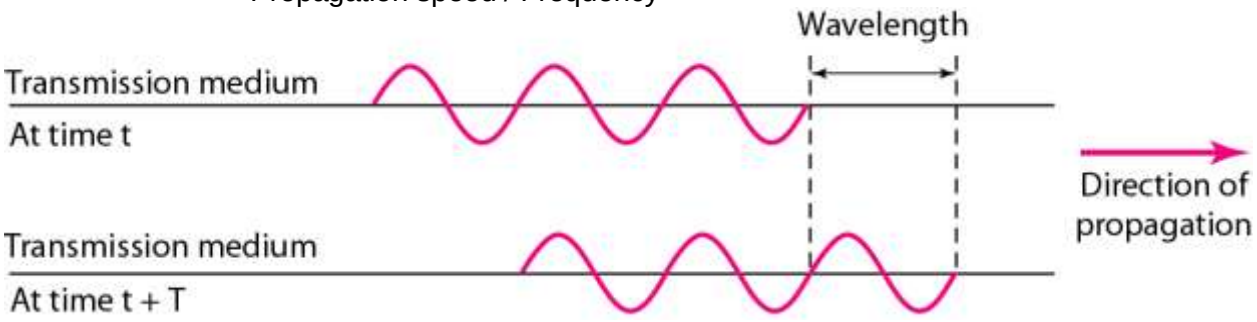
Solution

We know that 1 complete cycle is 360°. Therefore, 1/6 cycle is

$$\frac{1}{6} \times 360 = 60° = 60 \times \frac{2\pi}{360} \text{ rad} = \frac{\pi}{3} \text{ rad} = 1.046 \text{ rad}$$

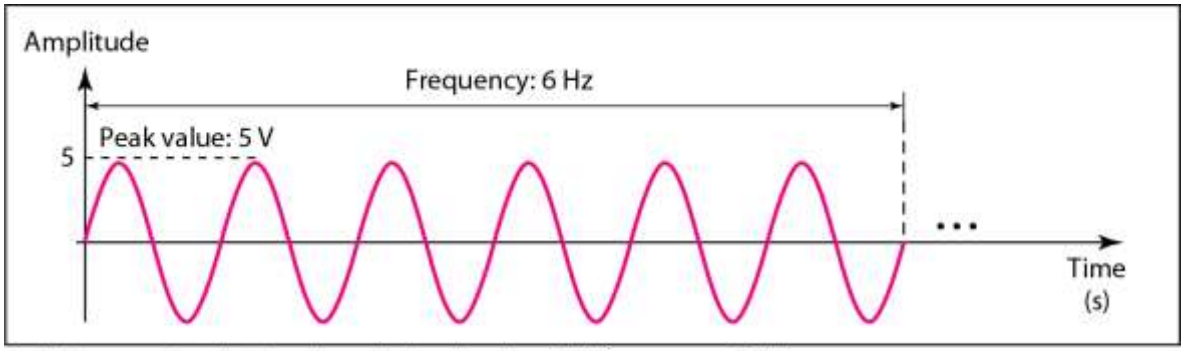
Wavelength and period

Wavelength = Propagation speed x Period
= Propagation speed / Frequency

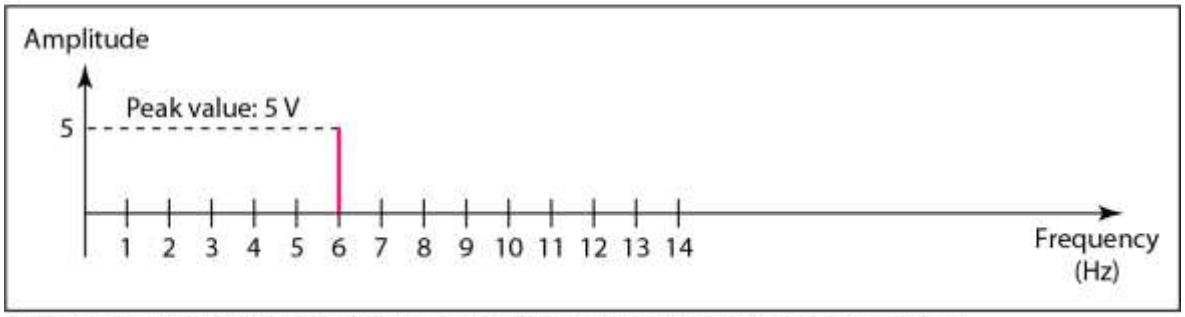


Time-domain and frequency-domain plots of a sine wave

A complete sine wave in the time domain can be represented by one single spike in the frequency domain.

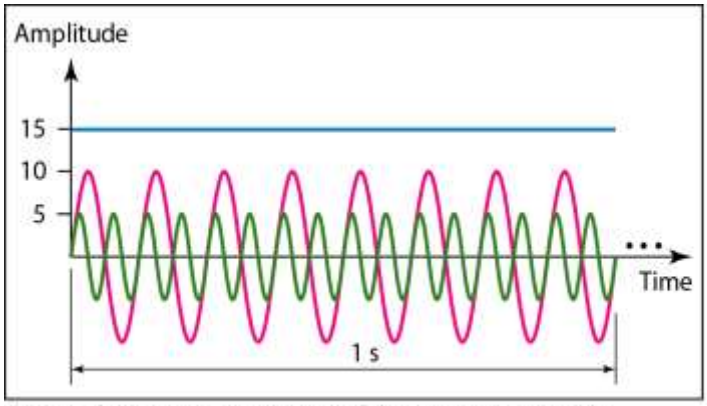


a. A sine wave in the time domain (peak value: 5 V, frequency: 6 Hz)

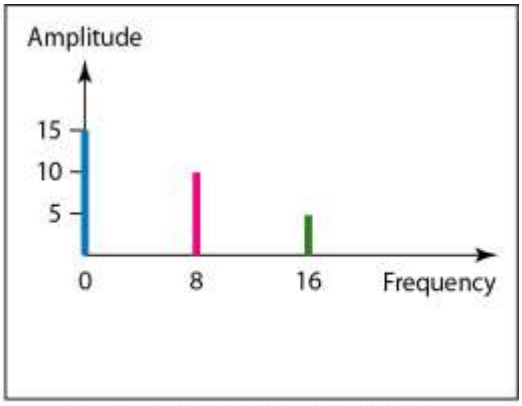


b. The same sine wave in the frequency domain (peak value: 5 V, frequency: 6 Hz)

Frequency Domain



a. Time-domain representation of three sine waves with frequencies 0, 8, and 16



b. Frequency-domain representation of the same three signals

- The frequency domain is more compact and useful when we are dealing with more than one sine wave.
- A single-frequency sine wave is not useful in data communication
- We need to send a composite signal, a signal made of many simple sine waves.

A single-frequency sine wave is not useful in data communications; we need to send a composite signal, a signal made of many simple sine waves.

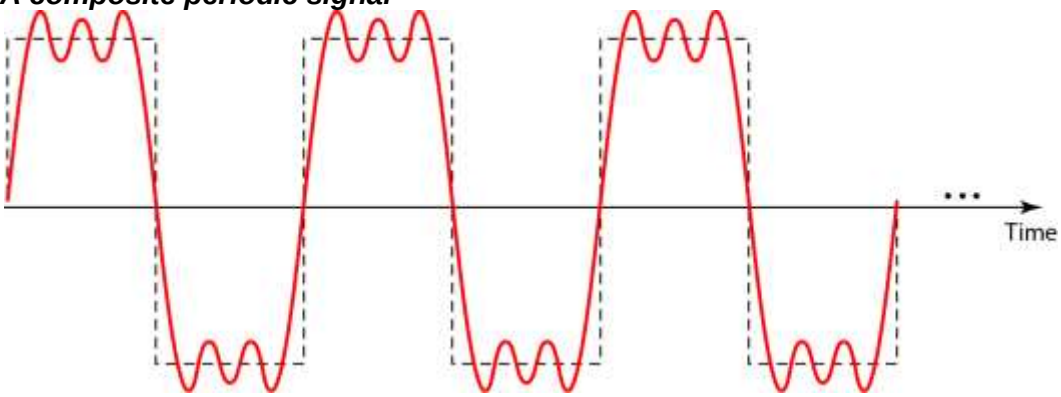
Fourier analysis

According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases.

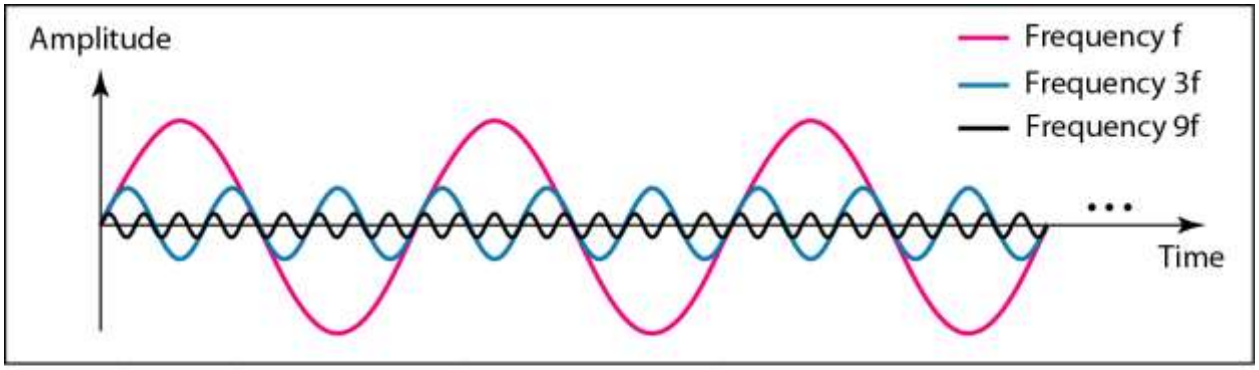
❖ If the composite signal is periodic, the decomposition gives a *series of signals with discrete frequencies*;

❖ If the composite signal is nonperiodic, the decomposition gives a *combination of sine waves with continuous frequencies*.

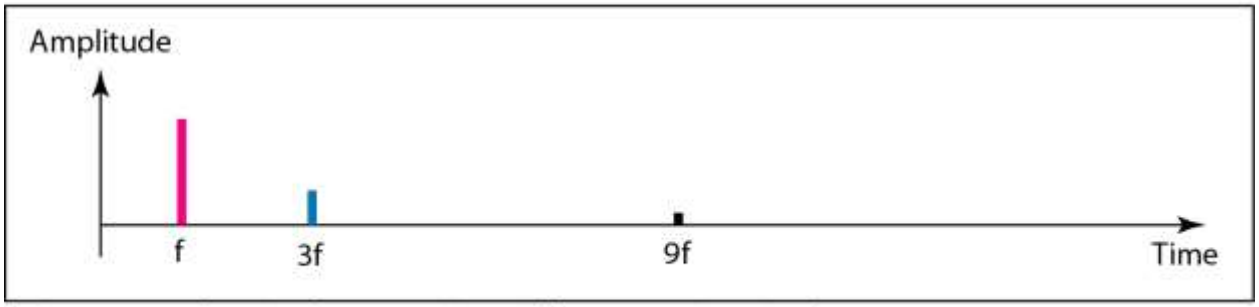
A composite periodic signal



Decomposition of a composite periodic signal in the time and frequency domains



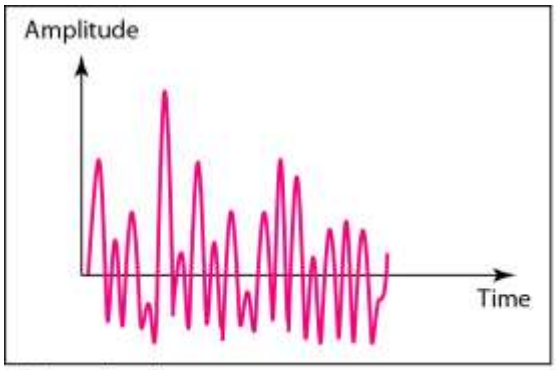
a. Time-domain decomposition of a composite signal



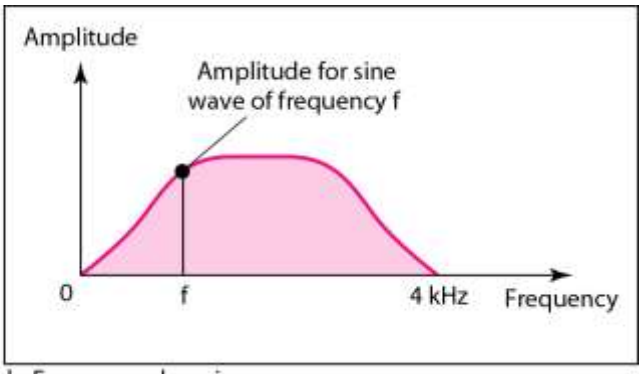
b. Frequency-domain decomposition of the composite signal

Time and frequency domains of a nonperiodic signal

- A nonperiodic composite signal
  - It can be a signal created by a microphone or a telephone set when a word or two is pronounced.
  - In this case, the composite signal cannot be periodic
    - because that implies that we are repeating the same word or words with exactly the same tone.



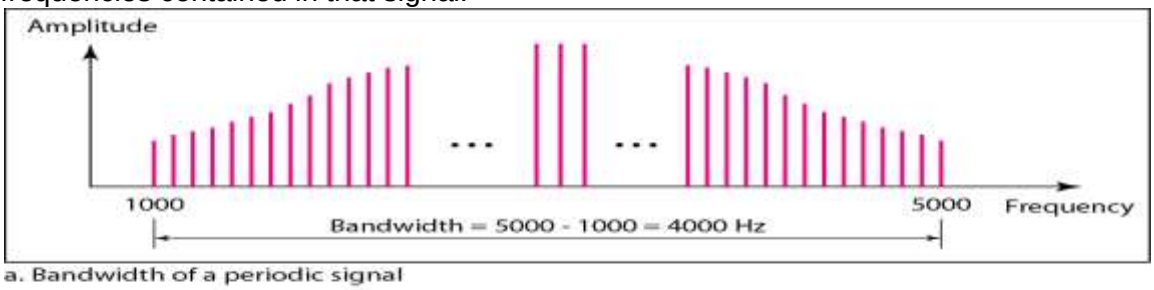
a. Time domain



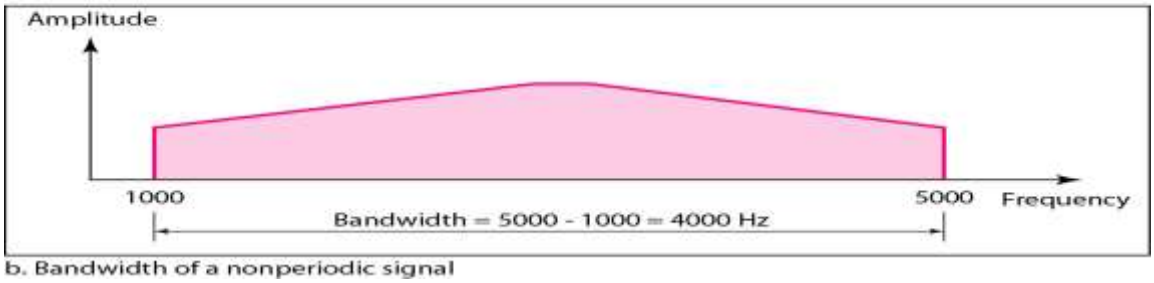
b. Frequency domain

Bandwidth

The bandwidth of a composite signal is the difference between the highest and the lowest frequencies contained in that signal.



a. Bandwidth of a periodic signal



b. Bandwidth of a nonperiodic signal

Example:

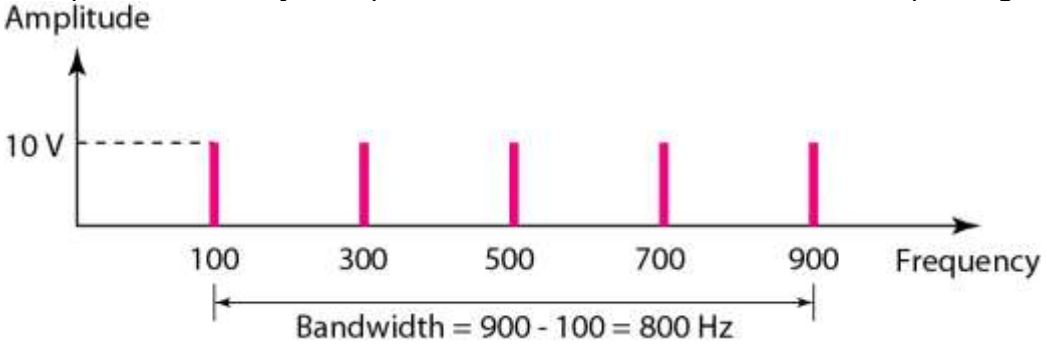
If a periodic signal is decomposed into five sine waves with frequencies of 100, 300, 500, 700, and 900 Hz, what is its bandwidth? Draw the spectrum, assuming all components have a maximum amplitude of 10 V.

Solution

Let f_h be the highest frequency, f_l the lowest frequency, and B the bandwidth. Then

$$B = f_h - f_l = 900 - 100 = 800 \text{ Hz}$$

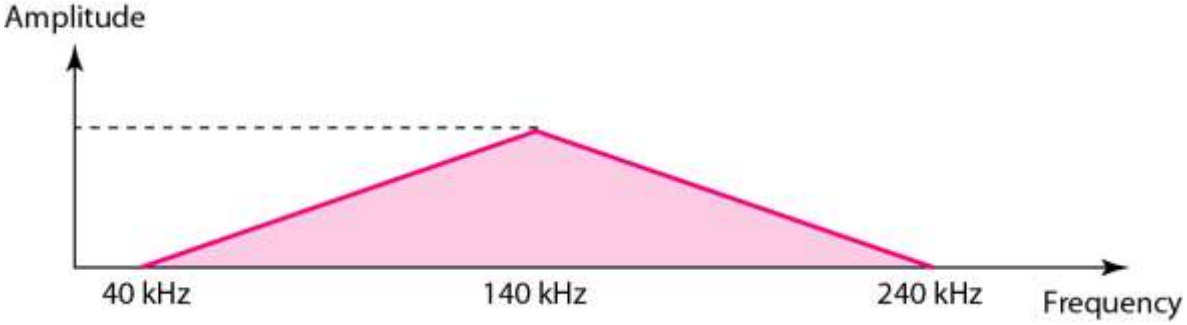
The spectrum has only five spikes, at 100, 300, 500, 700, and 900 Hz (see Figure)



Example:
A nonperiodic composite signal has a bandwidth of 200 kHz, with a middle frequency of 140 kHz and peak amplitude of 20 V. The two extreme frequencies have an amplitude of 0. Draw the frequency domain of the signal.

Solution

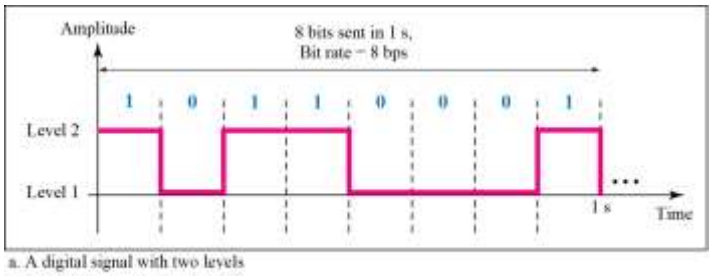
The lowest frequency must be at 40 kHz and the highest at 240 kHz. Figure 3.15 shows the frequency domain and the bandwidth.



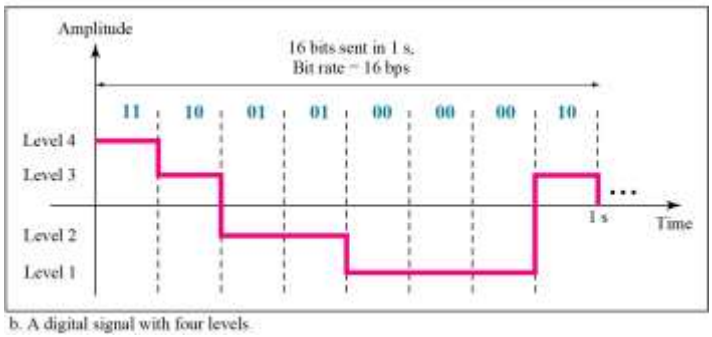
DIGITAL SIGNALS

- In addition to being represented by an analog signal, information can also be represented by a digital signal.
- For example, a 1 can be encoded as a positive voltage and a 0 as zero voltage.
- A digital signal can have more than two levels.
- In this case, we can send more than 1 bit for each level.
- Bit rate is the no of bits transmitted per sec.
- Bit interval is the time required to send one bit.
- Signal level is the no of bits required to represent a particular signal.
- Data level is the no of bits used to represent the data.

Figure Two digital signals: one with two signal levels and the other with four signal levels



a. A digital signal with two levels



b. A digital signal with four levels

Examples

1. A digital signal has 8 levels. How many bits are needed per level?

We calculate the number of bits from the formula

$$\text{Number of bits per level} = \log_2 8 = 3$$

Each signal level is represented by 3 bits.

2. A digital signal has 9 levels. How many bits are needed per level?
   Each signal level is represented by 3.17 bits.
   The number of bits sent per level needs to be an integer as well as a power of 2.
   Hence, 4 bits can represent one level.

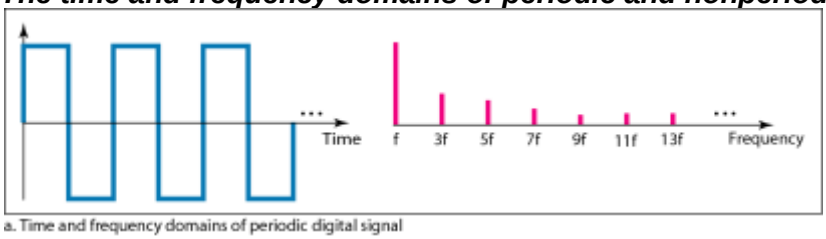
3. Assume we need to download files at a rate of 100 pages per minute. A page is an average of 24 lines with 80 characters in each line where one character requires 8 bits. What is the required bit rate of the channel?

$$100 \times 24 \times 80 \times 8 = 1{,}636{,}000 \text{ bps} = 1.636 \text{ Mbps}$$

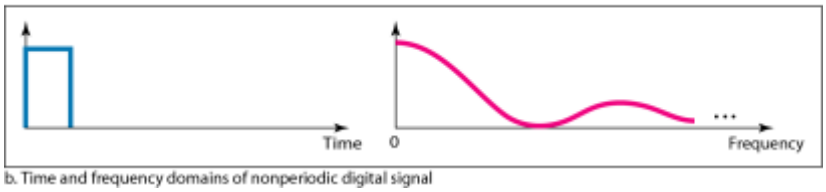
4. A digitized voice channel is made by digitizing a 4-kHz bandwidth analog voice signal. We need to sample the signal at twice the highest frequency (two samples per hertz). Assume that each sample requires 8 bits. What is the required bit rate?

$$2 \times 4000 \times 8 = 64{,}000 \text{ bps} = 64 \text{ kbps}$$

The time and frequency domains of periodic and nonperiodic digital signals



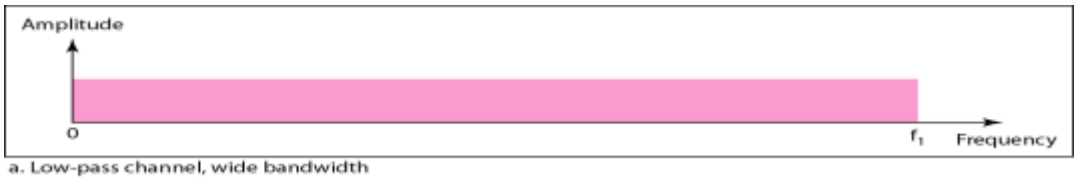
a. Time and frequency domains of periodic digital signal



b. Time and frequency domains of nonperiodic digital signal

Bandwidths of two low-pass channels

Digital transmission needs a low-pass channel whereas analog transmission can use a band-pass channel. Baseband transmission of a digital signal that preserves the shape of the digital signal is possible only if we have a low-pass channel with an infinite or very wide bandwidth.



a. Low-pass channel, wide bandwidth

b. Low-pass channel, narrow bandwidth

DATA RATE LIMITS

A very important consideration in data communications is how fast we can send data, in bits per second, over a channel. Data rate depends on three factors:

1. The bandwidth available
2. The level of the signals we use
3. The quality of the channel (the level of noise)

Increasing the levels of a signal may reduce the reliability of the system

Nyquist Theorem

For noiseless channel,

BitRate = 2 x Bandwith x $\log_2$Levels

In baseband transmission, we said the bit rate is 2 times the bandwidth if we use only the first harmonic in the worst case.

However, the Nyquist formula is more general than what we derived intuitively; it can be applied to baseband transmission and modulation.

Also, it can be applied when we have two or more levels of signals.

Examples:

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels. What is the maximum bit rate?

$$\text{BitRate} = 2 \times 3000 \times \log_2 2 = 6000 \text{ bps}$$

Consider the same noiseless channel transmitting a signal with four signal levels (for each level, we send 2 bits). What is the maximum bit rate?

$$\text{BitRate} = 2 \times 3000 \times \log_2 4 = 12{,}000 \text{ bps}$$

Shannon Capacity

In reality, we can not have a noiseless channel. For noisy channel,

Capacity = Bandwidth x $\log_2$(1+SNR)

The Shannon capacity gives us the upper limit; the Nyquist formula tells us how many signal levels we need.

Example:

We have a channel with a 1-MHz bandwidth. The SNR for this channel is 63.

What are the appropriate bit rate and signal level?

Solution

First, we use the Shannon formula to find the upper limit.

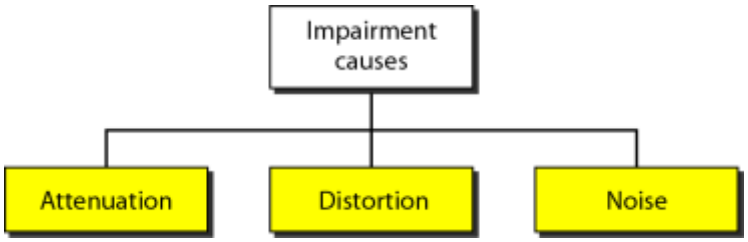
$$C = B \log_2 (1 + \text{SNR}) = 10^6 \log_2 (1 + 63) = 10^6 \log_2 64 = 6 \text{ Mbps}$$

The Shannon formula gives us 6 Mbps, the upper limit. For better performance we choose something lower, 4 Mbps, for example.
Then we use the Nyquist formula to find the number of signal levels.

$$4 \text{ Mbps} = 2 \times 1 \text{ MHz} \times \log_2 L \quad \rightarrow \quad L = 4$$

TRANSMISSION IMPAIRMENT

- Signals travel through transmission media, which are not perfect.
- The imperfection causes signal impairment.
- This means that the signal at the beginning of the medium is not the same as the signal at the end of the medium.
- What is sent is not what is received.
- Three causes of impairment are attenuation, distortion, and noise.



Attenuation: Loss of energy i.e. Loss in signal strength. It is measured in decibel (dB)
Suppose a signal travels through a transmission medium and its power is reduced to one-half. This means that P_2 is (1/2)P_1. In this case, the attenuation (loss of power) can be calculated as

$$10 \log_{10} \frac{P_2}{P_1} = 10 \log_{10} \frac{0.5 P_1}{P_1} = 10 \log_{10} 0.5 = 10(-0.3) = -3 \text{ dB}$$

Distortion: The change in shape or form of a signal. This occurs mainly in composite signals.
Noise: Unwanted signal mixed with the original signal. The noise can be of different types like Thermal Noise, Induced Noise, Cross Talk and Impulse Noise.
Thermal Noise: Produced due to random movement of free electrons in a wire creating extra unwanted signal.
Induced Noise: Produced if the source is an electrical motor or appliance.
Cross talk: Effect of one wire over another.
Impulse Noise: Is a spike produced during earth quake, thunder and lightning etc.
Example:
The power of a signal is 10 mW and the power of the noise is 1 μW; what are the values of SNR and SNR_{dB}?
Solution
The values of SNR and SNR_{dB} can be calculated as follows:

$$\text{SNR} = \frac{10{,}000 \ \mu\text{W}}{1 \text{ mW}} = 10{,}000$$
$$\text{SNR}_{\text{dB}} = 10 \log_{10} 10{,}000 = 10 \log_{10} 10^4 = 40$$

More about Signals:

In data communication four other measurements are used. They are Throughput, Propagation speed, Propagation time and wave length.

Throughput: It is the measurement of how fast data can pass through an entity.

Propagation Speed: It measures the distance that a signal or bit can pass through the medium in one second.

Propagation time: The time required by a signal to travel from one point of transmission to another. Propagation time= Distance/Propagation speed

Wave length: Wave length= Propagation speed X period

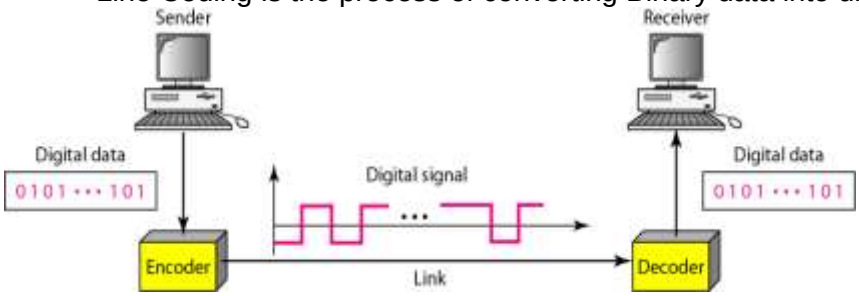
DIGITAL TRANSMISSION

DIGITAL-TO-DIGITAL CONVERSION:

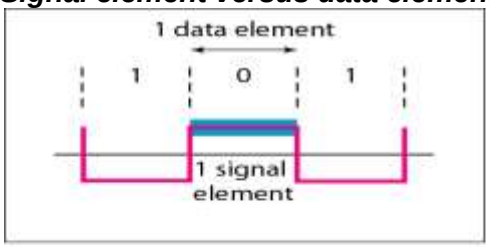
- We can represent digital data by using digital signals.
- The conversion involves three techniques: line coding, block coding, and scrambling.
  - Line coding is always needed.
  - Block coding and scrambling may or may not be needed.

Line Coding & Decoding:

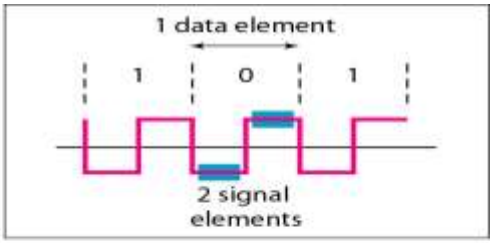
Line Coding is the process of converting Binary data into digital signals.



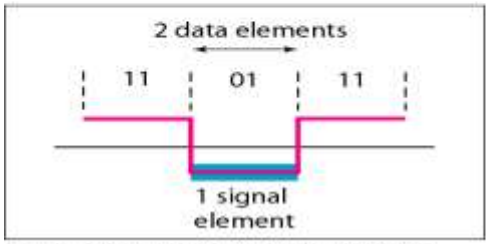
Signal element versus data element



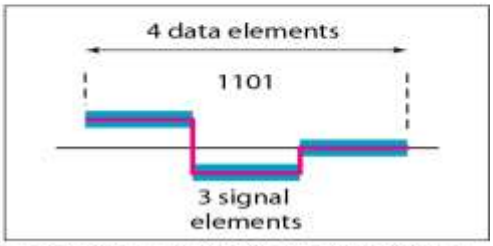
a. One data element per one signal element (r = 1)



b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$



c. Two data elements per one signal element (r = 2)

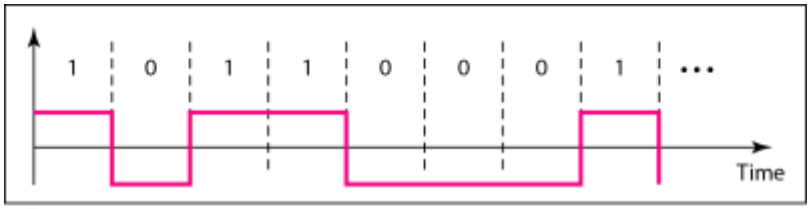


d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

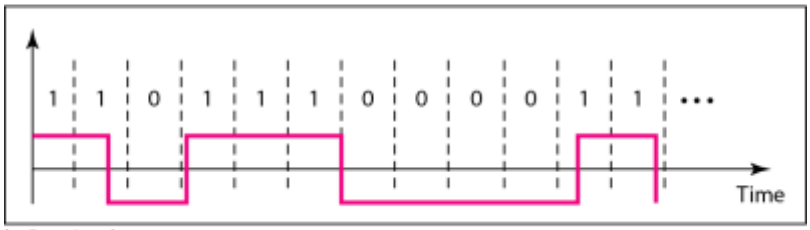
Effect of lack of synchronization

Self-synchronization To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals. If the receiver clock is faster or slower, the bit intervals are not matched and the receiver might misinterpret the signals.

A self-synchronizing digital signal includes timing information in the data being transmitted. This can be achieved if there are transitions in the signal that alert the receiver to the beginning, middle, or end of the pulse. If the receiver's clock is out of synchronization, these points can reset the clock

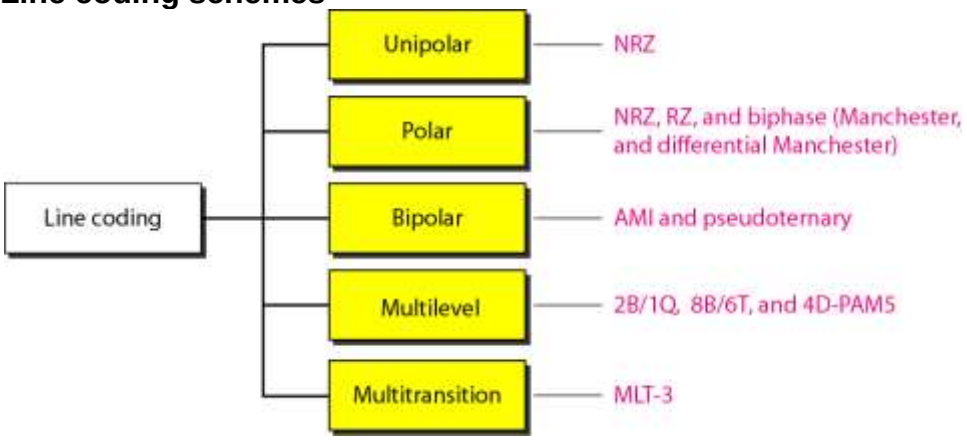


a. Sent



b. Received

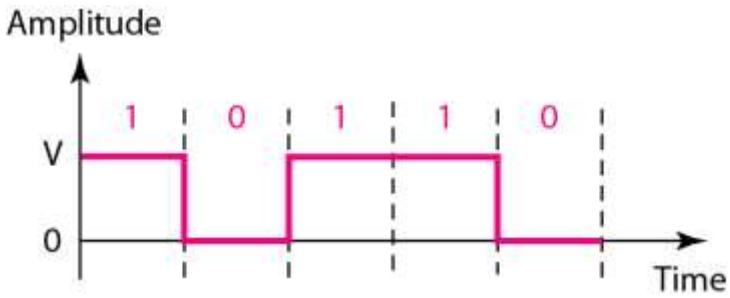
Line coding schemes



Unipolar NRZ scheme:

Unipolar encoding uses only one polarity.0 is represented by zero voltage and 1 is represented by positive voltage. It is inexpensive to implement. Unipolar encoding has two problems :

- Lack of synchronization
- A dc component



$$\frac{1}{2}V^2 + \frac{1}{2}(0)^2 = \frac{1}{2}V^2$$

Normalized power

Polar NRZ-L and NRZ-I schemes:

NRZ

The level of the signal is always either positive or negative.

NRZ-L

The level of the signal depends on the type of bit it represents.

The bit 0 is represented by positive voltage

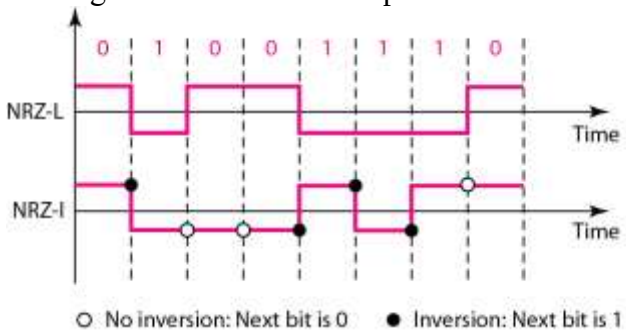
The bit 1 is represented by negative voltage.

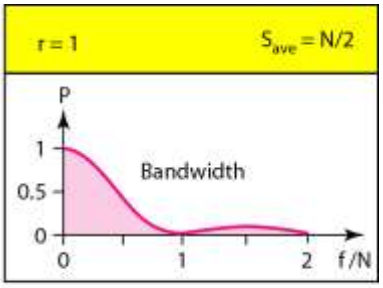
NRZ-I

The 1 bit is represented by an inversion (transition between a positive and a negative voltage) of the voltage level.

The existence of 1's in the data stream allows the receiver to resynchronize its timer to the actual arrival of the transmission.

A string of 0's can still cause problems.





In NRZ-L the level of the voltage determines the value of the bit. In NRZ-I the inversion or the lack of inversion determines the value of the bit.

Polar RZ scheme:

RZ

It uses three values

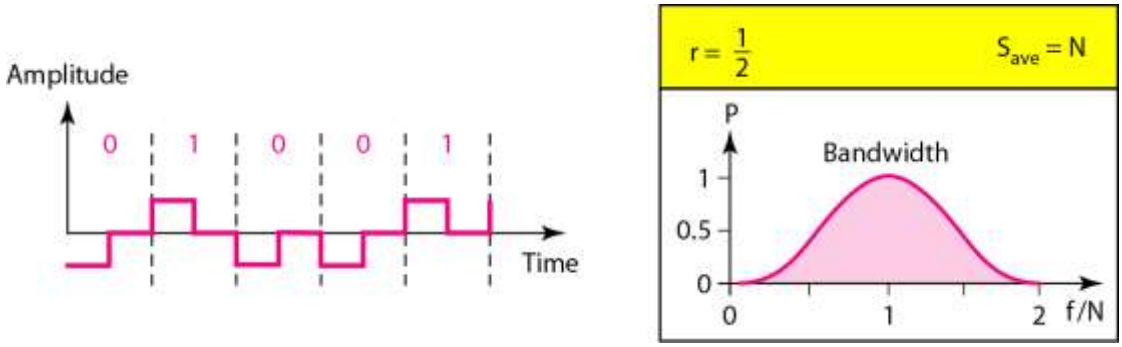
- ❖ Positive
- ❖ Negative
- ❖ Zero

In RZ the signal changes during each bit. A 1 bit is actually represented by positive-to-zero and A 0 bit is actually represented by negative-to-zero.

Demerits

It requires two signal changes to encode one bit.

It occupies more bandwidth.



Polar biphase: Manchester and differential Manchester schemes:

Biphase

The signal changes at the middle of the bit interval and does not return to zero.

There are two types of biphase encoding

- ❖ Manchester
- ❖ Differential Manchester

Manchester

- ❖ It uses the inversion at the middle of each bit interval for both synchronization and bit representation.
- ❖ The bit 1 is represented by negative -to-positive transition.
- ❖ The bit 0 is represented by positive-to-negative transition
  - ❖ Transition at the middle is used for synchronization
  - ❖ The minimum bandwidth is 2 times that of NRZ

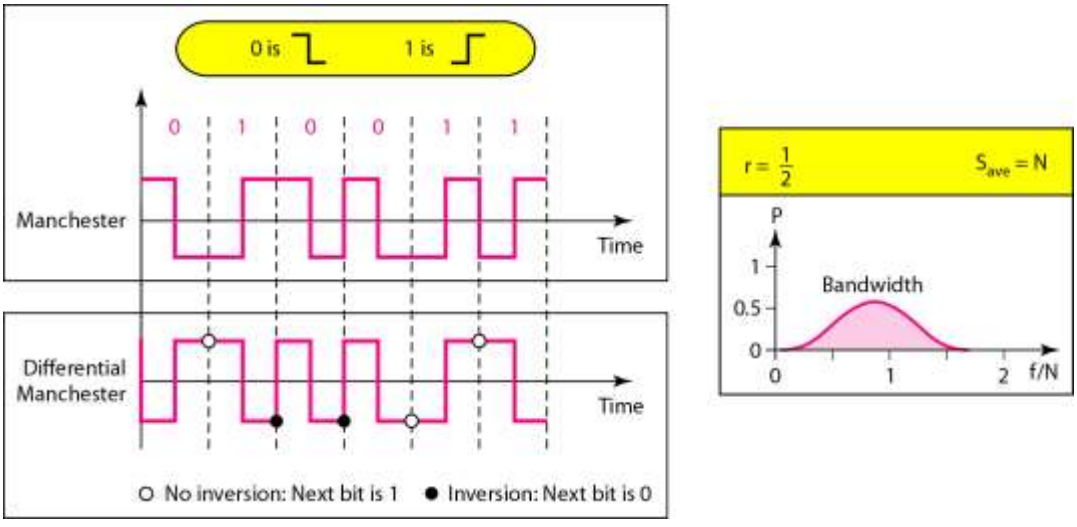
Differential Manchester

Inversion at the middle of the bit interval is used for synchronization.

Presence or absence of additional transition at the beginning of the interval is used to identify the bit.

A bit 0 is represented by a transition. A bit 1 means no transition.

It requires two signal changes to represent binary 0,but only one to represent binary 1.



The minimum bandwidth of Manchester and differential Manchester is 2 times that of NRZ.

Bipolar schemes: AMI and pseudo ternary:

In bipolar encoding, we use three levels: positive, zero, and negative.

- The bit 0 is represented by zero level

- The 1s are represented by alternate positive and negative voltages. If the first 1 bit is represented by positive amplitude, the second will be represented by the negative amplitude, and so on.

Bipolar Alternate Mark Inversion

A binary 0 is represented by zero voltage.

A binary 1s are represented by alternate positive and negative voltages.

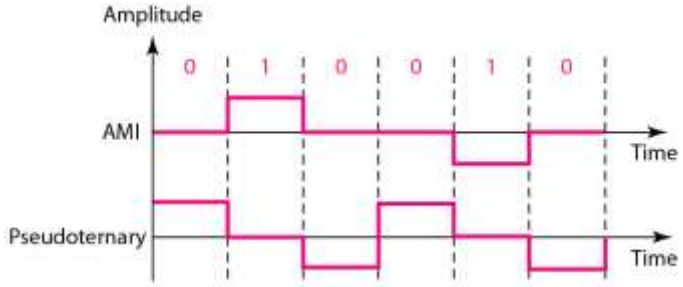
Merits

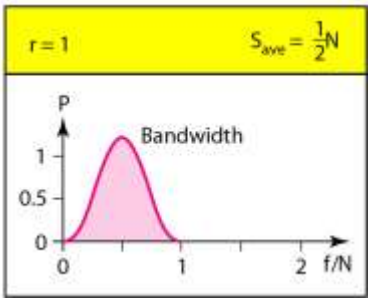
By inverting on each occurrence of 1, The dc component is zero

A long sequence of 1s stays synchronized.

Pseudoternary

A binary 0 alternate between positive and negative voltages.



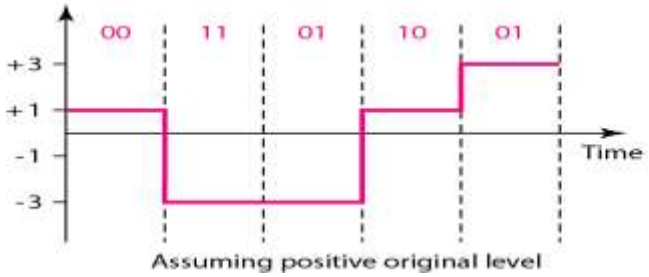


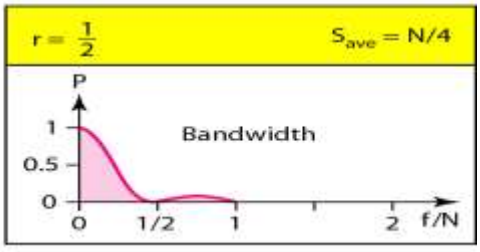
In *m*B*n*L schemes, a pattern of *m* data elements is encoded as a pattern of *n* signal elements in which $2^m \leq L^n$.

Multilevel: 2B1Q scheme

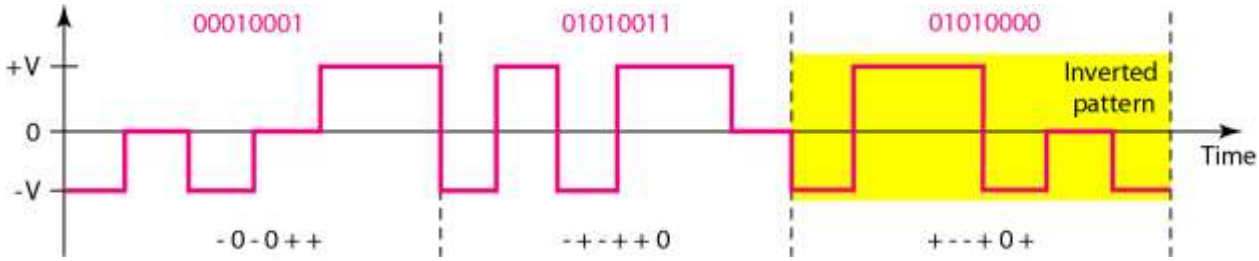
Next bits	Previous level: positive — Next level	Previous level: negative — Next level
00	+1	-1
01	+3	-3
10	-1	+1
11	-3	+3

Transition table

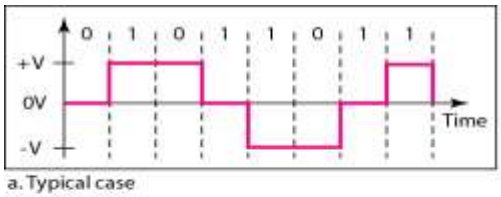




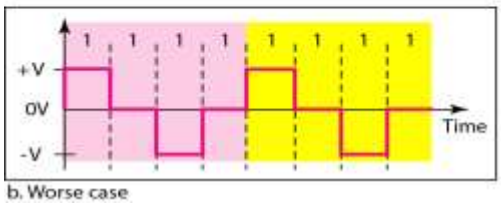
Multilevel: 8B6T scheme



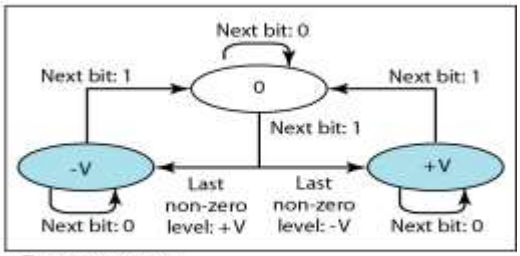
Multitransition: MLT-3 scheme



a. Typical case



b. Worse case



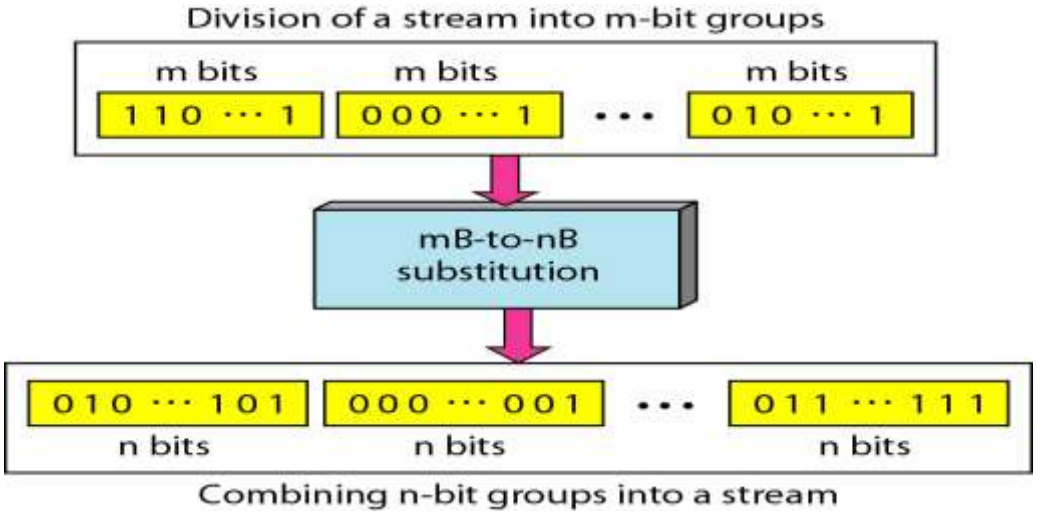
c. Transition states

Block coding concept

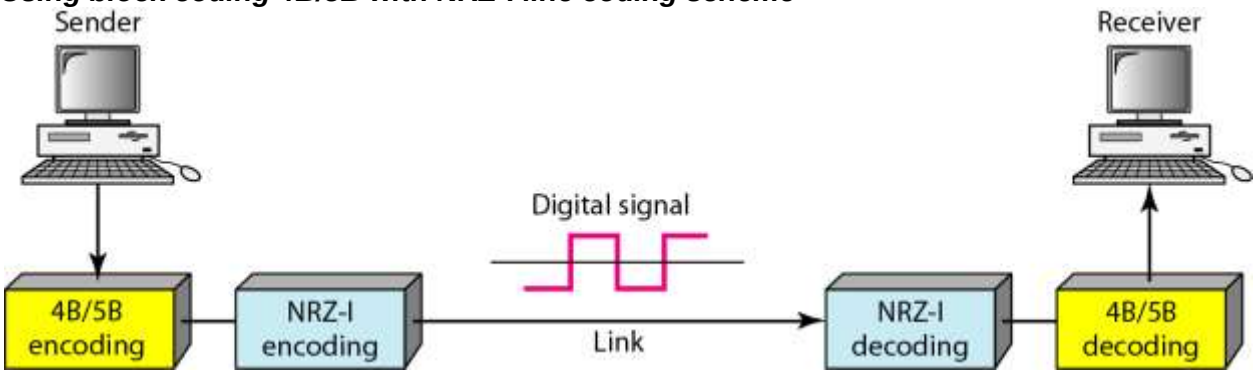
Block coding is normally referred to as mB/nB coding; it replaces each m-bit group with an n-bit group.

Block coding normally involves three steps:

- ❖ Division : In the division step, a sequence of bits is divided into groups of m bits. For example, in 4B/5B encoding, the original bit sequence is divided into 4-bit groups.
- ❖ Substitution: In substitution step, we substitute an m-bit group for an n-bit group. For example, in 4B/5B encoding we substitute a 4-bit code for a 5-bit group.
- ❖ Combination: The n-bit groups are combined together to form a stream. The new stream has more bits than the original bits.



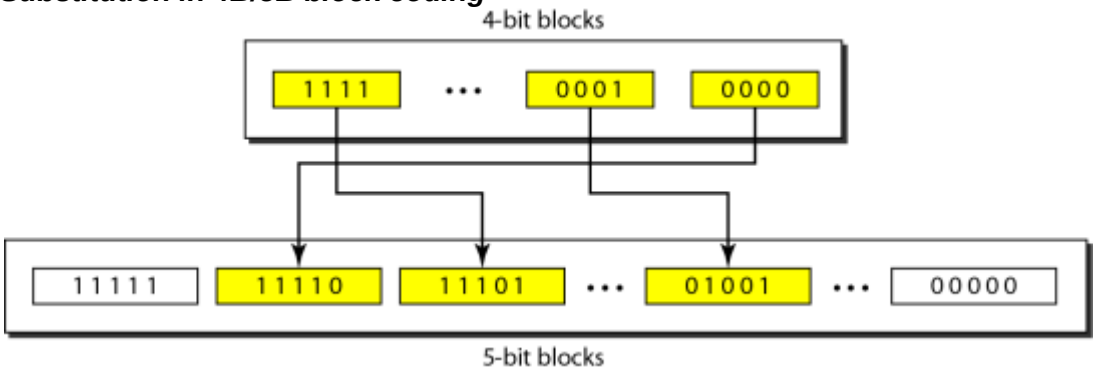
Using block coding 4B/5B with NRZ-I line coding scheme



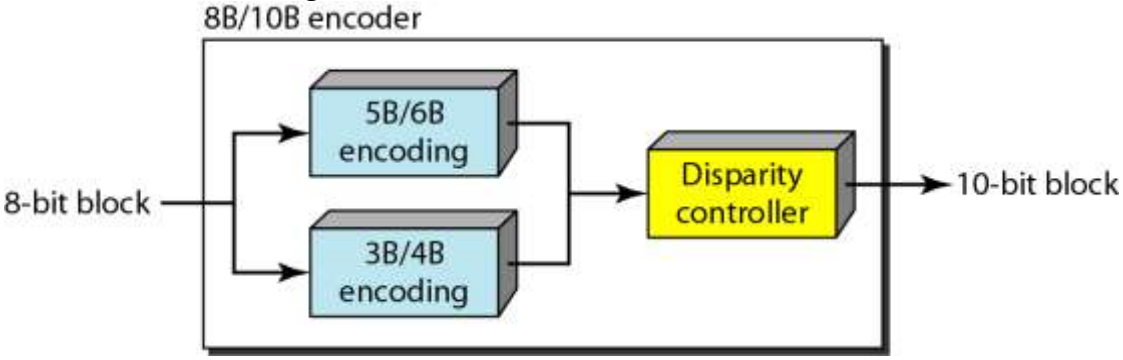
4B/5B Mapping Codes:

Data Sequence	Encoded Sequence	Control Sequence	Encoded Sequence
0000	11110	Q (Quiet)	00000
0001	01001	I (Idle)	11111
0010	10100	H (Halt)	00100
0011	10101	J (Start delimiter)	11000
0100	01010	K (Start delimiter)	10001
0101	01011	T (End delimiter)	01101
0110	01110	S (Set)	11001
0111	01111	R (Reset)	00111
1000	10010		
1001	10011		
1010	10110		
1011	10111		
1100	11010		
1101	11011		
1110	11100		
1111	11101		

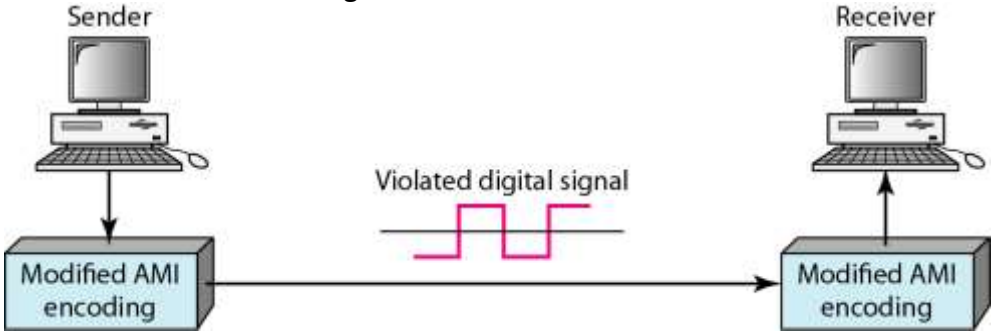
Substitution in 4B/5B block coding



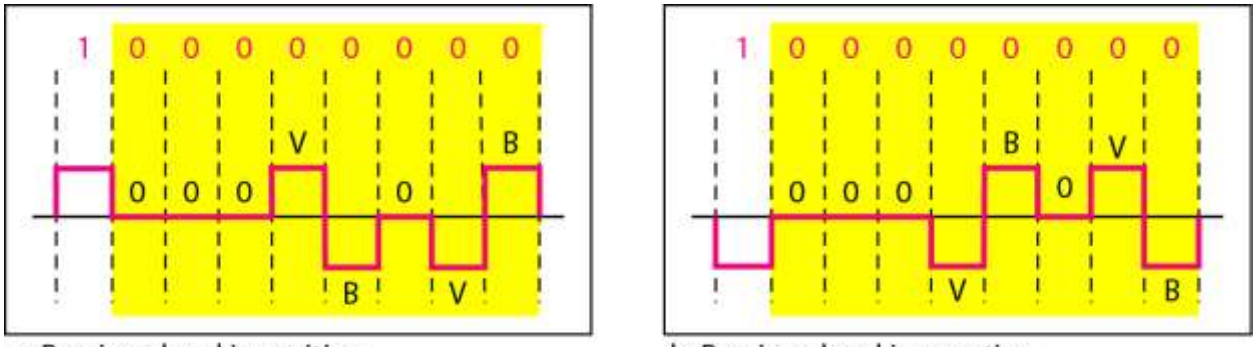
8B/10B block encoding



AMI used with scrambling



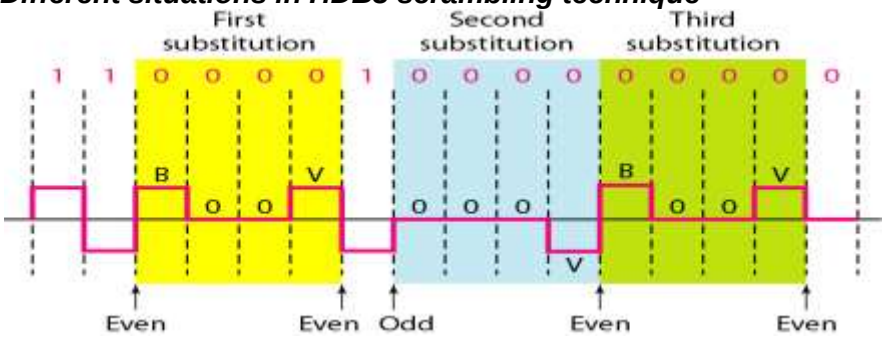
Two cases of B8ZS scrambling technique



a. Previous level is positive. b. Previous level is negative.

B8ZS substitutes eight consecutive zeros with 000VB0VB.

Different situations in HDB3 scrambling technique



HDB3 substitutes four consecutive zeros with 000V or B00V depending on the number of nonzero pulses after the last substitution.

ANALOG-TO-DIGITAL CONVERSION

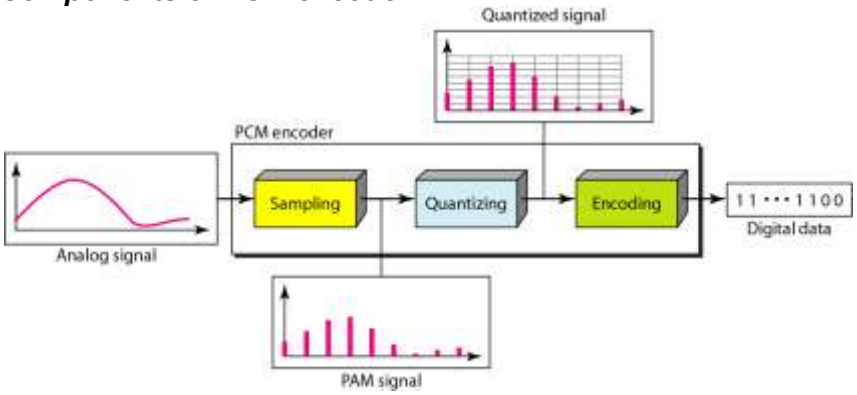
- A digital signal is superior to an analog signal.
- The tendency today is to change an analog signal to digital data.
- In this section we describe two techniques, pulse code modulation and delta modulation.

PCM:

The most common technique to change an analog signal to digital data (digitization) is called pulse code modulation (PCM). A PCM encoder has three processes.

1. The analog signal is sampled.
2. The sampled signal is quantized.
3. The quantized values are encoded as streams of bits.

Components of PCM encoder



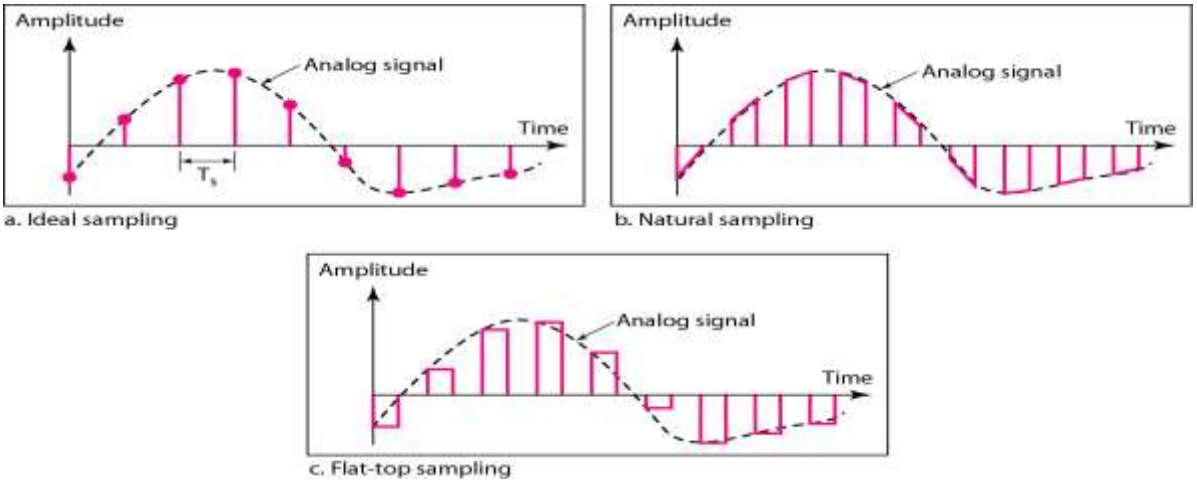
The process of Pulse code modulation consists of the following steps:

- Sampling: measuring the amplitude of the signal at equal intervals.

- ❖ Quantization: Process of assigning integral values in a specific range to the long range of sampled instances.
- ❖ Binary Coding: To bring into binary form
- ❖ Line Coding: conversion of binary data into digital signal.

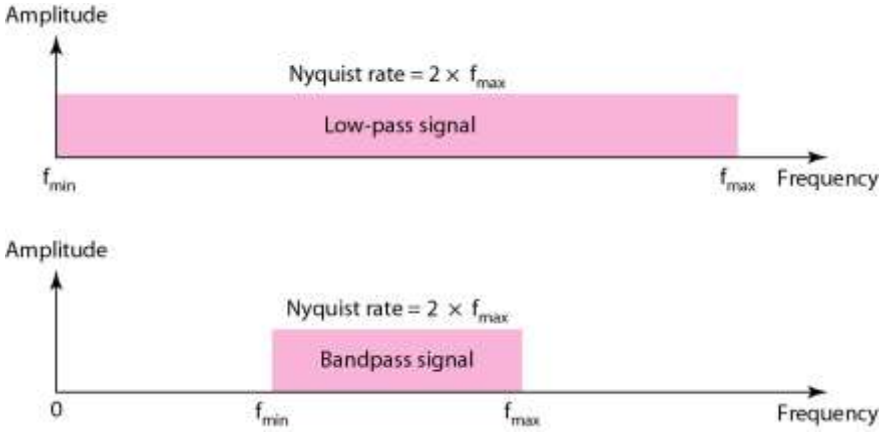
There are three sampling methods: ideal, natural, and flat-top.

- ❖ In ideal sampling, pulses from the analog signal are sampled. This is an ideal sampling method and cannot be easily implemented.
- ❖ In natural sampling, a high-speed switch is turned on for only the small period of time when the sampling occurs. The result is a sequence of samples that retains the shape of the analog signal.
- ❖ The most common sampling method, called sample and hold, however, creates flat-top samples by using a circuit.



Nyquist sampling rate for low-pass and band-pass signals

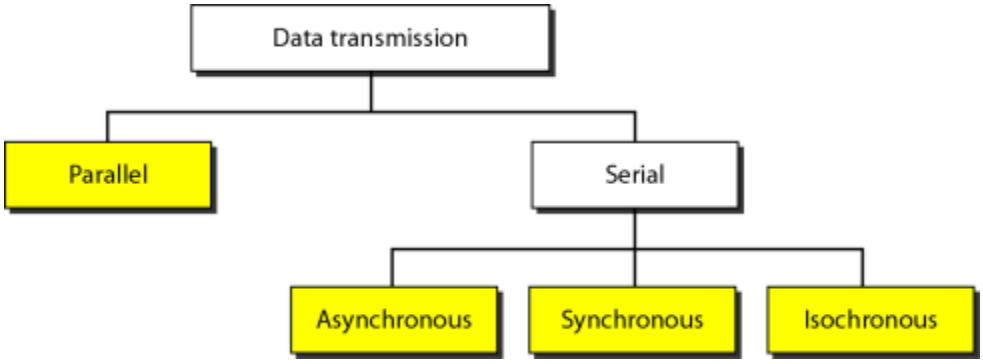
According to the Nyquist theorem, the sampling rate must be at least 2 times the highest frequency contained in the signal.



TRANSMISSION MODES

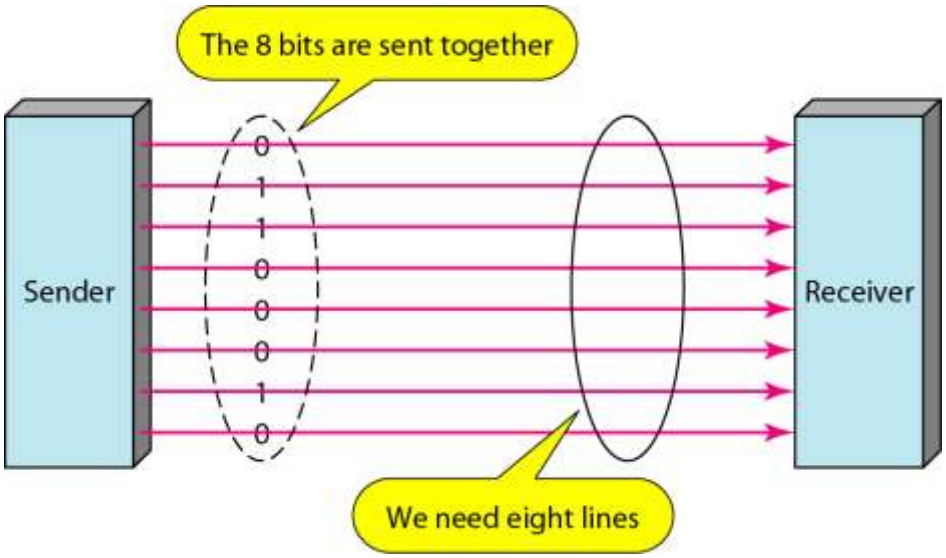
- ❖ The transmission of binary data across a link can be accomplished in either parallel or serial mode.
- ❖ In parallel mode, multiple bits are sent with each clock tick.
- ❖ In serial mode, 1 bit is sent with each clock tick.
- ❖ While there is only one way to send parallel data, there are three subclasses of serial transmission: *asynchronous, synchronous,* and *isochronous*.

Data transmission and modes

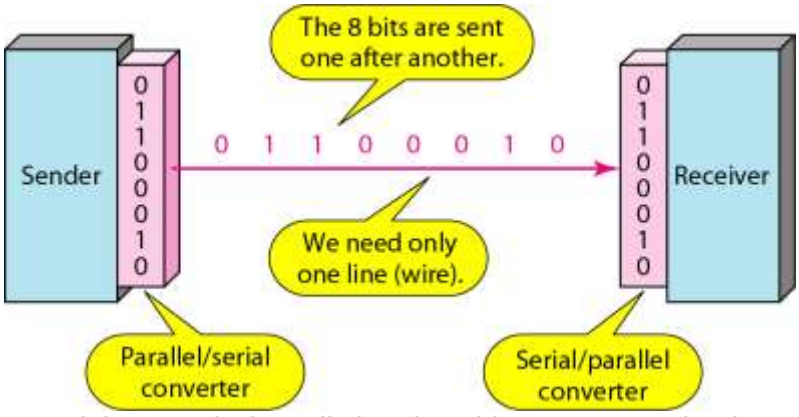


Parallel transmission

The advantage of parallel transmission is speed. All else being equal, parallel transmission can increase the transfer speed by a factor of n over serial transmission. But there is a significant disadvantage: cost. Parallel transmission requires n communication lines (wires in the example) just to transmit the data stream. Because this is expensive, parallel transmission is usually limited to short distances.



Serial transmission

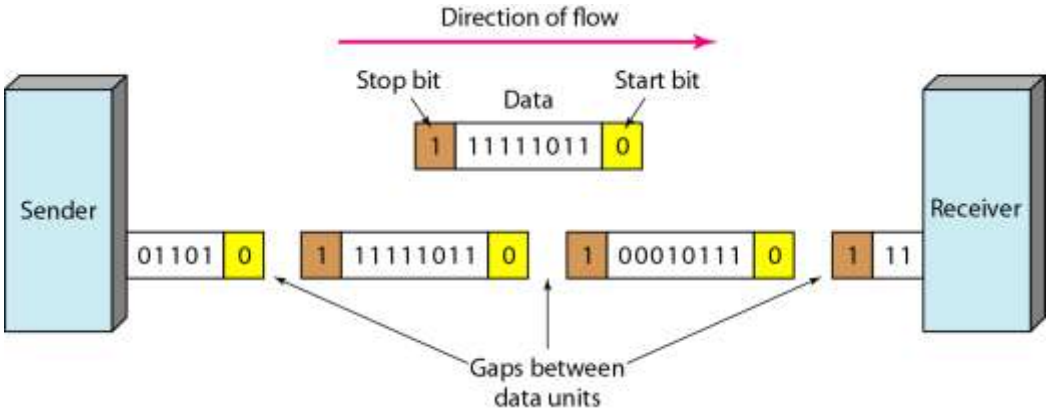


In serial transmission all the data bits are transmitted across a single wire in sequence. The advantage of serial over parallel transmission is that with only one communication channel, serial transmission reduces the cost of transmission over parallel by roughly a factor of n.

Asynchronous transmission

Asynchronous transmission is so named because the timing of a signal is unimportant. We send 1 start bit (0) at the beginning and 1 or more stop bits (1s) at the end of each byte.

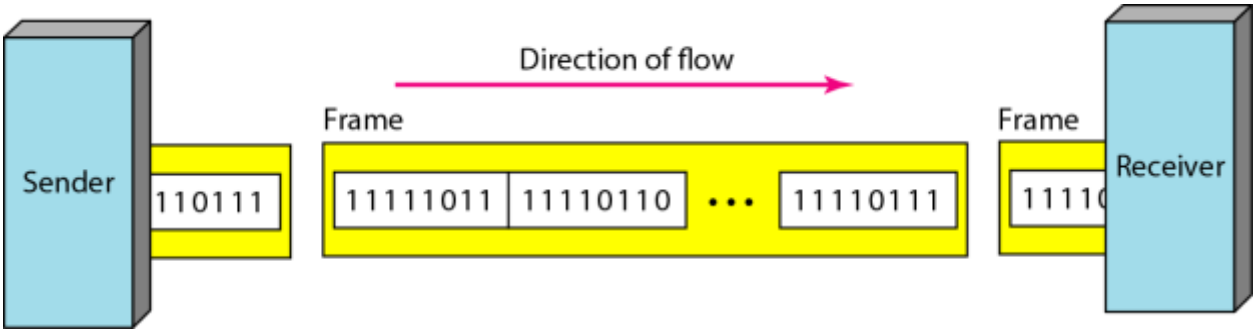
There may be a gap between each byte.



It is "asynchronous at the byte level," bits are still synchronized; their durations are the same. The addition of stop and start bits and the insertion of gaps into the bit stream make asynchronous transmission slower than forms of transmission that can operate without the addition of control information. But it is cheap and effective, two advantages that make it an attractive choice for situations such as low-speed communication.

Synchronous transmission

In synchronous transmission, we send bits one after another without start or stop bits or gaps. It is the responsibility of the receiver to group the bits. ***Timing*** becomes very important, therefore, because the accuracy of the received information is completely dependent on the ability of the receiving device to keep an accurate count of the bits as they come in. ***The advantage*** of synchronous transmission is speed. With no extra bits or gaps, synchronous transmission is faster than asynchronous transmission.



Isochronous

In real-time audio and video, in which uneven delays between frames are not acceptable, synchronous transmission fails. For example, TV images are broadcast at the rate of 30 images per second; they must be viewed at the same rate. If each image is sent by using one or more frames, there should be no delays between frames. For this type of application, synchronization between characters is not enough; the entire stream of bits must be synchronized. The isochronous transmission guarantees that the data arrive at a fixed rate.

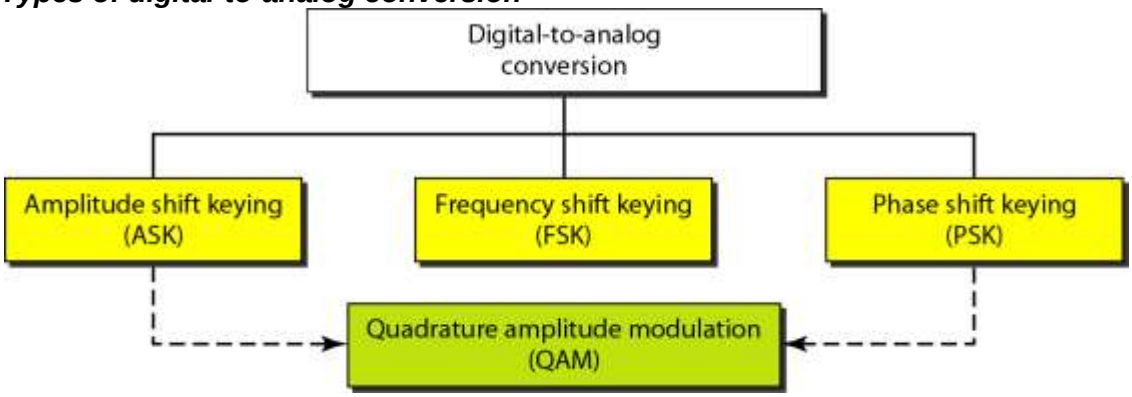
Analog Transmission

DIGITAL-TO-ANALOG CONVERSION

Digital-to-Analog conversion is the process of changing one of the characteristics of an analog signal based on the information in digital data.

- Digital data needs to be carried on an analog signal.
- A carrier signal (frequency f_c) performs the function of transporting the digital data in an analog waveform.
- The analog carrier signal is manipulated to uniquely identify the digital data being carried

Types of digital-to-analog conversion



- Bit rate is the number of bits per second.
- Baud rate is the number of signal elements per second.
- In the analog transmission of digital data, the baud rate is less than or equal to the bit rate.

Example

An analog signal carries 4 bits per signal element. If 1000 signal elements are sent per second, find the bit rate.

Solution

In this case, r = 4, S = 1000, and N is unknown. We can find the value of N from

$$S = N \times \frac{1}{r} \quad \text{or} \quad N = S \times r = 1000 \times 4 = 4000 \text{ bps}$$

Q. An analog signal has a bit rate of 8000 bps and a baud rate of 1000 baud.

- How many data elements are carried by each signal element?
- How many signal elements do we need?

Solution

S = 1000, N = 8000, and r and L are unknown.

We find first the value of r and then the value of L.

$$S = N \times \frac{1}{r} \rightarrow r = \frac{N}{S} = \frac{8000}{1000} = 8 \text{ bits/baud}$$

$$r = \log_2 L \rightarrow L = 2^r = 2^8 = 256$$

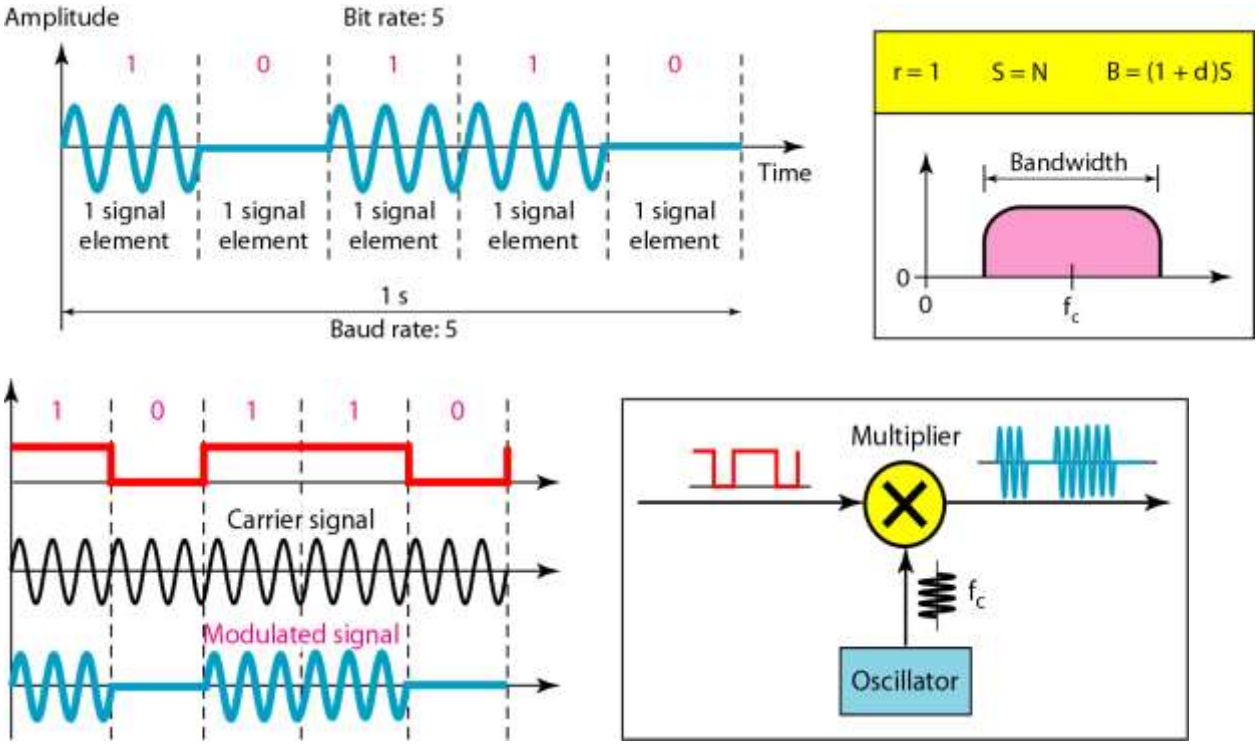
Amplitude Shift Keying (ASK)

- ASK is implemented by changing the amplitude of a carrier signal to reflect amplitude levels in the digital signal.
- For example: a digital “1” could not affect the signal, whereas a digital “0” would, by making it zero.
- The line encoding will determine the values of the analog waveform to reflect the digital data being carried.

Bandwidth of ASK

- The bandwidth B of ASK is proportional to the signal rate S.
  B = (1+d)S
- "d" is due to modulation and filtering, lies between 0 and 1.

Binary amplitude shift keying



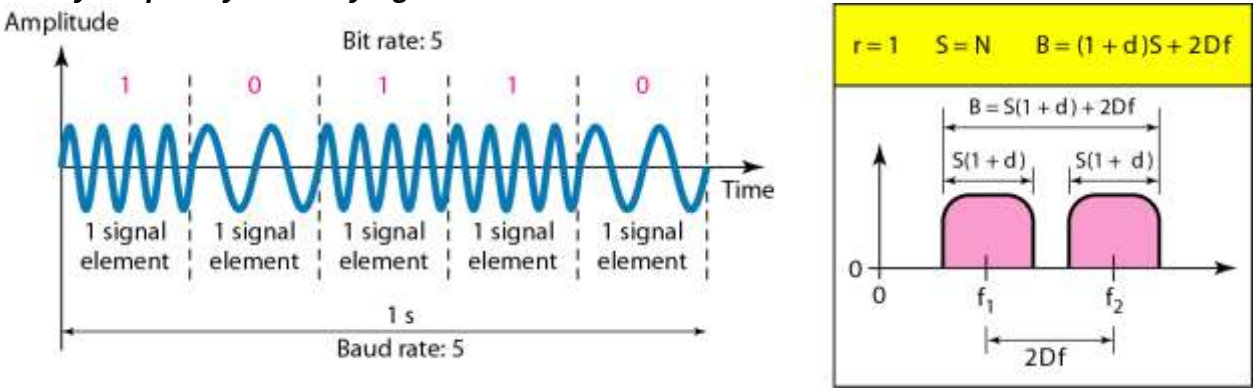
Frequency Shift Keying

- The digital data stream changes the frequency of the carrier signal, f_c.
- For example, a "1" could be represented by $f_1=f_c+\Delta f$, and a "0" could be represented by $f_2=f_c-\Delta f$.

Bandwidth of FSK

- If the difference between the two frequencies (f_1 and f_2) is $2\Delta f$, then the required BW B will be: $B = (1+d)xS + 2\Delta f$

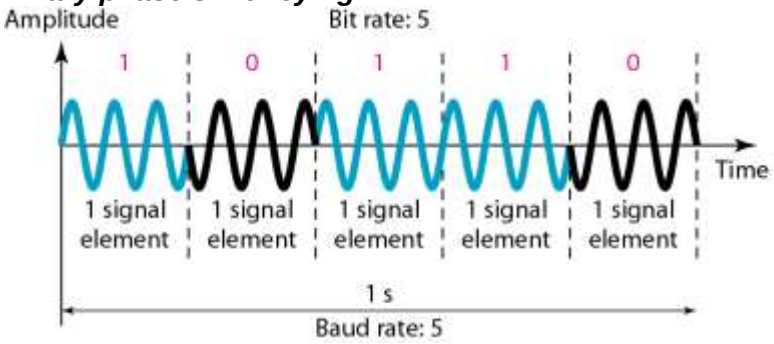
Binary frequency shift keying

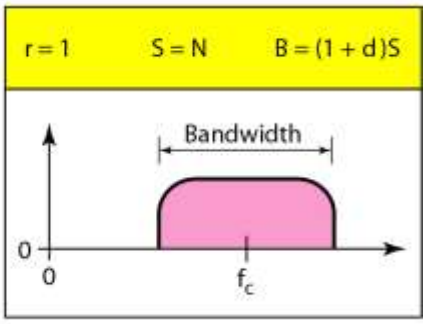


Phase Shift Keying

- We vary the phase shift of the carrier signal to represent digital data.
- The bandwidth requirement, B is: B = (1+d)xS
- PSK is much more robust than ASK as it is not that vulnerable to noise, which changes amplitude of the signal.

Binary phase shift keying





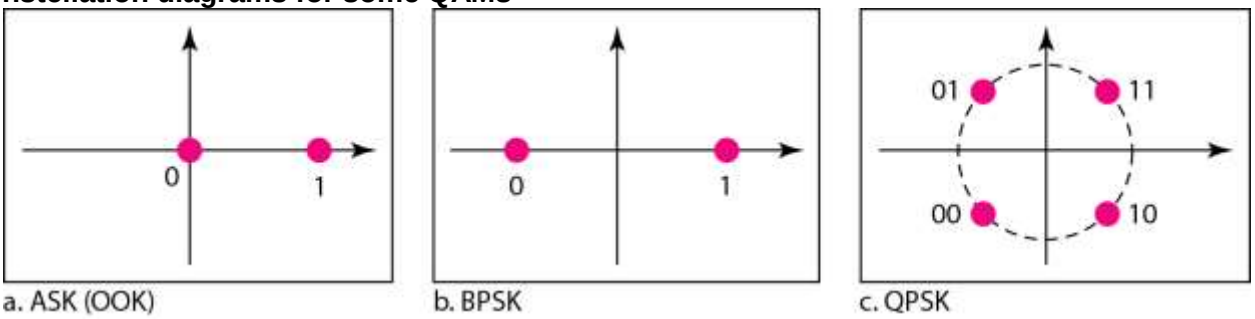
Quadrature PSK

- To increase the bit rate, we can code 2 or more bits onto one signal element.
- In QPSK, we parallelize the bit stream so that every two incoming bits are split up and PSK a carrier frequency. One carrier frequency is phase shifted 90^o from the other - in Quadrature.
- The two PSKed signals are then added to produce one of 4 signal elements. L = 4 here.
- Quadrature amplitude modulation is a combination of ASK and PSK.

Constellation Diagrams

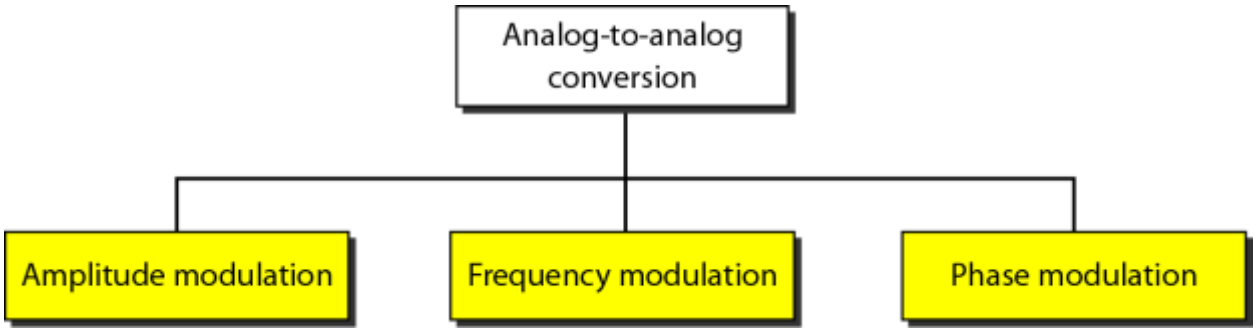
- A constellation diagram helps us to define the amplitude and phase of a signal when we are using two carriers, one in quadrature of the other.
- The X-axis represents the in-phase carrier and the Y-axis represents quadrature carrier.

Constellation diagrams for some QAMs

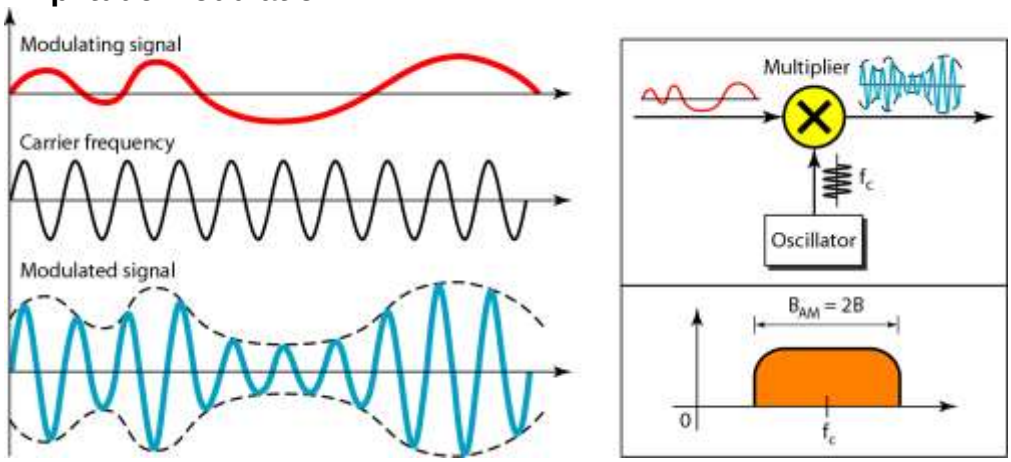


ANALOG-TO-ANALOG CONVERSION

- Analog-to-analog conversion is the representation of analog information by an analog signal.
- One may ask why we need to modulate an analog signal; it is already analog.
- Modulation is needed if the medium is band-pass in nature or if only a band-pass channel is available to us.

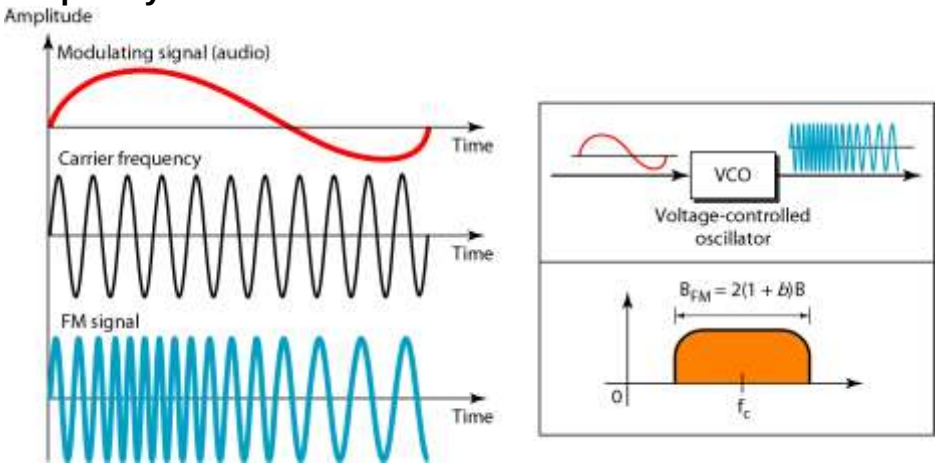


Amplitude Modulation



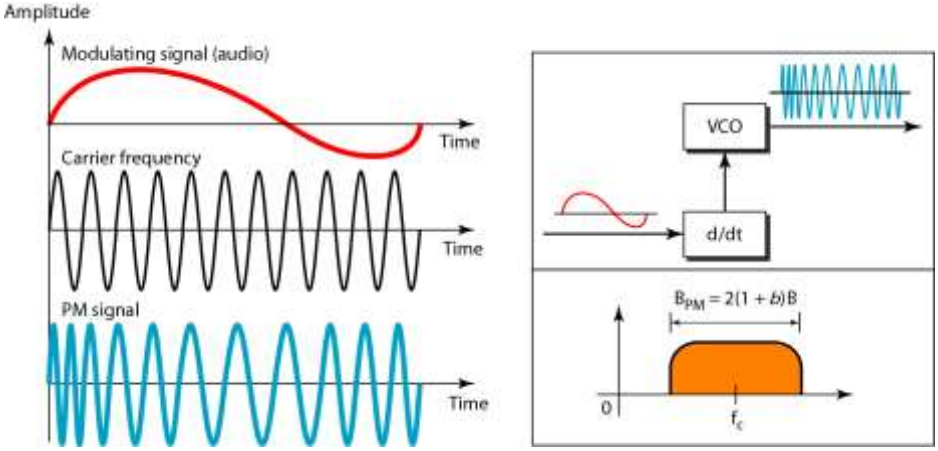
The total bandwidth required for AM can be determined from the bandwidth of the audio signal: $B_{AM} = 2B$.

Frequency Modulation



The total bandwidth required for FM can be determined from the bandwidth of the audio signal: $B_{FM} = 2(1 + \beta)B$.

Phase Modulation



The total bandwidth required for PM can be determined from the bandwidth and maximum amplitude of the modulating signal: $B_{PM} = 2(1 + \beta)B$.

TELEPHONE MODEM

MODEMS

The term modem is a composite word that refers to the two functional entities that make up the device; a signal modulator and a signal demodulator. A modulator creates a band-pass analog signal from binary data. A demodulator recovers the binary data from the modulated signal.
Modem stands for modulator and demodulator.

TELEPHONE MODEMS

Traditional telephone lines can carry frequencies between 300 and 3300 HZ, giving them BW of 3000 Hz; All this range is used for transmitting voice, where a great deal of interference and distortion can be accepted without loss of intelligibility.

The effective BW of a telephone line being used for data Transmission is 2400 Hz, covering the range from 600 to 3000 Hz.

Modem standards:

V-series standards published by the ITU-T.

- V.32
- V.32bis
- V.34bis
- V.90
- V.92

V.32

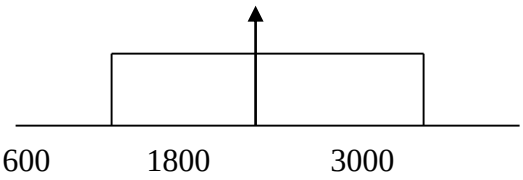
This modem uses a combined modulation and demodulation encoding technique called trellis-coded modulation. Trellis is essentially QAM plus a redundant bit. The Data stream is divided into 4-bit sections. Instead of a quad bit, however, a pentabit is transmitted. The value of the extra bit is calculated from the values of the data bits.

In any QAM system, the receiver compares each received signal point to all valid points in the constellation and selects the closest point as the intended value.. A signal distorted by transmission noise can arrive closer in value to an adjacent point than to the intended point, resulting in a misidentification of the point and an error in the received data.

By adding a redundant bit to each quad bit, trellis-coded modulation increases the amount of information used to identify each bit pattern thereby reduces the number of possible matches.

The V.32 calls for 32-QAM with a baud rate of 2400. Because only 4 bits of each pentabit represents data, the resulting speed is 4*2400=9600.

FDX 2400 baud 9600 bps 2-wire



Bandwidth diagram

V.32 bis

The V.32 bis modem support 14,400-bps transmission. The V.32 uses 128-QAM transmission.

V.34 bis

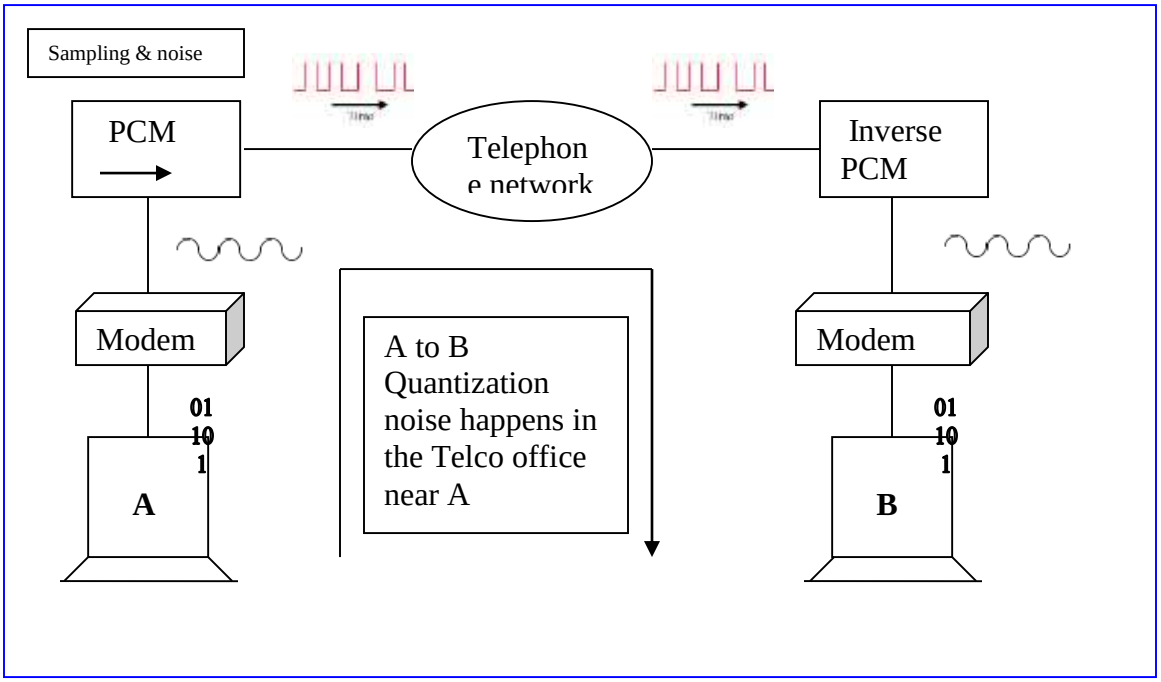
The V.34 bis modem support 28,800-bps transmission with a 960-point constellation to a bit rate of 33,600 with a 1664-point constellation.

V.90

Traditional modems have a limitations on the data rate.V.90 modems with a bit rate of 56,000 bps, called 56Kmodems, are available. Downloading rate is 56K, while the uploading rate is a maximum of 33.6 kbps.

Traditional modems

In traditional modems data exchange is between two computers, A and B, Through digital telephone network.



After modulation by the modem, an analog signal reaches the telephone company Switching station. Where it is sampled and digitized to be passed through the digital network. The quantization noise introduced in the signal at the sampling point limits the data rate according to the capacity. This limit is 33.6 Kbps.

56K Modems

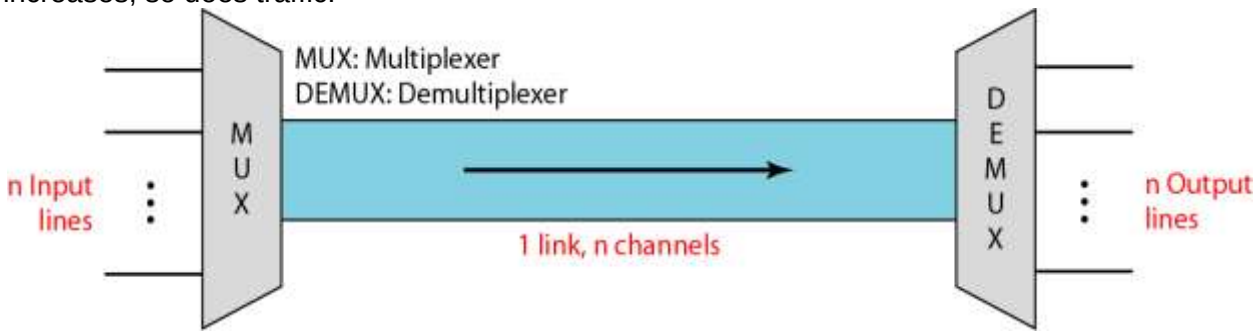
Communication today is via the Internet. In Uploading, The analog signal must still be sampled at the switching station, which means the data rate in the uploading is limited to 33.6 Kbps. There is no sampling in downloading. Data rate in downloading is 56Kbps.

V.92

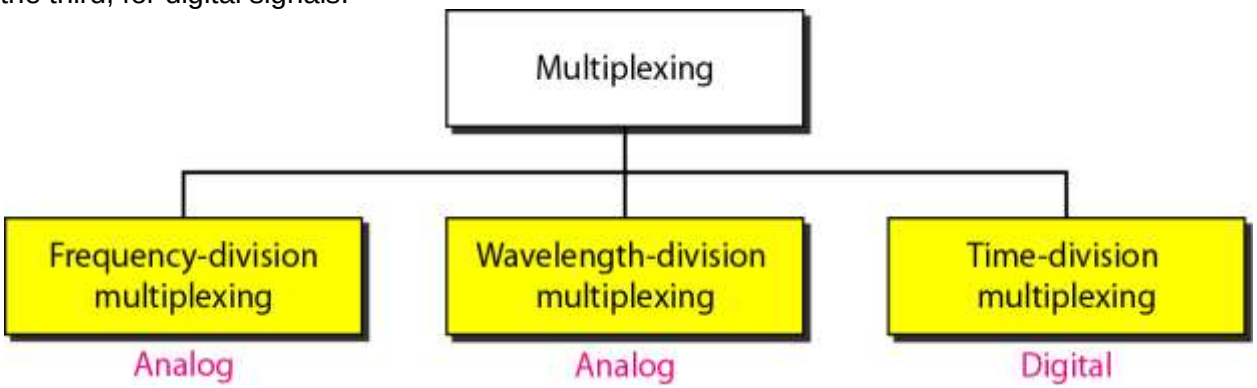
The standard above V.92 is called V.92. These modems can adjust their speed, and if the noise allows, they can upload data at the rate of 48 Kbps. The modem has additional features. For example, the modem can interrupt the internet connection when there is an incoming call if the lines has call-waiting service.

MULTIPLEXING

Whenever the bandwidth of a medium linking two devices is greater than the bandwidth needs of the devices, the link can be shared. Multiplexing is the set of techniques that allows the (simultaneous) transmission of multiple signals across a single data link. As data and telecommunications use increases, so does traffic.



There are three basic multiplexing techniques: frequency-division multiplexing, wavelength-division multiplexing, and time-division multiplexing. The first two are techniques designed for analog signals, the third, for digital signals.



In a multiplexed system, n lines share the bandwidth of one link. The lines on the left direct their transmission streams to a multiplexer (MUX), which combines them into a single stream (many-to-one).

At the receiving end, that stream is fed into a demultiplexer (DEMUX), which separates the stream back into its component transmissions (one-to-many) and directs them to their corresponding lines.

In the figure, the word ***link*** refers to the physical path. The word ***channel*** refers to the portion of a link that carries a transmission between a given pair of lines. One link can have many (n) channels.

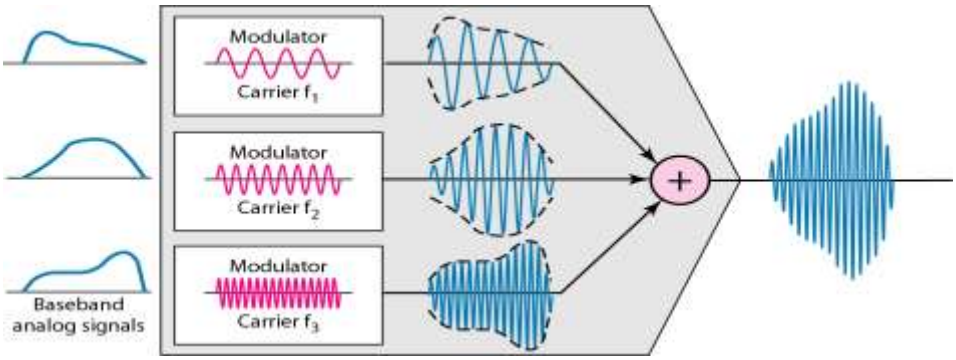
Frequency-division multiplexing (FDM):

It is an analog technique that can be applied when the bandwidth of a link (in hertz) is greater than the combined bandwidths of the signals to be transmitted.

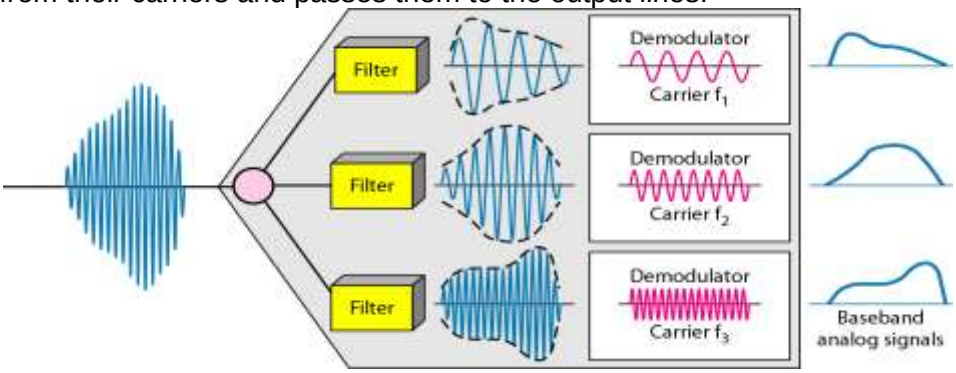
- In FDM, signals generated by each sending device modulate different carder frequencies. These modulated signals are then combined into a single composite signal that can be transported by the link.
- Carrier frequencies are separated by sufficient bandwidth to accommodate the modulated signal. These bandwidth ranges are the channels through which the various signals travel.

- Channels can be separated by strips of unused bandwidth **guard bands** to prevent signals from overlapping.

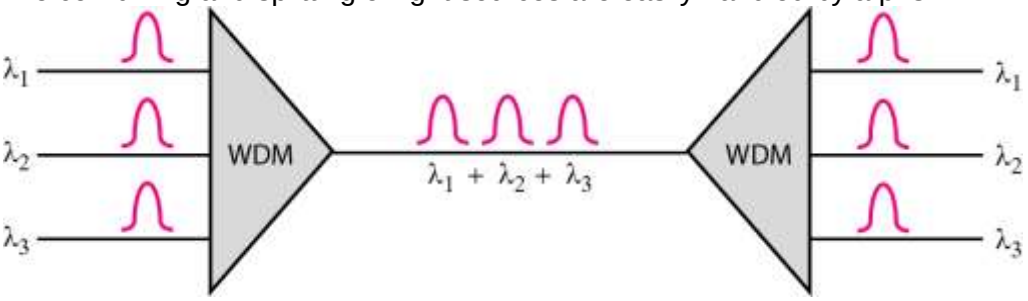
FDM Process:



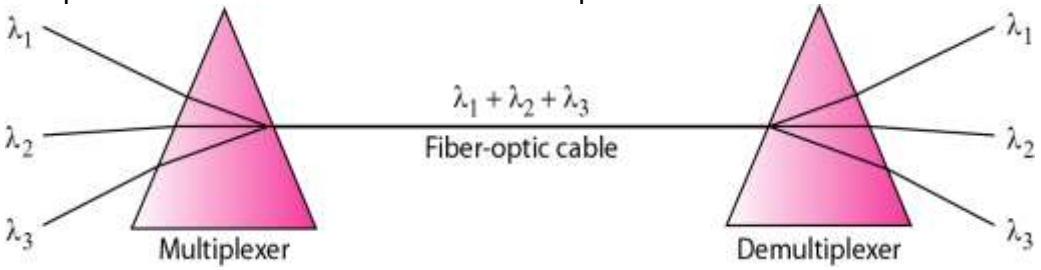
The de-multiplexer uses a series of filters to decompose the multiplexed signal into its constituent component signals. The individual signals are then passed to a demodulator that separates them from their carriers and passes them to the output lines.



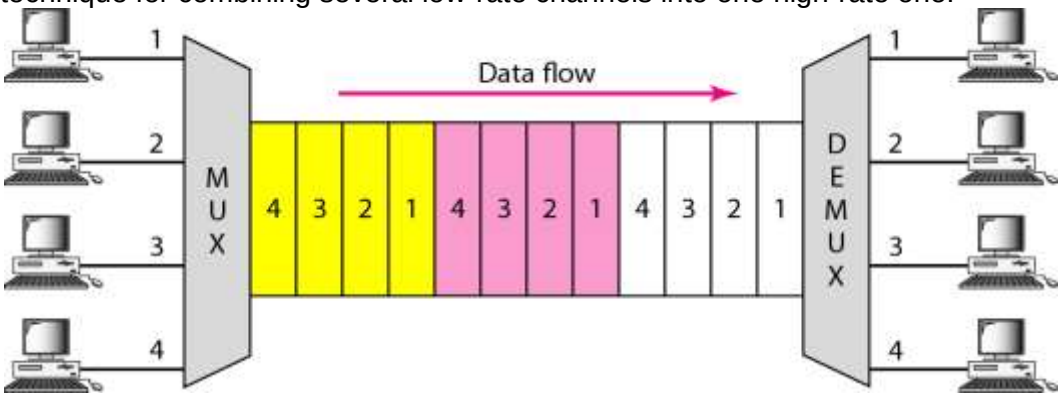
Wavelength-division multiplexing (WDM) is designed to use the high-data-rate capability of fiber-optic cable. The optical fiber data rate is higher than the data rate of metallic transmission cable. Using a fiber-optic cable for one single line wastes the available bandwidth. Multiplexing allows us to combine several lines into one. WDM is an analog multiplexing technique to combine optical signals. The combining and splitting of light sources are easily handled by a prism.



Using this technique, a multiplexer can be made to combine several input beams of light, each containing a narrow band of frequencies, into one output beam of a wider band of frequencies. A de-multiplexer can also be made to reverse the process.



Time-Division Multiplexing: Time-division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a link. Instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link. Note that the same link is used as in FDM; here, however, the link is shown sectioned by time rather than by frequency. In the figure, portions of signals 1, 2, 3, and 4 occupy the link sequentially. TDM is a digital multiplexing technique for combining several low-rate channels into one high-rate one.



We can divide TDM into two different schemes*:* synchronous and statistical.
In synchronous TDM, each input connection has an allotment in the output even if it is not sending data. In synchronous TDM, the data rate of the link is *n* times faster, and the unit duration is *n* times shorter.

Interleaving

- The process of taking a group of bits from each input line for multiplexing is called interleaving.
- We interleave bits (1 - n) from each input onto one output.

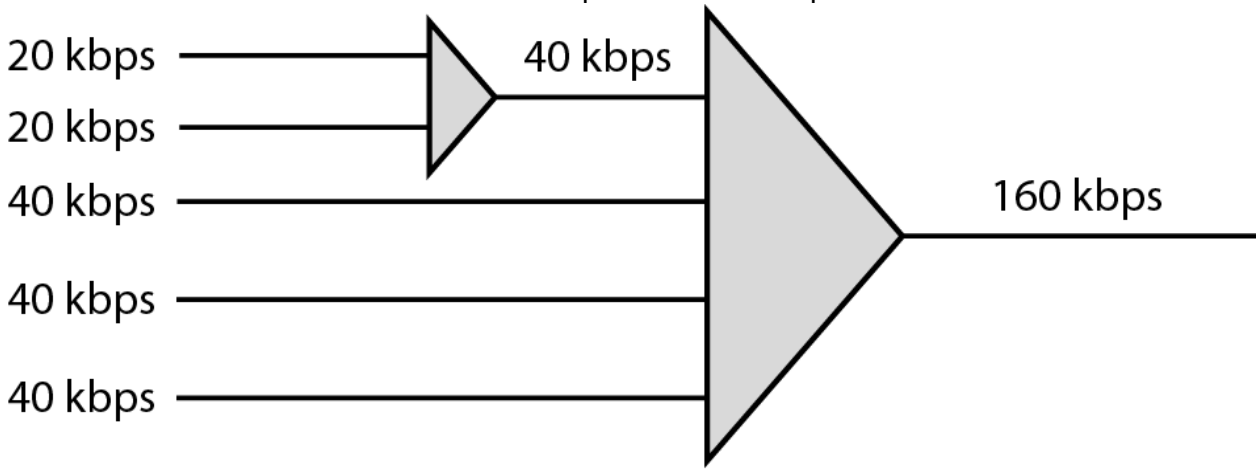
TDM can be visualized as two fast-rotating switches, one on the multiplexing side and the other on the demultiplexing side. The switches are synchronized and rotate at the same speed, but in opposite directions. On the multiplexing side, as the switch opens in front of a connection, that connection has the opportunity to send a unit onto the path. This process is called interleaving.

Empty Slots

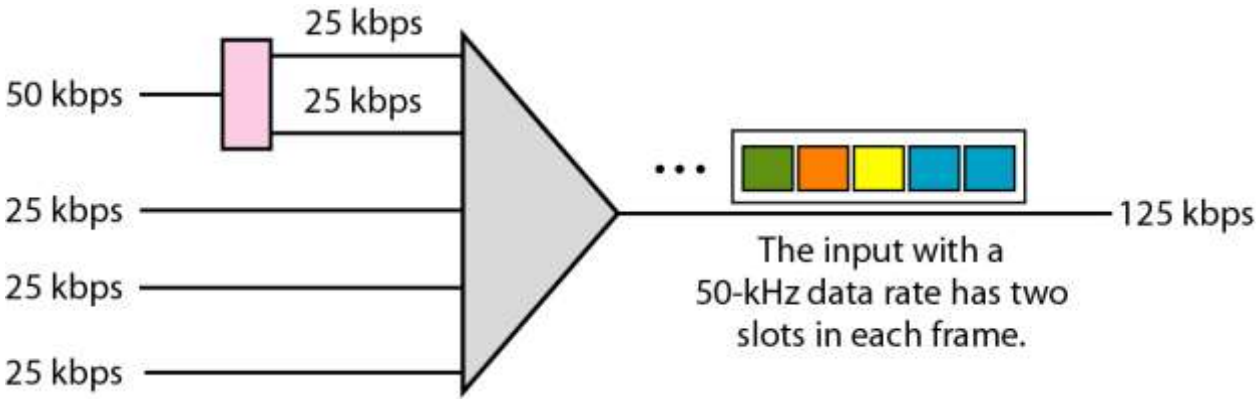
Synchronous TDM is not as efficient as it could be. If a source does not have data to send, the corresponding slot in the output frame is empty.

Data Rate Management:

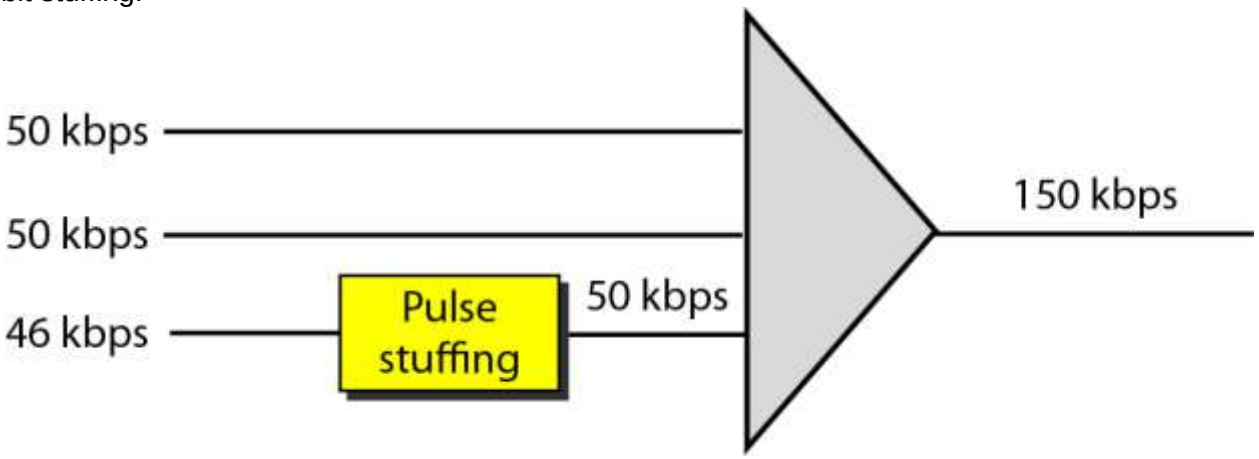
- There are three strategies that can be used to overcome the data rate mismatch: multilevel, multislot and pulse stuffing
  - **Multilevel:** used when the data rate of the input links are multiples of each other.



  - **Multislot:** used when there is a GCD between the data rates. The higher bit rate channels are allocated more slots per frame, and the output frame rate is a multiple of each input link.



- **Pulse Stuffing** Sometimes the bit rates of sources are not multiple integers of each other. Therefore, neither of the above two techniques can be applied. One solution is to make the highest input data rate the dominant data rate and then add dummy bits to the input lines with lower rates. This will increase their rates. This technique is called pulse stuffing, bit padding, or bit stuffing.



Statistical Time-Division Multiplexing: As we saw in the previous section, in synchronous TDM, each input has a reserved slot in the output frame. This can be inefficient if some input lines have no data to send. In statistical time-division multiplexing, slots are dynamically allocated to improve bandwidth efficiency. Only when an input line has a slot's worth of data to send is it given a slot in the output frame. In statistical multiplexing, the number of slots in each frame is less than the number of input lines. The multiplexer checks each input line in round- robin fashion; it allocates a slot for an input line if the line has data to send; otherwise, it skips the line and checks the next line.

Addressing: in statistical TDM, a slot needs to carry data as well as the address of the destination. In statistical multiplexing, there is no fixed relationship between the inputs and outputs because there are no preassigned or reserved slots.

Slot Size

Since a slot carries both data and an address in statistical TDM, the ratio of the data size to address size must be reasonable to make transmission efficient.

No Synchronization Bit

The frames in statistical TDM need not be synchronized, so we do not need synchronization bits.

Bandwidth

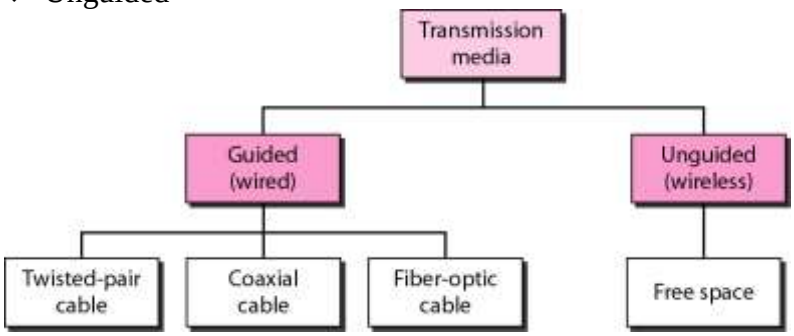
In statistical TDM, the capacity of the link is normally less than the sum of the capacities of each channel. The designers of statistical TDM define the capacity of the link based on the statistics of the load for each channel. If on average only x percent of the input slots are filled, the capacity of the link reflects this.

Transmission Media

A transmission media define as anything that can carry information from a source to a destination.
Transmission media are actually located below the physical layer and directly controlled by the physical layer.

Transmission media can be divided into two broad categories

- Guided
- Unguided



Guided media

Guided media, which are those that provide a conduit from one device to another, include twisted-pair cable, coaxial cable, and fiber-optic cable.

1. ***Twisted-Pair Cable***

A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together.

One of the wires is used to carry signals to the receiver, and the other is used only as a ground reference. The receiver uses the difference between the two. In addition to the signal sent by the sender on one of the wires, interference (noise) and crosstalk may affect both wires and create unwanted signals.

The most common twisted-pair cable used in communications is referred to as unshielded twisted-pair (UTP). IBM has also produced a version of twisted-pair cable for its use called shielded twisted-pair (STP). STP cable has a metal foil or braided- mesh covering that encases each pair of insulated conductors. Although metal casing improves the quality of cable by preventing the penetration of noise or crosstalk, it is bulkier and more expensive.

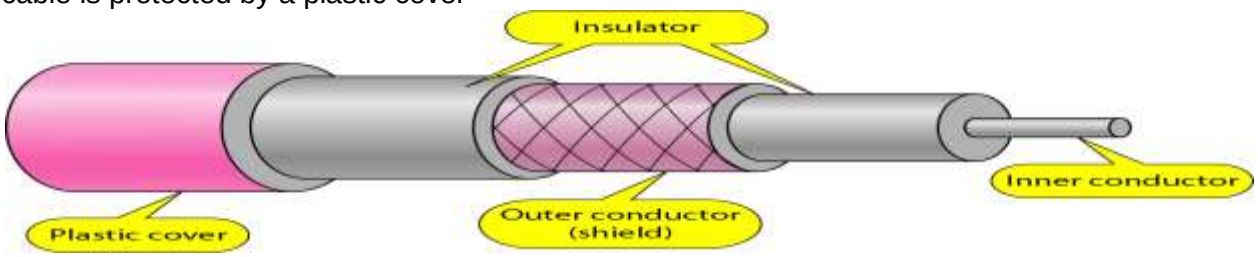
Connector: Registered Jack (RJ 45)

Applications

Twisted-pair cables are used in telephone lines to provide voice and data channels.

2. ***Coaxial Cable***

Coaxial cable (or coax) carries signals of higher frequency ranges than those in twisted- pair cable, in part because the two media are constructed quite differently. Instead of having two wires, coax has a central core conductor of solid or stranded wire (usually copper) enclosed in an insulating sheath, which is, in turn, encased in an outer conductor of metal foil, braid, or a combination of the two. The outer metallic wrapping serves both as a shield against noise and as the second conductor completing the circuit. This outer conductor is also enclosed in an insulating sheath, and the whole cable is protected by a plastic cover



Coaxial cable Standards

Coaxial cables are categorized by their radio government (RG) ratings .Each RG number denotes a set of physical specifications such as,

- wire gauge of the inner conductor
- thickness and type of the inner insulator
- the construction of the shield
- the size and type of outer casing

Categories of coaxial cables

Category	Impedance	Use
RG-59	75Ω	Cable TV
RG-58	50Ω	Thin Ethernet
RG-11	50Ω	Thick ethernet

Coaxial Cable Connectors

Coaxial Cable Connectors are used to connect coaxial cable to devices. The most common type of connector is the Bayone Neill-concelman or BNC connectors. There are three popular types of connectors

- BNC connector
- BNC T connector &
- BNC terminator

BNC connector

It is used to connect the end of the cable to a device such as a TV set.

BNC T connector

It is used in Ethernet networks to branch out a cable for connection to a computer or other devices.

BNC terminator

It is used at the end of the cable to prevent the reflection of the signal.

Performance

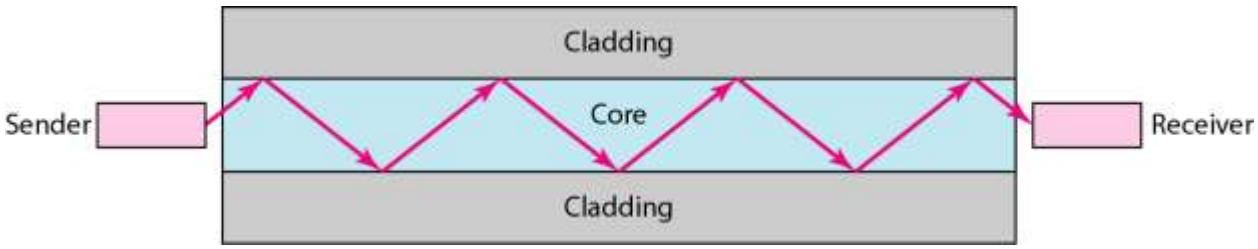
- Attenuation is much higher in coaxial cables than in twisted pair cable.
- Coaxial cable has a much higher bandwidth the signal weakens rapidly and needs the frequent use of repeaters.

Applications

- Coaxial cable is used in analog telephone network where a single coaxial cable could carry 10,000 voice signals.
- It is also used in digital telephone network where a cable could carry digital data up to 600 Mbps.
- Cable TV networks also used RG-59 coaxial cables.
- It is also used in traditional Ethernets.

Fiber Optic Cable.

A fiber optic cable is made of glass or plastic and transmits signals in the form of light. Optical fibers use reflection to guide light through a channel.



A glass or plastic core is surrounded by a cladding of less dense glass or plastic. The difference in the density of the two materials must be such that a beam of light moving through the core is reflected off the cladding.

Propagation Modes

There are two modes for propagating light along optical channels; each requires fiber with different physical characteristics

- Multimode
- Single mode

Multimode

Multiple beams from a light source move through the core in different paths.

Multimode can be implemented in two forms

- Step-index
- Graded index

Multimode Step –index fiber

- In Multimode Step –index fiber the density of the fiber remains constant from the center to the edges
- A beam of light moves through this constant density in a straight line.
- When it reaches the interface of the core and the cladding, there is an abrupt change to a lower density that alters the angle of the beams motion.
- Step-index -> the suddenness of this change.

Multimode Graded-index fiber

- It decreases the distortion of the signal through the cable.
- Density is highest at the center of the core and decreases gradually to its lowest at the edge.

Single-Mode

- It uses step-index fiber and a highly focused source of light that limits beams to a small range of angles, all close to the horizontal
- The Single-Mode fiber itself is manufactured with a smaller diameter than that of multimode fiber and with lower density.
- This results in a critical angle that is close enough to 90· to make it horizontal.
- All the beams arrive at the destination together and can be recombined with little distortion to the signal.

Fiber Sizes

Optical fibers are defined by the ratio of the diameter of their core to the diameter of their cladding expressed in micrometers.

Fiber-optic cable connectors.

Three different types of connectors are used by fiber –optic cable.

SC (subscriber channel) Connector:

It is used in cable TV.

ST (Straight-tip) Connector:

It is used for connecting cable to networking devices.

Performance:

- Attenuation is flatter than in the case of twisted pair cable and coaxial cable.
- Few repeaters are needed when we use fiber optic cable.

Application

It is used in cable TV and LAN (Fast Ethernet and 100Base –X.

Advantages

Higher bandwidth: It can support higher bandwidth than twisted pair or coaxial cable.

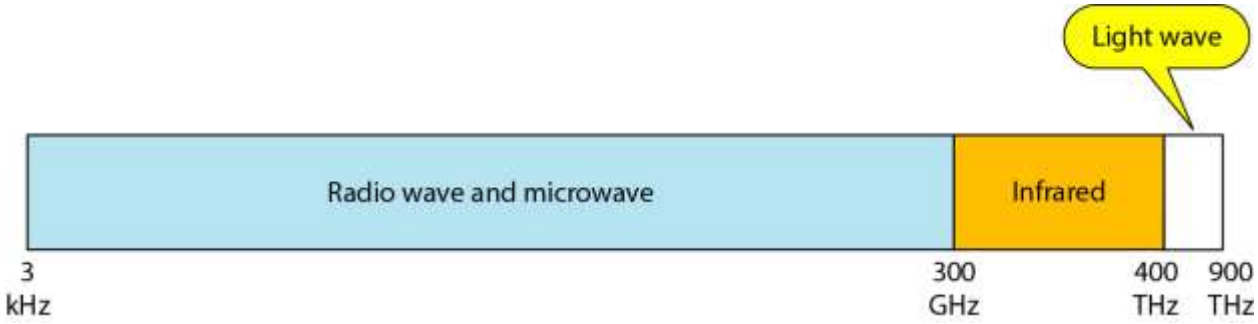
Less signal attenuation: Transmission distance is greater than that of other guided media. Signals can be transmitted for 50 km without requiring regeneration.

Immunity to electromagnetic Interference: Electromagnetic noise can not affect fiber-optic cables
Resistance to corrosive materials: glass is more resistant to corrosive materials.
Light-weight: It is of less weight than the copper cables.
More Immune to taping: Fiber-optic cables are more immune to taping than copper cables.

Disadvantages:
Installation/Maintenance. Installation/Maintenance need expertise since it is a new technology.
Unidirectional: Propagation of light is unidirectional. Bidirectional communication is achieved by means of two optical fibers.
Cost: It is more expensive and the use of optical fiber cannot be justified if the need for bandwidth is not high.

UNGUIDED MEDIA: WIRELESS

Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication.



Unguided signal can travel from the source to destination in several ways:

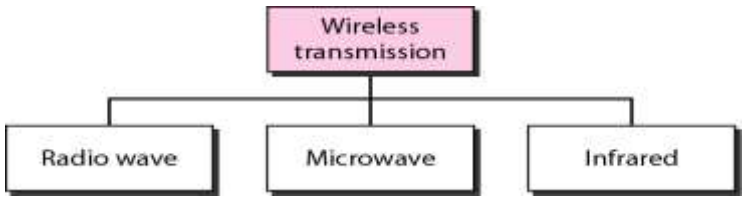
1. Ground Propagation:

- Radio waves travel through the lowest portion of the atmosphere, hugging the earth.
- The low frequency signal follows the curvature of the planet.
- Distance depends on the amount of the power.

2. Sky Propagation:

- Higher frequency radio radiate upward into the ionosphere where they arereflected back to the earth.
- Sky propagation allow for greater distance with lower power output.

3. line-of-sight Propagation: Very high frequency signals are transmitted in straight lines directly from antenna to antenna.



Radio Waves:

- Radio Waves: Between 3 KHz – 1 GHz.
- Radio waves use omnidirectional antenna.
- Radio waves used for multicast communication, such as radio and television.
- Sky Propagation. This makes radio waves a good candidate for long-distance broadcasting such as AM radio.

Micro Waves:

1. Microwave propagation is line-of-sight. Since the towers with the mounted antennas need to be in direct sight of each other, towers that are far apart need to be very tall.
2. Very high-frequency microwaves cannot penetrate walls. This characteristic can be a disadvantage if receivers are inside buildings.

3. The microwave band is relatively wide, almost 299 GHz. Therefore wider subbandscan be assigned, and a high data rate is possible Use of certain portions of the band requires permission from authorities.

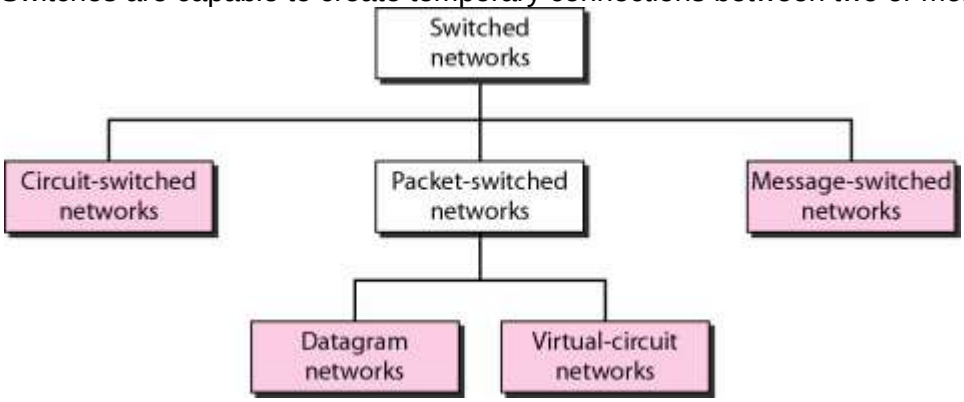
Infrared waves:

- Between 300 GHz-400 THz
- Used for short-range communication.
- Very common with remote control devices, but can also be used for device-to-device transfers, such as PDA to computer.
- Line-of-sight propagation.

CIRCUIT SWITCHING AND TELEPHONE NETWORK

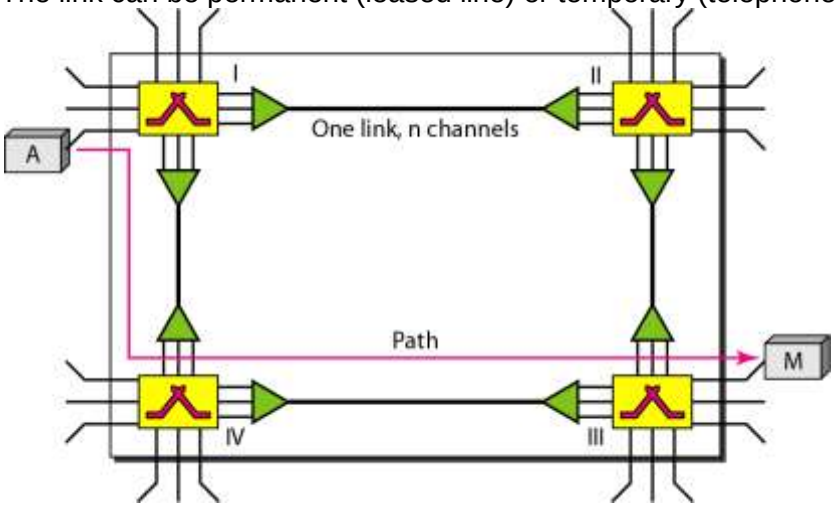
Introduction:

- None of the previous works in larger networks with large physical separation or consisting of a large number of computers
- The solution is a switching network.
- Consists of a series of interlinked nodes called switches.
- Switches are capable to create temporary connections between two or more devices

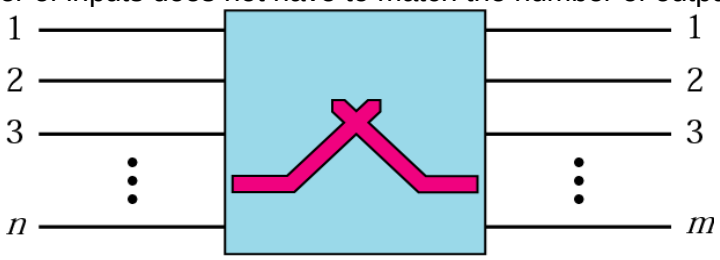


Circuit Switching:

- A circuit-switched network consists of a set of switches connected by physical links.
- A connection between two stations is a dedicated path made of one or more links
- each connection uses only one dedicated channel on each link
- Each link is normally divided into n channels by using FDM or TDM.
- The link can be permanent (leased line) or temporary (telephone)



- A circuit switch is a device with n inputs and m outputs that creates a temporary connection between an input link and an output link.
- The number of inputs does not have to match the number of outputs.

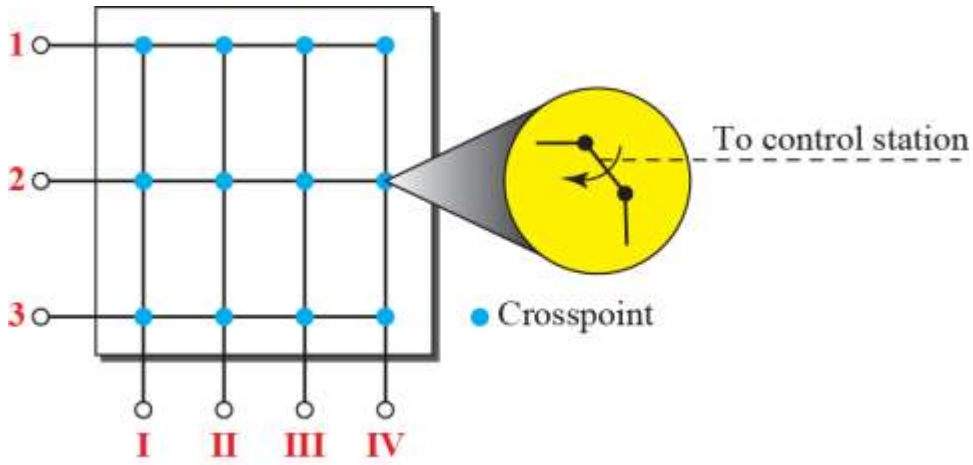


The common categories of switch are:

1. Space division Switch
2. Time division switch

Space division switch:

- ❖ Developed for analog environment, but has been carried over into digital communication
- ❖ Requires separate physical paths for each signal connection
- ❖ Uses metallic or semiconductor “gates”



- ❖ Its basic device is the Crossbar switch
- ❖ Number of crosspoints grows as square of number of stations
- ❖ Loss of crosspoint prevents connection
- ❖ Inefficient use of crosspoints
- ❖ All stations connected, only a few crosspoints in use
- ❖ Non-blocking

- ❖ paths in the circuit are separated from each other spatially.

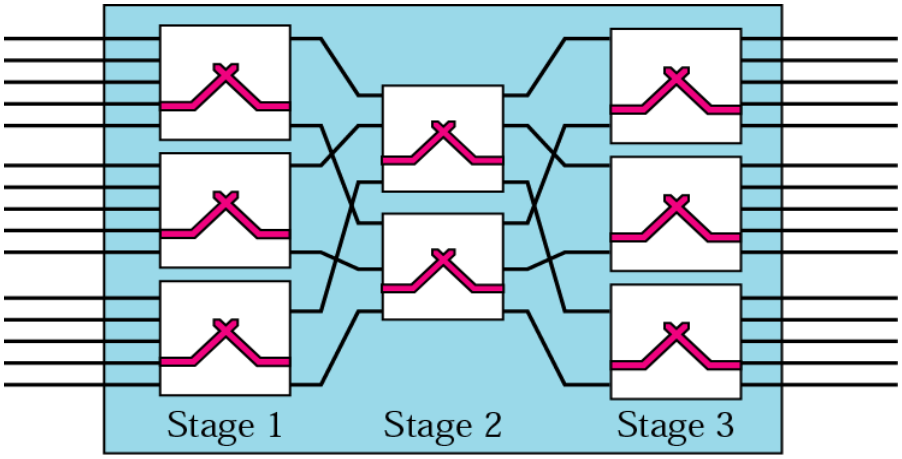
Crossbar Switch

- ❖ Crossbar switch connects n inputs to m outputs in a grid, using electronic micro-switches (transistors) at each cross-point.
- ❖ Limitation is the number of cross-points required.

- ❑ Advantages:
  - ❖ simple to implement
  - ❖ simple control
  - ❖ strict sense non-blocking
  - ❖ Multicast
  - ❖ Single source multiple destination ports
- ❑ Drawbacks
  - ❖ number of crosspoints, N^2
  - ❖ large VLSI space
  - ❖ vulnerable to single faults

Multistage switch:

- ■ Multistage switch combines crossbar switches in several stages.
- ■ Design of a multistage switch depends on the number of stages and the number of switches required (or desired) in each stage.
- ■ Normally, the middle stages have fewer switches than do the first and last stages.



- Multiple paths are available in multistage switches.
- Blocking refers to times when two inputs are looking for the same output. The output port is blocked.

Time-Division Switch

- Time-division switching uses time-division multiplexing to achieve switching. Two methods used are:
- Time-slot interchange (TSI) changes the order of the slots based on the desired connection.
- TDM bus

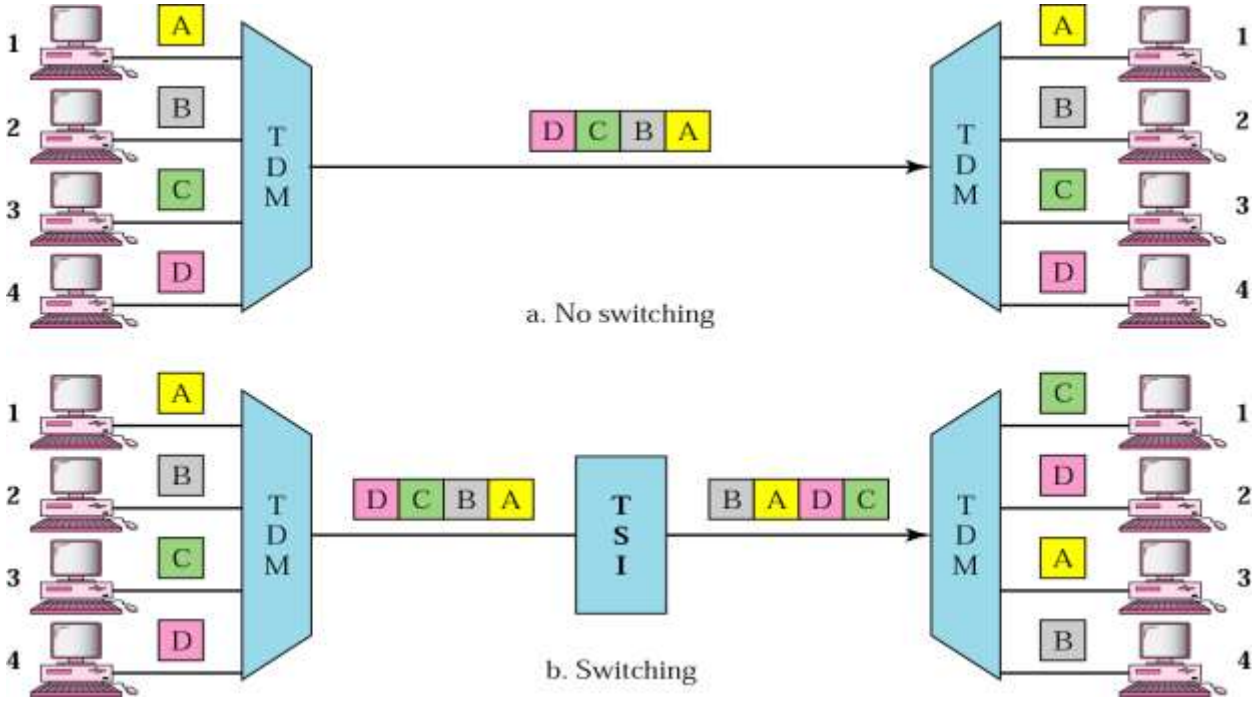
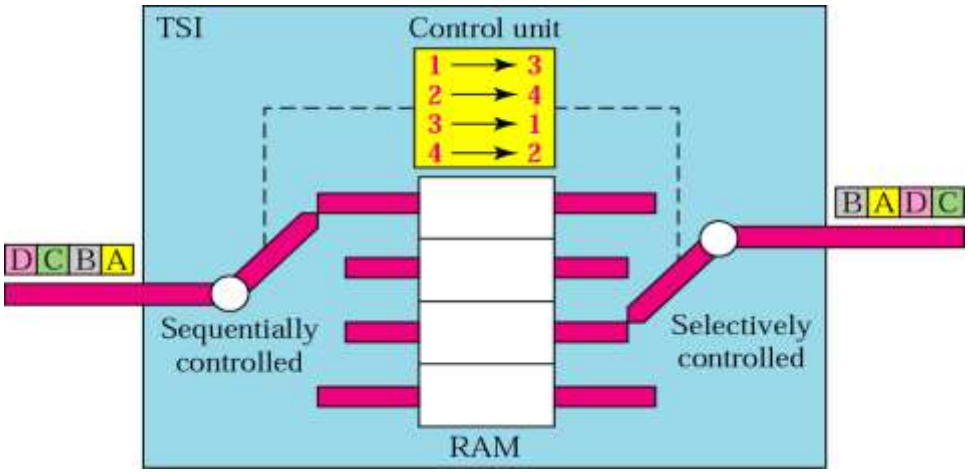


FIG: Time-division multiplexing, without and with a time-slot interchange

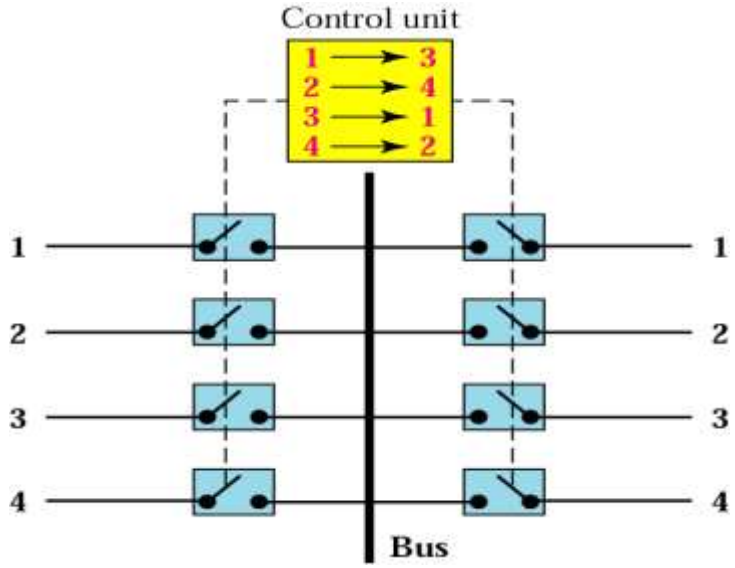
Time Slot Interchange (TSI)

- TSI consists of random access memory (RAM) with several memory locations. The size of each location is the same as the size of a single time slot.
- The number of locations is the same as the number of inputs.
- The RAM fills up with incoming data from time slots in the order received. Slots are then sent out in an order based on the decisions of a control unit.



TDM bus

- Input and output lines are connected to a high-speed bus through input and output gates (microswitches)
- Each input gate is closed during one of the four slots.
- During the same time slot, only one output gate is also closed. This pair of gates allows a burst of data to be transferred from one specific input line to one specific output line using the bus.
- The control unit opens and closes the gates according to switching need.

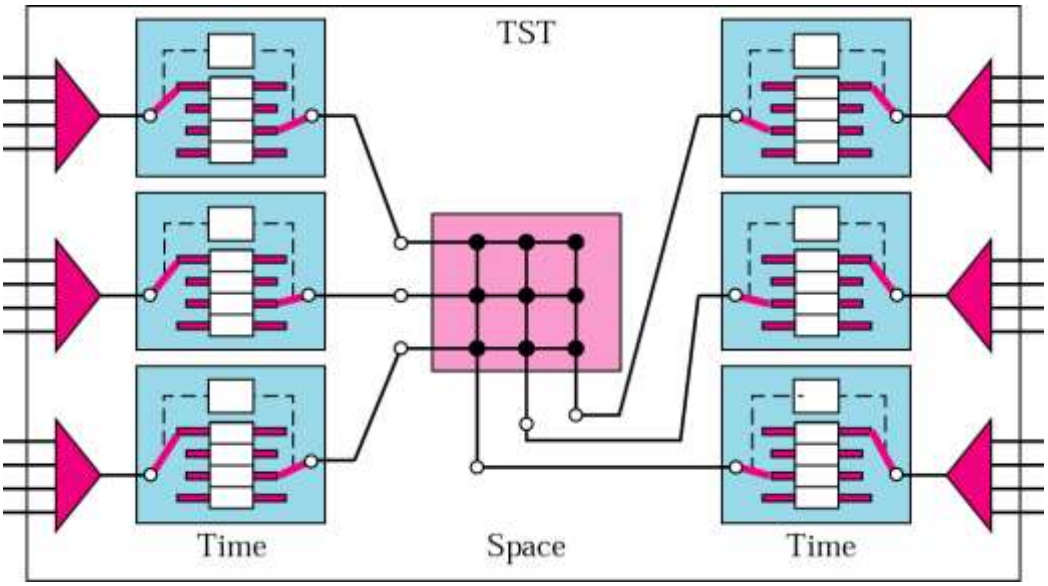


Comparison of SDM and TDM

- SDM
  - Advantage:
    - Instantaneous.
  - Disadvantage:
    - Number of cross points required.
- TDM
  - Advantage:
    - No cross points.
  - Disadvantage:
    - Processing delay.

TST switch

- Combine Space division and time division switching.
- This results in switches that are optimized both physically (the number of crosspoints) and temporally (the amount of delay).
- Various types are: time-space-time (TST), time-space-space-time (TSST), space-time-time-space (STTS), etc.



TELEPHONE NETWORK:

- Telephone networks use circuit switching.
- In 1800s, Plain old telephone system (POTS) was an analog system using analog signals to transmit voice.
- In 1980s, POTS started carrying data along with voice and also has become digital instead of analog.
- Major components of Telephone network: Local loops, trunk, and switching office.
- Different levels of switching offices: End offices, tandem offices, and regional offices.
- Local loop: Twisted pair cable that connects the subscriber telephone to the nearest end office or local central office. It has a bandwidth of 4000 Hz for voice. The first three digits of local telephone number define the office, and the next four digits define the local loop number.
- Trunks: Transmission media that handle communication between offices. It handles hundreds or thousands of connections through multiplexing. Transmission is usually through optical fibers or satellite links.
- Switching office: To avoid having a permanent physical link between any two subscribers, switches are located here. Switch connects several local loops or trunks and allows different subscribers to connect.

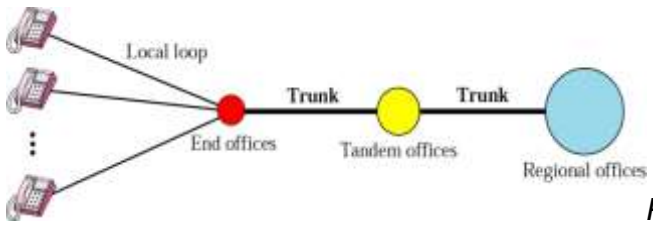


Fig: A Telephone Network

- LATA (Local access transport areas)
- Services offered by the common carriers (telephone companies) inside a LATA are called intra-LATA services. The carrier that handles these services is called a local exchange carrier (LEC).

- Intra-LATA services are provided by local exchange carriers. Since 1996, there are two types of LECs: incumbent local exchange carriers (ILEC) and competitive local exchange carriers (CLEC)
- ILEC would provide main services and owns the local loop. CLEC would provide other services such as mobile telephone service, toll calls inside a LATA, …
- Communication inside a LATA is handled by end switches and tandem switches. A call that can be completed by using only end offices is called toll-free. A call that has to go through a tandem office (intra-LATA toll office) is charged.

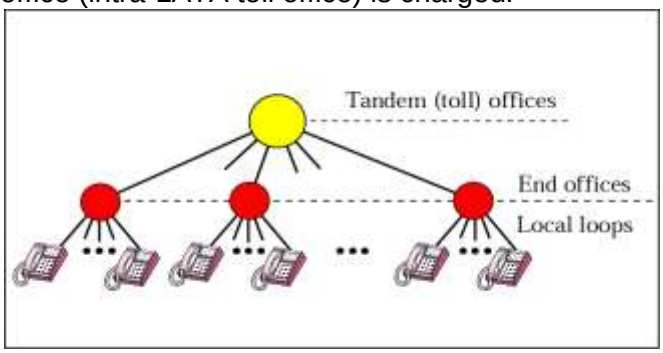


Fig. Switching offices in a LATA

- Interexchange carriers (IXCs) or long-distance companies handle services between LATAs.
- Carriers that provide inter-LATAs include AT&t, MCI, WorldCom.
- A telephone call going through an IXC is normally digitized, with the carriers using several types of networks to provide service.
- Intra-LATA services can be provided by several LECs (one ILEC and possibly more than one CLEC).
- Point of Presence (POP) is a switching office.
- Each IXC that wants to provide inter-LATA services in a LATA must have a POP in that LATA.

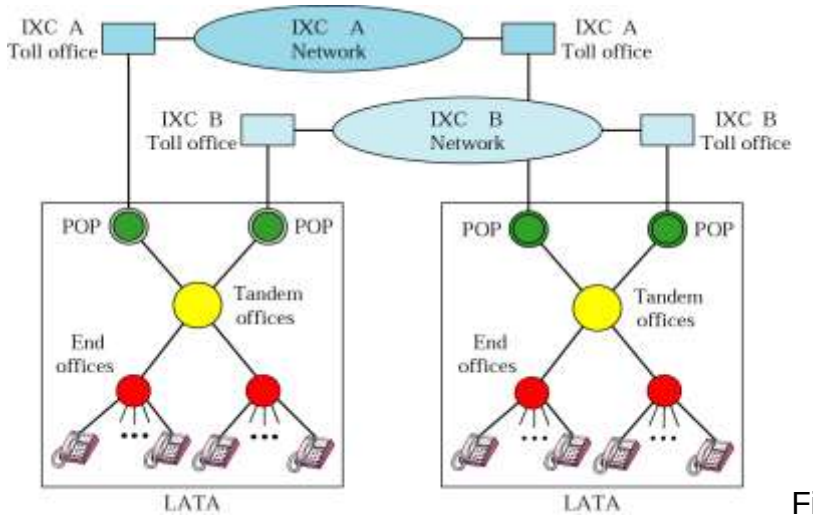


Fig: POPs

Making a Connection:

- Accessing the switching station at the end offices is accomplished through dialing.
- In the past, telephones featured rotary or pulse dialing, in which a digital signal was sent to the end office for each number dialed. This type of dialing was prone to errors due to the inconsistency of humans during the dialing process.
- Today, dialing is accomplished through the touch-tone technique. In this method, instead of sending a digital signal, the user sends two small bursts of analog signals, called dual tone. The frequency of the signals sent depends on the row and column of the pressed pad.
- Pressing number 8 will generate two bursts of analog signals with frequencies 852 and 1336 Hz to the end office.

Analog Services [Analog Switches Service]

- Local loop is analog; bandwidth is usually between 0 and 4000 Hz.
- With switched lines, when the caller dials a number, the call is conveyed to a switch, or series of switches, at the exchange. The appropriate switches are then activated to link the caller's line to that of the person being called. The switch connects the two lines for the duration of the call.
- Local Call services: Flat monthly rate OR rate for each call or a set of calls.
- Toll Call services:
  - Toll call can be intra-LATA or inter-LATA.
  - Inter-LATA calls are long distance calls [that pass via a tandem office (toll office)] and are charged for.
- 800 Services:
  - If a subscriber (normally an organization) needs to provide free connections for other subscribers (normally customers), it can request an 800 service [also 888, 877, 866]. Call is free for caller but it is paid by the callee. Rate is less expensive than a normal long distance call.
- WATS: Wide-Area Telephone Service
  - It is the opposite of 800/888 service. Charged for outbound calls.
  - Service is a less expensive alternative to regular toll calls; charges are based on number of calls.
  - Service can be specified as outbound calls to the same state, to several states, or to the whole country, with rates charged accordingly.
- 900 Services:
  - Call is paid by the caller and is normally much more expensive than a normal long-distance call. The reason is that the carrier charges two fees; the first is the long-distance toll, and the second is the fee paid to the callee for each call.
  - This service is used by organization that needs to charge customers for its services.
- Analog Leased Services
  - Offers customers the opportunity to lease a line, sometimes called a dedicated line, that is permanently connected to another customer.
  - Although the connection still passes through the switches in the telephone network, subscribers experience it as a single line because the switch is always closed, no dialing is needed.
- Digital Services
  - Digital Services are less sensitive than analog services to noise and other forms of interference.
  - Common digital services are switched/56 and digital data service (DDS).
- Switched/56 Service
    - Digital version of an analog switched line. It is a switched digital service that allows data rates of up to 56 Kbps. To communicate through this service, both parties must subscribe. A caller with normal telephone service cannot connect to a telephone or computer with switched/56 even if using a modem.
    - On the whole, digital and analog services represent two completely different domains for the telephone companies.
    - Switched/56 service is digital and so subscribers do not need modems to transmit digital data. However, the do need another device called a digital service unit (DSU). This device provides 56 Kbps and encodes the digital data in the format used by service provider.
    - Supports bandwidth on demand, video conferencing, fast facsimile, multimedia, fast data transfer, etc. Also allows subscribers to obtain higher speeds by using more than one line (inverse multiplexing).

- Digital Data Service

- Digital version of an analog leased line; it is a digital leased line with a maximum data rate of 64 Kbps.

- Telephone history
  - Before 1984
    - Local and long-distance services were provided by AT&T Bell System.
    - By law, this monopoly company was broken into AT&T Long lines, 23 Bell Operating Companies (BOCs) and others.
    - Telephone rates became lower after this law.
  - Between 1984 and 1996
    - LATAs and IXCs were formed.
    - No LEC provide long-distance services and no IXC provide local services.
  - After 1996
    - A common carrier company provides both inside the LATA and between LATA services.
    - To avoid recabling of residents, the carrier that was given intra-LATA services (ILEC) continues to provide the main services; the new competitors (CLEC) provide other services.

ERROR DETECTION AND CORRECTION

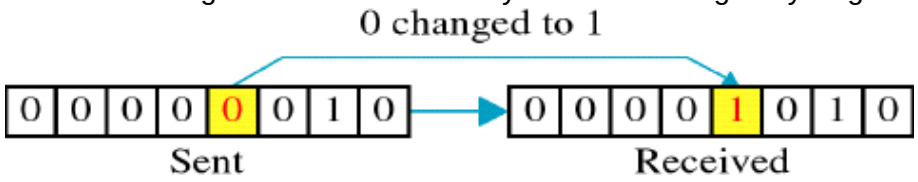
ERROR:

Data can be corrupted during transmission. For reliable communication, errors must be detected and corrected.

TYPES OF ERRORS:

- **Single bit Error:**

The term single bit error means that only one bit of a given data unit is changed from 1 to 0 or 0 to 1. 010101 is changed to 110101 here only one bit is changed by single bit error.

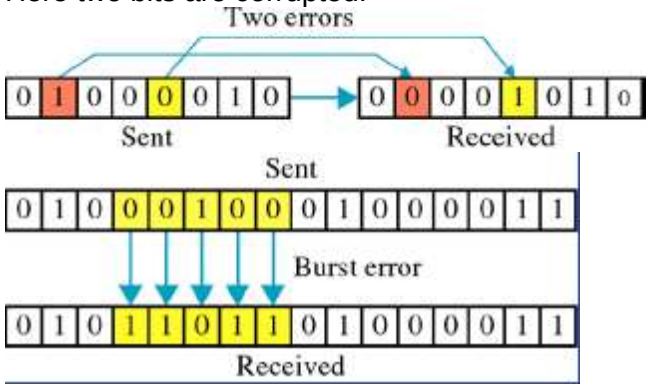


- **Burst Error:**

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

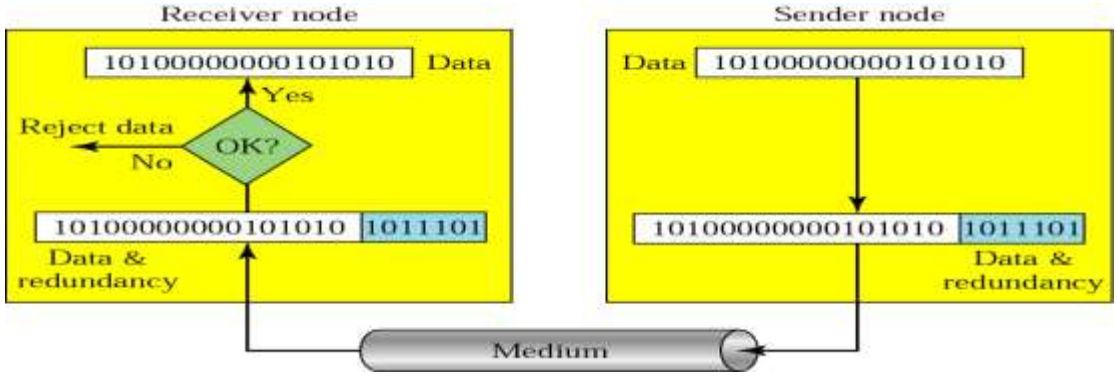
Example:

Here two bits are corrupted.



Redundancy

Error detection use the concept of redundancy, which means adding extra bits for detecting errors at the destination .i.e., instead of repeating the entire data stream, a shorter group of bits may be appended to the end of each unit.



Detection methods

- **Parity check**
- **Cyclic redundancy check**
- **checksum**

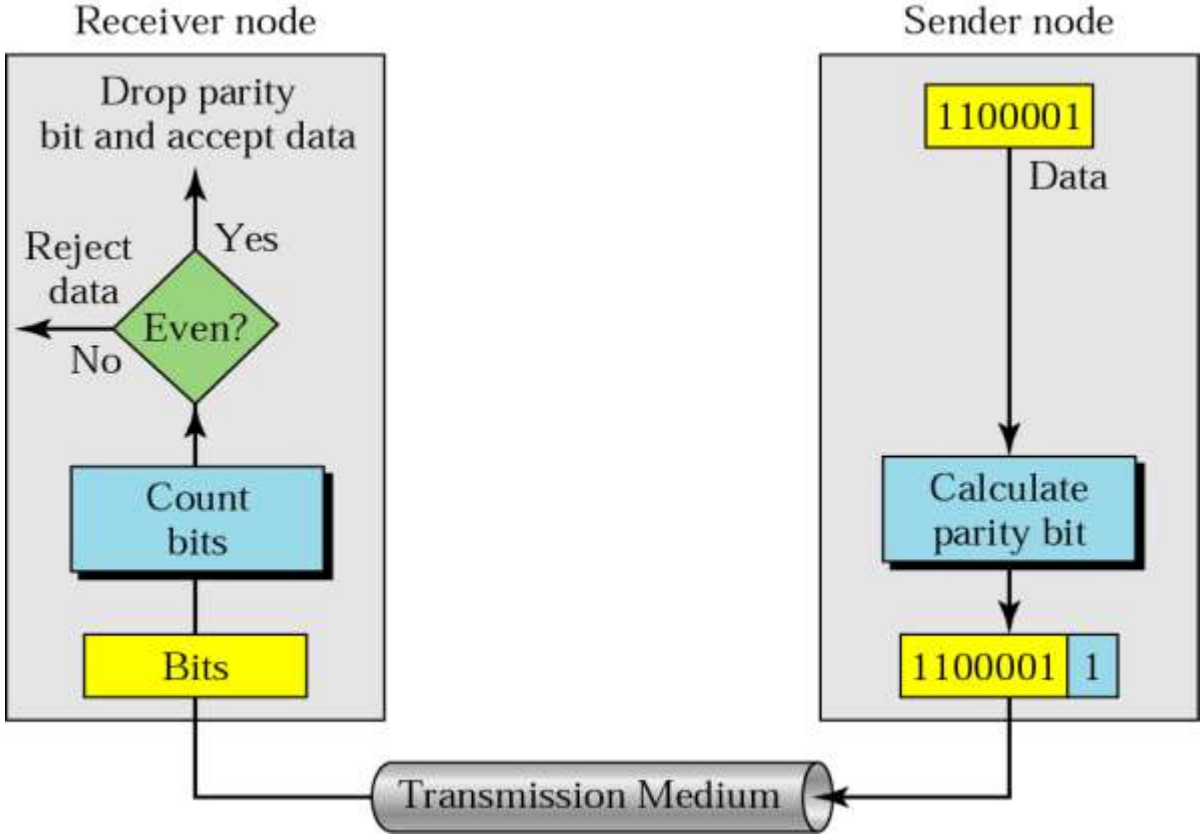
Parity check

A redundant bit called parity bit is added to every data unit so that the total number of 1's in the unit becomes even (or odd).

SIMPLE PARITY CHECK

In a simple parity check a redundant bit is added to a string of data so that total number of 1's in the data become even or odd.

The total data bit is then passed through parity checking function. For even parity, it checks for even number of 1's and for odd parity it checks even number of 1's. If an error is detected the data is rejected.



Example 1: data to be transmitted = 10110101

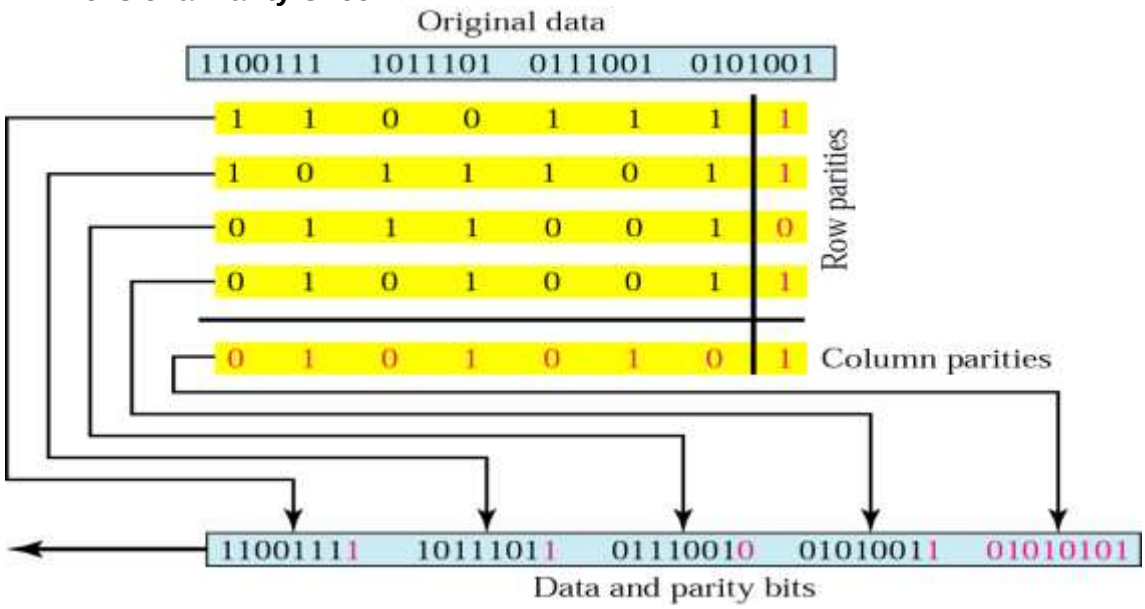
- 5 1's in the data
- Parity bit is 1 to make even
- Transmitted codeword = 101101011
- If receiver gets 101101011, parity check ok ---accept (OK)
- If receiver gets 101100011, parity check fails ---reject (OK), ask for frame to be re-transmitted
- If receiver gets 101110011, parity check ok ---accept (NOT OK: even number of errors undetected)
- If receiver gets 001100011, parity check ok ---accept (NOT OK: even number of errors undetected)
-

Example 2: data to be transmitted = 10110001

- 4 1's in the data
- parity bit is 0

- Transmitted codeword = 101100010

2-Dimensional Parity Check:



Form data into a 2-dimensional array; add single parity check bits to each row and each column; transmit row-by-row

Example: data = 1110001 1000111 0011001

- Form 3×7 array and add row and column parity bits:

Data bits	
1110001	0
1000111	0 row
0011001	1 parity bits
0101111	1
Column parity bits	

- transmitted: 11100010 10001110 00110011 01011111
- Receiver knows to form received bit string into 4×8 array, then check the row and column parity bits…
- Can **detect** any odd number of bit errors in a row or column, and can detect an even number of bit errors if they're in a single row (using the column parity checks) or in a single column (using the row parity checks); and can **correct** any single bit error

•Example (cont.): suppose bit in position (1,3) is received in error (in other words, 1 bit error)

1 1 0 0 0 0 1 ***0*** row 1 parity check fails
1 0 0 0 1 1 1 ***0*** row 2 parity check ok
0 0 1 1 0 0 1 ***1*** row 3 parity check ok
0 1 0 1 1 1 1 1 row 4 parity check ok

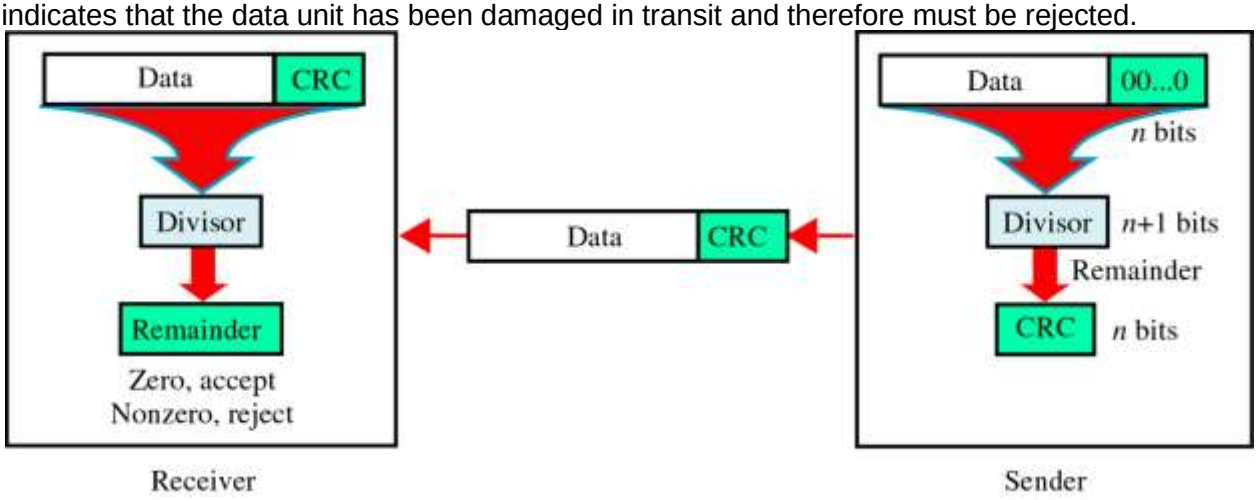
column 1 parity check fails
column 2 parity check ok
column 3 parity check fails
column 4 parity check ok
column 5 parity check ok
column 6 parity check ok
column 7 parity check ok

column 8 parity check ok

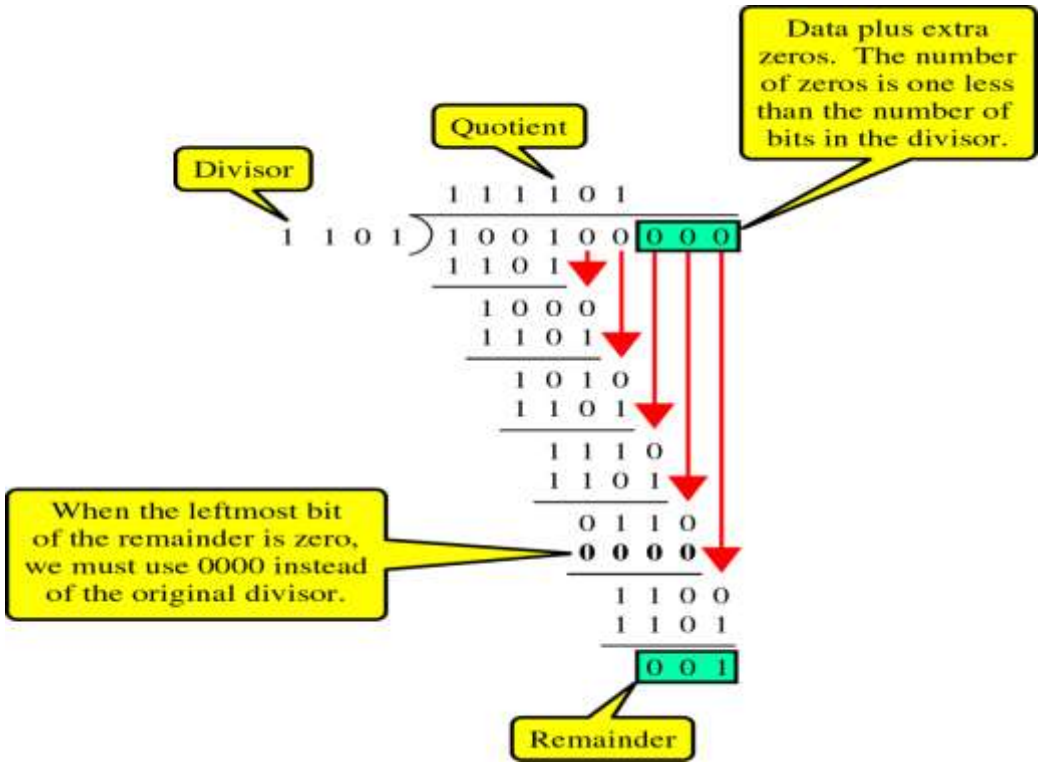
Therefore the receiver can detect that bit errors Occurred, but it cannot Correct them (here, if the Bit errors were in positions (1,3) and (2,1) instead, the receiver parity checks would be the same)

CYCLIC REDUNDANCY CHECK

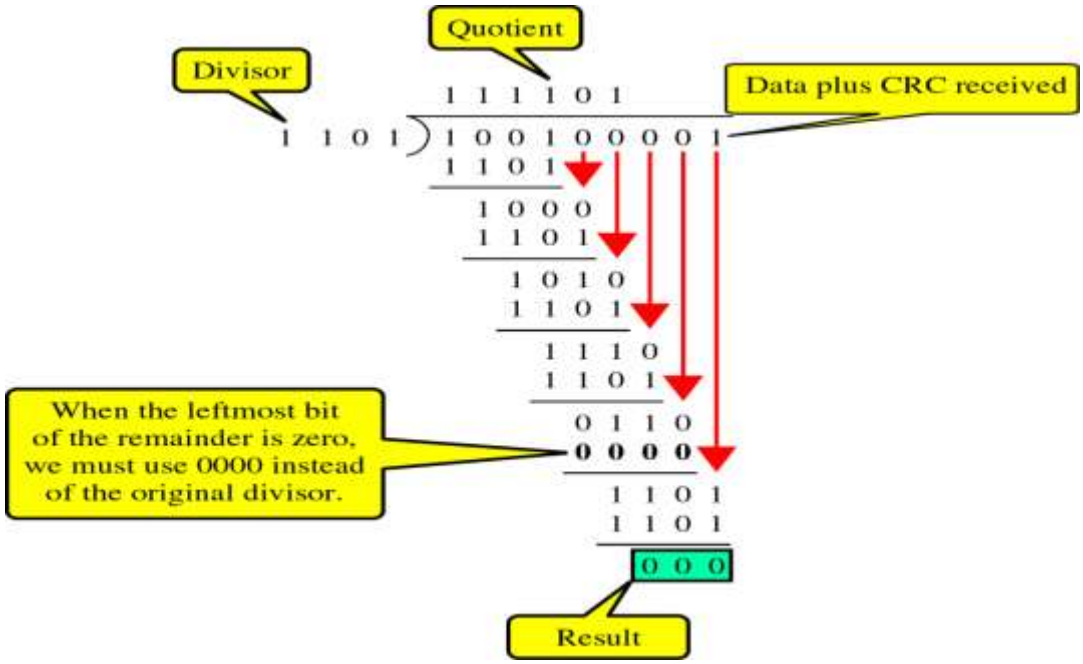
CRC is based on binary division. In CRC, instead of adding bits to achieve the desired parity, a sequence of redundant bits, called the CRC or the CRC remainder, is appended to the end of the data unit so that the resulting data unit becomes exactly divisible by a second, predetermined binary number. At its destination, the incoming data unit is assumed to be intact and is therefore accepted. A remainder indicates that the data unit has been damaged in transit and therefore must be rejected.



CRC GENERATOR:

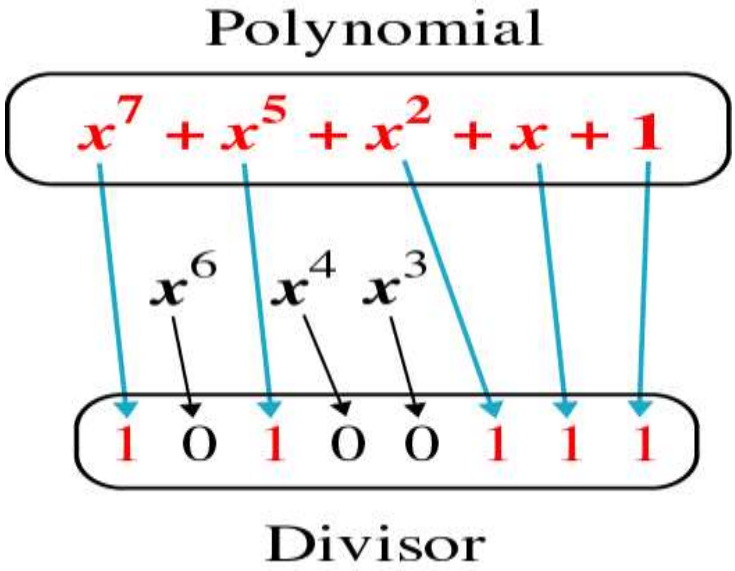


CRC CHECKER:



POLYNOMIALS

The divisor in the CRC most often represented not as a string of 1s and 0s, but as an algebraic polynomial. The polynomial format is useful to solve the concept mathematically.



Standard Polynomials:

CRC-12

$x^{12} + x^{11} + x^3 + x + 1$

CRC-16

$x^{16} + x^{15} + x^2 + 1$

CRC-ITU-T

$x^{16} + x^{12} + x^5 + 1$

CRC-32

$x^{32} + x^{26} + x^{23} + x^{22} + x^{16} + x^{12} + x^{11} + x^{10} + x^8 + x^7 + x^5 + x^4 + x^2 + x + 1$

Checksum:

~ used by the higher layer protocols
~ is based on the concept of redundancy (VRC, LRC, CRC ….)

❑ To create the checksum the sender does the following:

❖ The unit is divided into K sections, each of n bits.
❖ Section 1 and 2 are added together using one's complement.
❖ Section 3 is added to the result of the previous step.
❖ Section 4 is added to the result of the previous step.
❖ The process repeats until section k is added to the result of the previous step.
❖ The final result is complemented to make the checksum.

At Sender:

Original data: 10101001 00111001
10101001
00111001

11100010 Sum
00011101 Checksum
10101001 00111001 00011101

At Receiver:

Received data: 10101001 00111001 00011101
10101001
00111001
00011101

11111111 ← Sum
00000000 ← Complement

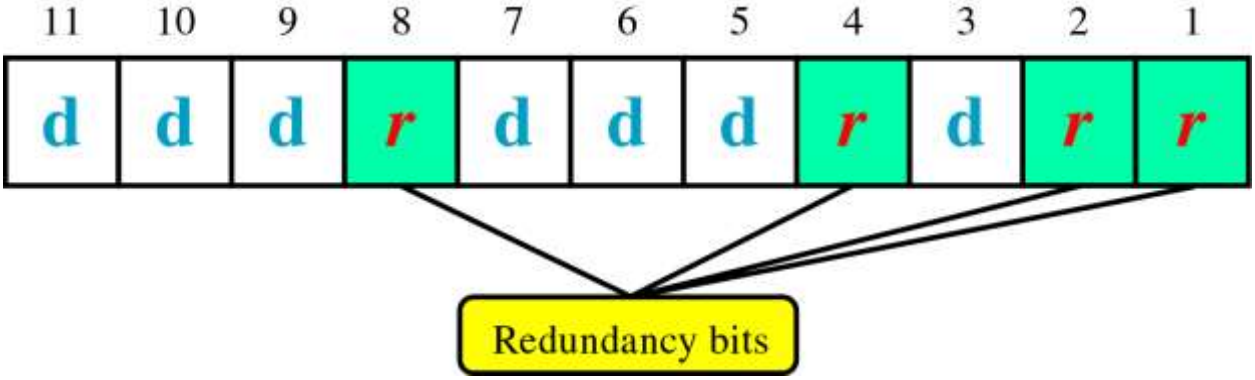
Error Correction (Hamming Encoding Algorithm):

❑ Redundancy Bits
To calculate the number of redundancy bits (R) required to correct a given number of data bit (M)

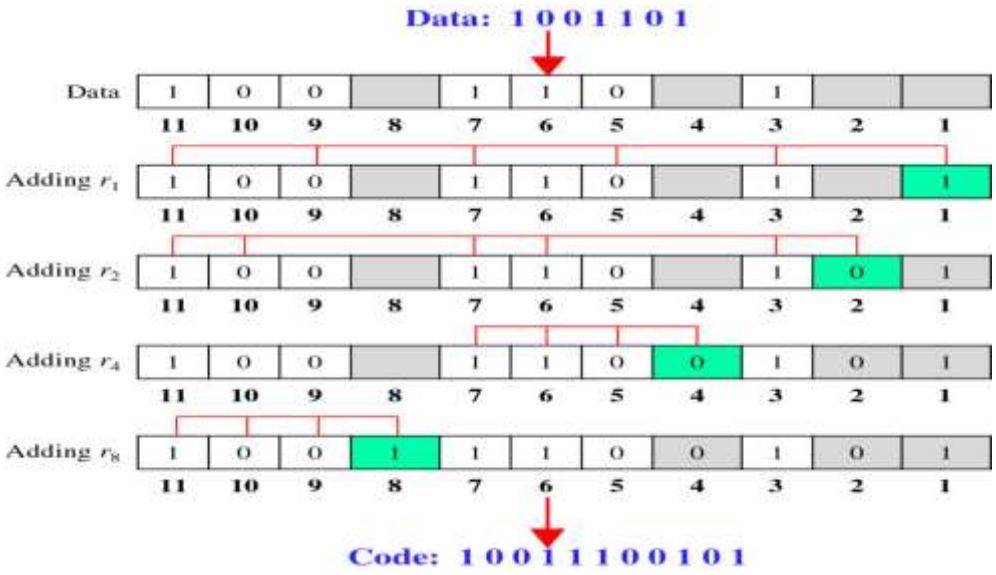
❑ If the total number of bits in a transmittable unit is m+r, then r must be able to indicate at least m+r+1 different states
$2^r \geq m + r + 1$

ex) For value of m is 7(ASCII), the smallest r value that can satisfy this equation is 4
$2^4 \geq 7 + 4 + 1$

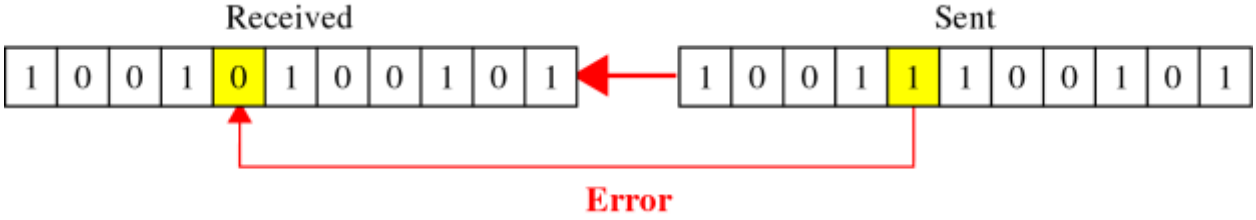
❑ **positions of redundancy bits in Hamming code:**



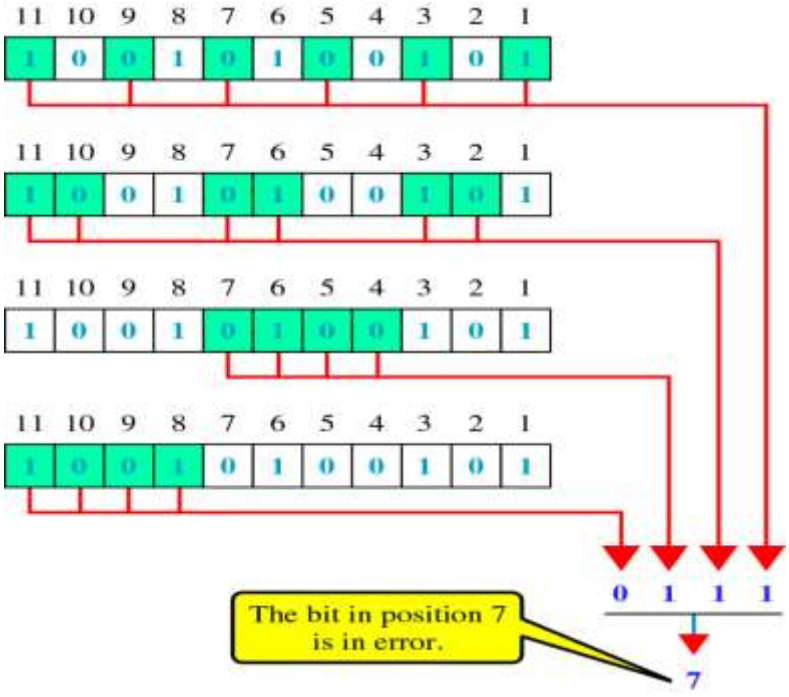
❑ each r bit is the VRC bit for one combination of data bits
r_1 = bits 1, 3, 5, 7, 9, 11
r_2 = bits 2, 3, 6, 7, 10, 11
r_4 = bits 4, 5, 6, 7
r_8 = bits 8, 9, 10, 11



- **Error Detection and Correction:**



- **Error detection using Hamming Code**



FLOW CONTROL AND ERROR CONTROL

The two main features of data kink layer are **flow** control and error control.

.**FLOW CONTROL**

Flow control coordinates that amount of data that can be sent before receiving ACK It is one of the most important duties of the data link layer.

ERROR CONTROL

- Error control in the data link layer is based on ARQ (automatic repeat request), which is the retransmission of data.
- The term error control refers to methods of error detection and retransmission.
- Anytime an error is detected in an exchange, specified frames are retransmitted. This process is called ARQ.

FLOW AND ERROR CONTROL MECHANISMS

1. STOP-AND WAIT ARQ.
2. GO-BACK-N ARQ.
3. SELECTIVE-REPEAT ARQ.

STOP-AND- WAIT ARQ

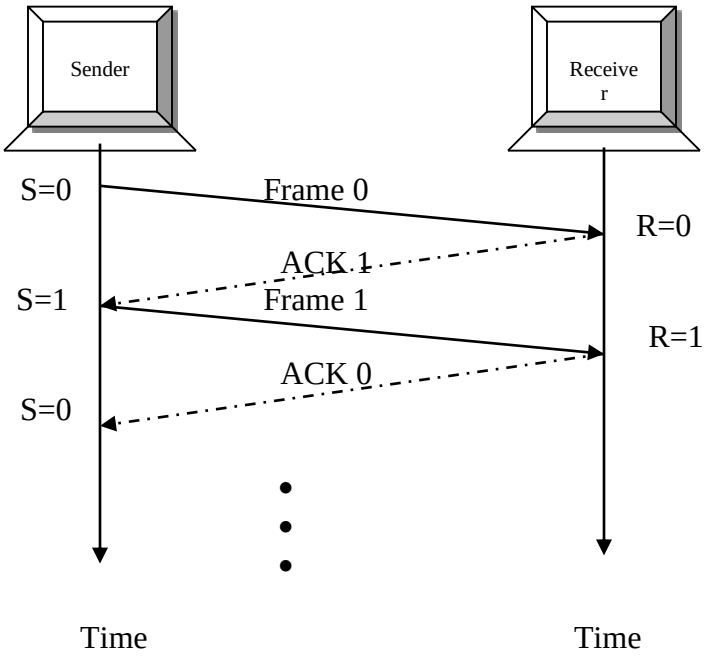
This is the simplest flow and error control mechanism. It has the following features.

- The sending devise keeps the copy of the last frame transmitted until it receives an acknowledgement for that frame. Keeping a copy allows the sender to re- transmit lost or damaged frames until they are received correctly.
- Both data and acknowledgement frames are numbered alternately 0 and 1. A data frame 0 is acknowledged by an ACK 1.
- A damaged or lost frame is treated in the same manner by the receiver. If the receiver detects an error in the received frame, it simply discards the frame and sends no acknowledgement.
- The sender has a control variable, which we call S, that holds the number of recently sent frame. The receiver has a control variable, which we call R that holds the number of the next frame expected.
- The sender starts a timer when it sends a frame. If an ACK is not received within an allotted time period the sender assumes that the frame was lost or damaged and resends it.
- The receivers send only positive ACK for frames received safe and sound; it is silent about the frames damaged or lost.

OPERATION:

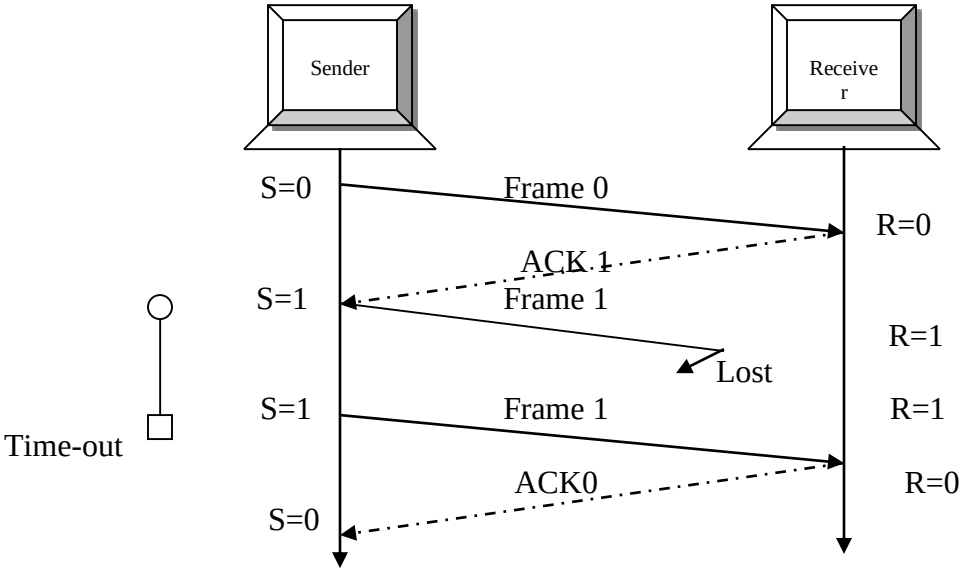
The possible operations are
Normal operation
lost frame
ACK lost
delayed ACK.
The sender sends frame 0 and wait to receive ACK 1. when ACK 1 is received it sends frame 1 and then waits to receive ACK 0, and so on.

The ACK must be received before the time out that is set expires. The following figure shows successful frame transmission.



Lost or damaged acknowledgement

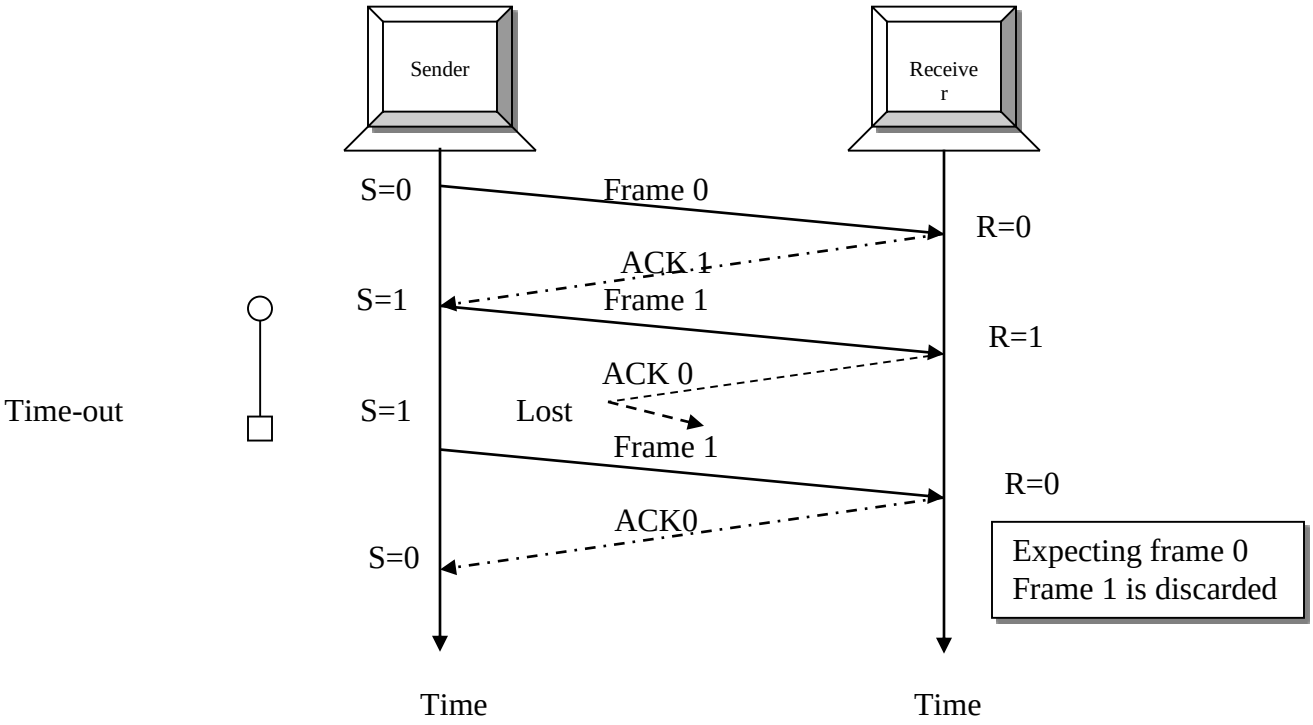
- When the receiver receives the damaged frame it discards it, which essentially means the frame is lost. The receiver remains silent about a lost frame and keeps its value of R.
- For example in the following figure the sender transmits frame 1, but it is lost. The receiver does nothing, retaining the value of R (1). After the timer at the sender site expires, another copy of frame 1 is sent.



Lost acknowledgement

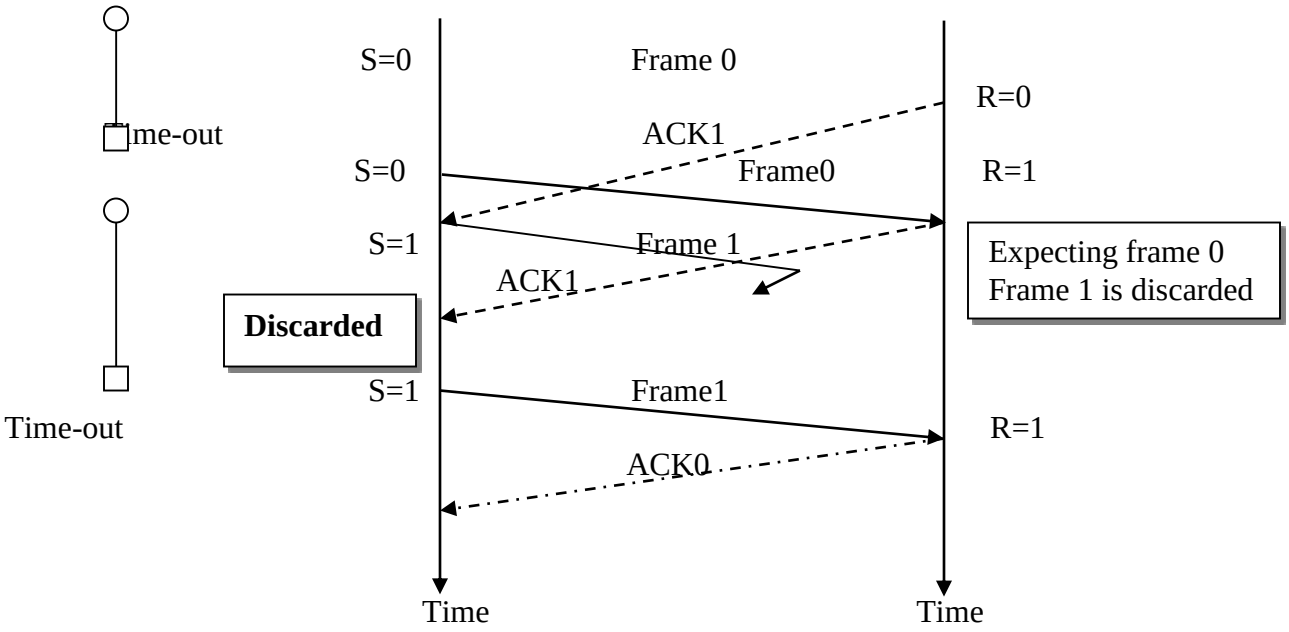
- A lost or damaged ACK is handled in the same by the sender; if the sender receives a damaged ACK, it discards it.
- The following figure shows a lost ACK 0.the waiting sender does not know if frame 1 has been received. When the timer for frame 1 expires the sender retransmits frame 1.

- Note that the receiver has already received frame 1 and is expecting to receive frame 0. Therefore, its silently discards the second copy of frame 1.
- The following figure shows a lost ACK 0.the waiting sender does not know if frame 1 has been received. When the timer for frame 1 expires the sender retransmits frame 1.
- Note that the receiver has already received frame 1 and is expecting to receive frame 0. Therefore, its silently discards the second copy of frame 1.



- **Delayed acknowledgement**

- An ACK can be delayed at the receiver or by some problem with the link. The following figure shows the delay of ACK 1; it ids received after the timer for frame 0 as already expired.
- The sender has already retransmitted a copy of frame 0. The receiver expects frame 1 so its simply discards the duplicate frame 0.
- The sender has now received two ACK's, one that was delayed and one that was sent after the duplicate frame 0 arrived. The second ACK 1 is discarded.

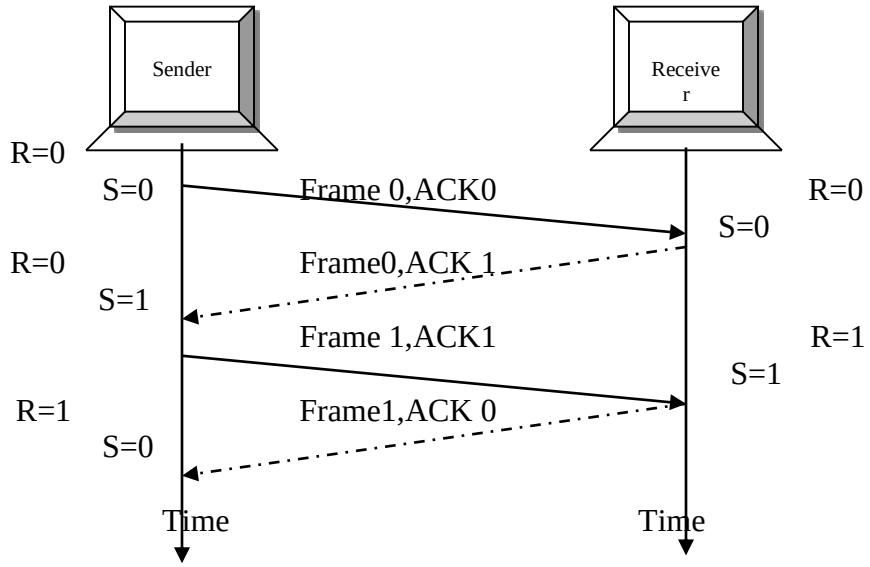


BIDIRECTIONAL TRANSMISSION

The stop – and – wait mechanism is unidirectional. We can have bi-directional transmission if the two parties have two separate channels for full duplex communication or share the same channel for off duplex transmission. In this case, each party needs both S and R variables to track frames sent and expected.

PIGGYBACKING

It's a method to combine a data frame with an ACK. In following figure both the sender and the receiver have data to send. Instead of sending separate data and ACK frames. It can save bandwidth because the overhead from a data frame and an ACK frame can be combined into just one frame.



GO-BACK-N ARQ

- As in Stop-and-wait protocol senders has to wait for every ACK then next frame is transmitted. But in GO-BACK-N ARQ number of frames can be transmitted without waiting for ACK. A copy of each transmitted frame is maintained until the respective ACK is received.

Features of GO-BACK-N ARQ

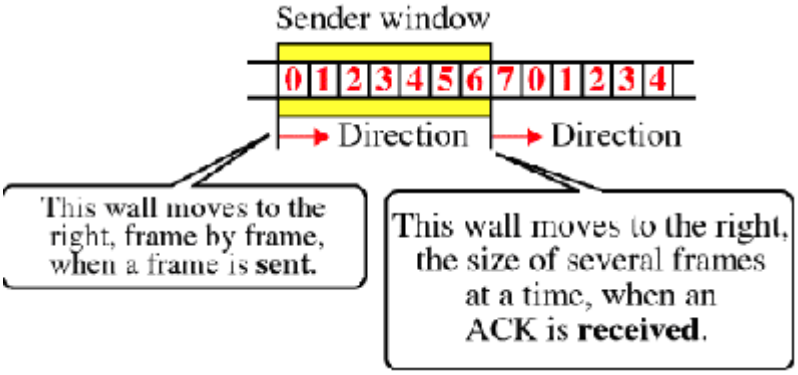
1.**sequence numbers.**

Sequence numbers of transmitted frames are maintained in the header of frame. If k is the number of bits for sequence number, then the numbering can range from 0 to 2k-1. Example: if k=3 means sequence numbers are 0 to 7.

2. sender sliding window:

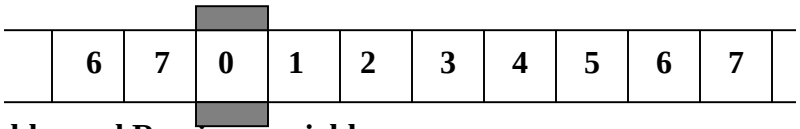
- Window is a set of frames in a buffer waiting for ACK. This window keeps on sliding in forward direction, the window size is fixed. As the ACK is received, the respective frame goes out of window and new frame to sent come into window. Figure illustrates the sliding window.
- If Sender receives. ACK 4, then it ***knows Frames upto*** and including Frame 3 were ***correctly received***

Window size=7



3. Receiver sliding window:

In the receiver side size of the window is always one. The receiver is expecting to arrive frame in specifies sequence. Any other frame is received which is out of order is discarded. The receiver slides over after receiving the expected frame. The following figure shows the receiver side-sliding window.



4. Control variables:

Sender variables and Receiver variables:

Sender deals with three different variables

S -> sequence number of recently sent frame

S_F -> sequence number of first frame in the window.

S_L -> sequence number of last frame in the window.

The receiver deals with only one variable

R -> sequence number of frame expected.

5. Timers

The sender has a timer for each transmitted frame. The receivers don't have any timer.

6. Acknowledgement:

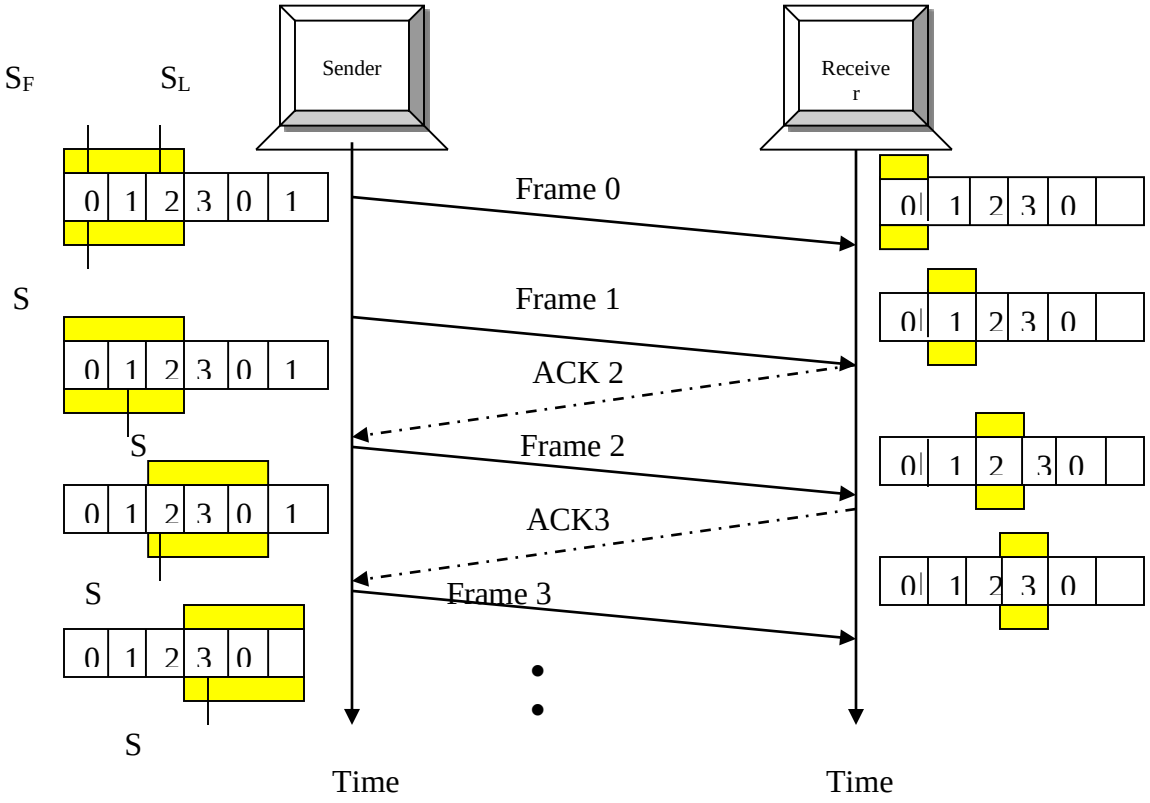
The receiver responds for frame arriving safely by positive ACK. For damaged or lost frames receiver doesn't reply, the sender has to retransmit it when timer of that frame elapsed. The receiver may ACK once for several frames.

7. resending frames:

If the timer for any frame expires, the sender has to resend that frame and the subsequent frame also, hence the protocol is called GO-BACK-N ARQ.

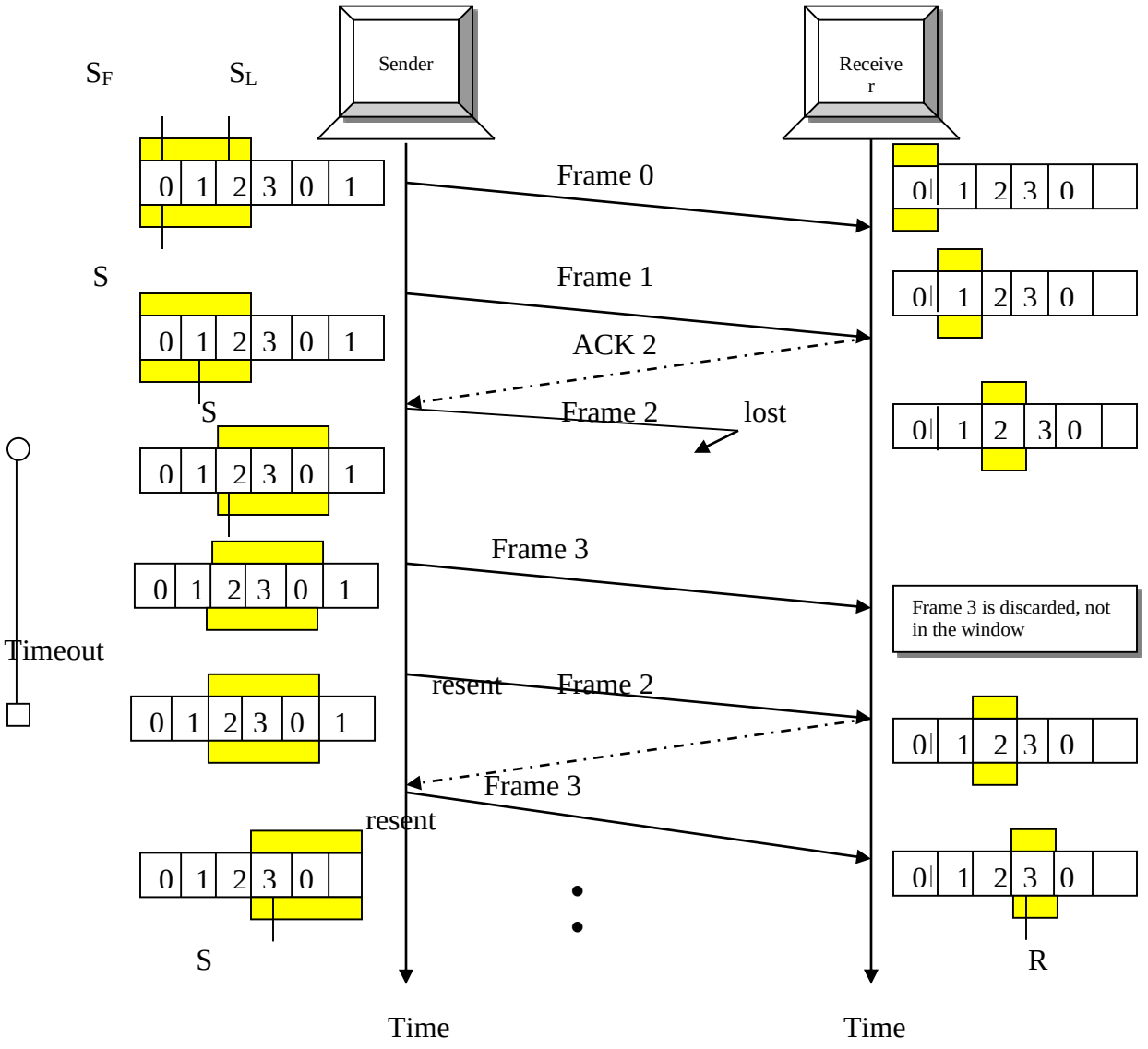
Operation

Normal operation: Following diagram shows this mechanism. The sender keeps track of the outstanding frames and updates the variables and windows as acknowledgements arrive.



Damaged or lost frame:

Figure shows that frame 2 is lost. Note that when the receiver receives frame 3, it is discarded because the receiver is expecting frame 2, not frame3. after the timer for frame 2 expires at the sender site, the sender sends frame 2 and 3.



Damaged or lost acknowledgement:

If an ACK is lost, we can have two situations. If the next ACK arrives before the expiration of timer, there is no need for retransmission of frames because ACK are cumulative in this protocol.. if the next ACK arrives after the timeout, the frame and all the frames after that are resent. The receiver never resends an ACK.

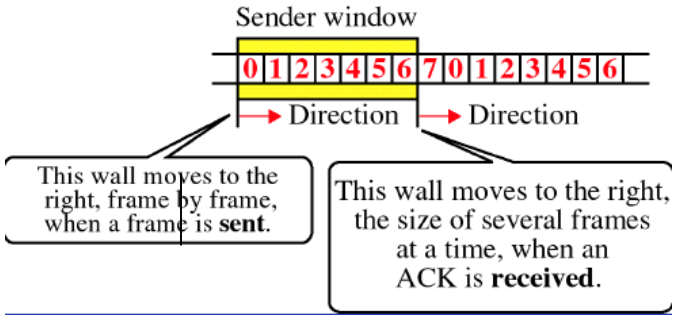
For diagrams refer your class work notes.

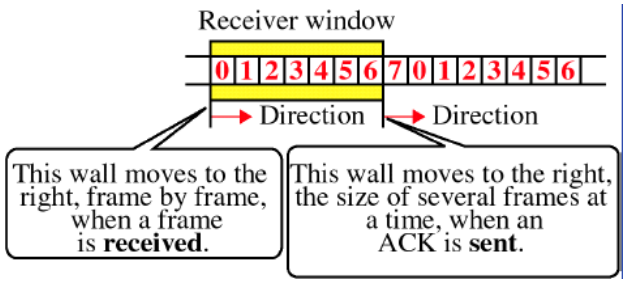
Delayed Acknowledgement:

A delayed ACK also triggers the resending of frames.

SELECTIVE REPEAT ARQ:

- The configuration and its control variables for this are same as those selective repeat ARQ.
- The size of the window should be one half of the value 2^m.
- The receiver window size must also be the size. In this the receiver is looking for a range of sequence numbers.
- The receiver has control variables R_F and R_L to denote the boundaries of the window.

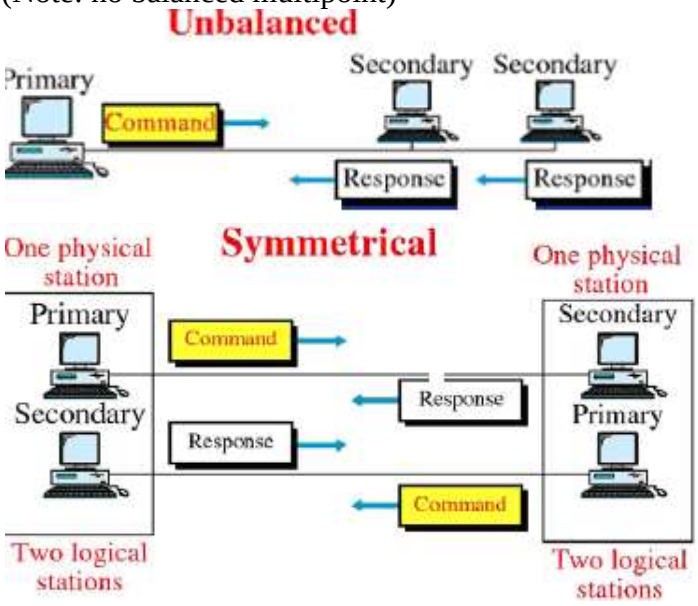


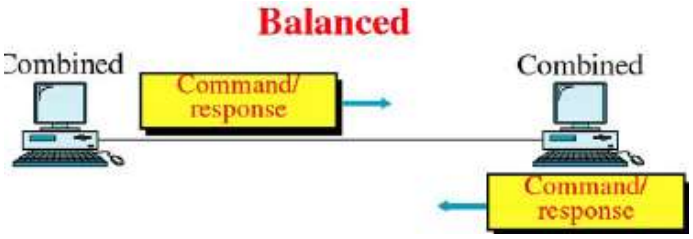


selective repeat also defines a negative ACK NAK that reports the sequence number of a damaged frame before the timer expires.

High-level Data Link Control (HDLC) protocol

- HDLC standardized ISO in 1979 and accepted by most other standards bodies (ITU-T, ANSI)
- 3 types of end-stations:
  Primary–sends commands
  Secondary–can only responds to Primary's commands
  Combined–can both command and respond
- 3 types of configuration
  (Note: no balanced multipoint)





TRANSFER MODE

- Normal Response Mode (NRM)
  - unbalanced config, primary initiates transfer
  - used on multi-drop lines, eg host + terminals
- Asynchronous Balanced Mode (ABM)
  - balanced config, either station initiates transmission, has no polling overhead, widely used
- Asynchronous Response Mode (ARM)
  - unbalanced config, secondary may initiate transmit without permission from primary, rarely used

FRAMES:

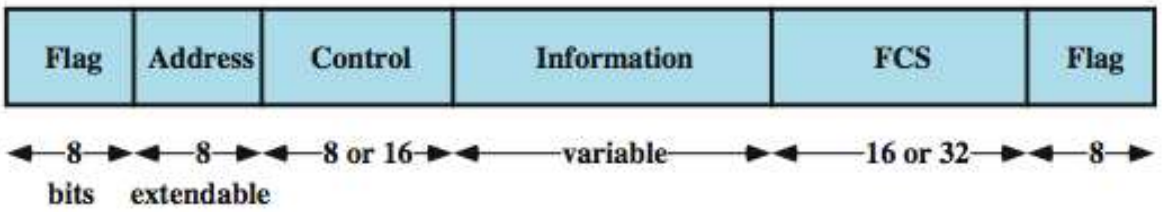
3 types of Frames are

I-Frame **–** transports user data and control info about user data**.**

S-Frame **–** supervisory Frame, only used for transporting control information

U-Frame **–** unnumbered Frame, reserved for system management(managing the link itself)

FRAME FORMAT



(a) Frame format

Flag Fields:

- Delimit frame at both ends
- 01111110
- Receiver hunts for flag sequence to synchronize
- Bit stuffing used to avoid confusion with data containing 01111110
  - — The transmitter inserts 0 bit after every sequence of five 1s with the exception of flag fields
  - — If receiver detects five 1s it checks next bit
    - If 0, it is deleted
    - If 1 and seventh bit is 0 (i.e., 10), accept as flag
    - If sixth and seventh bits 1 (i.e., 11), sender is indicating abort

Original Pattern:

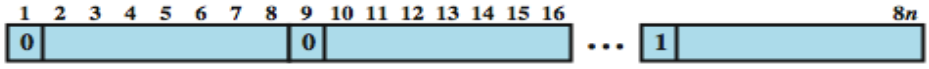
```
111111111111011111101111110
```

After bit-stuffing

```
111110111110110111110101111101O
```

Address Field:

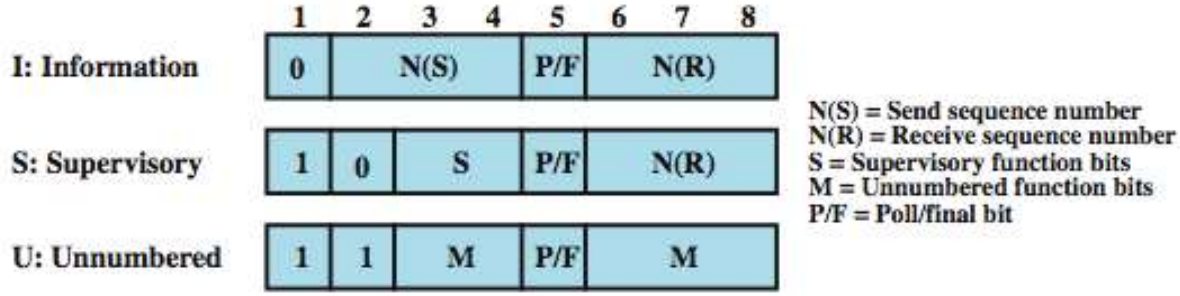
- identifies secondary station that sent or will receive frame
- usually 8 bits long
- may be extended to multiples of 7 bits
  - LSB indicates if is the last octet (1) or not (0)
- all ones address 11111111 is broadcast



- **(b) Extended Address Field**

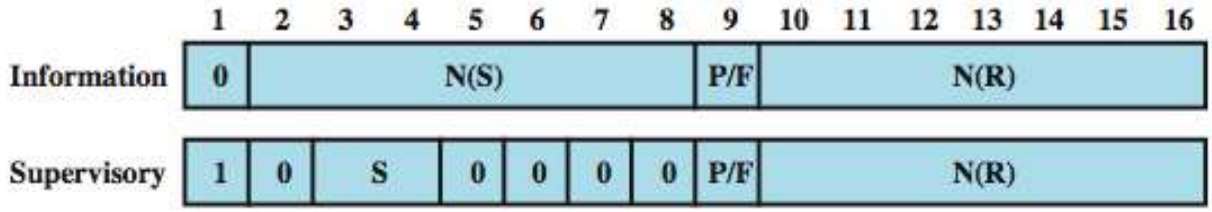
Control Field:

- different for different frame type
  - Information - data transmitted to user (next layer up)
    - Flow and error control piggybacked on information frames
  - Supervisory - ARQ when piggyback not used
  - Unnumbered - supplementary link control
- first 1-2 bits of control field identify frame type



(c) 8-bit control field format

- use of Poll/Final bit depends on context
- in command frame is P bit set to1 to solicit (poll) response from peer
- in response frame is F bit set to 1 to indicate response to soliciting command
- seq number usually 3 bits
  - can extend to 8 bits as shown below



(d) 16-bit control field format

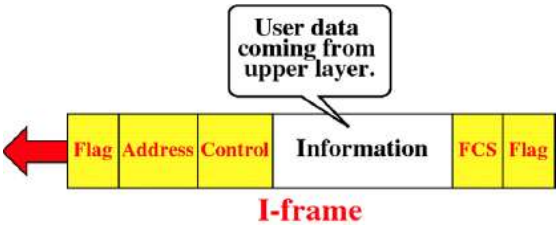
Information & FCS Fields:

- Information Field
  - in information and some unnumbered frames
  - must contain integral number of octets
  - variable length
- Frame Check Sequence Field (FCS)
  - used for error detection
  - either 16 bit CRC or 32 bit CRC

Poll/Final Bit

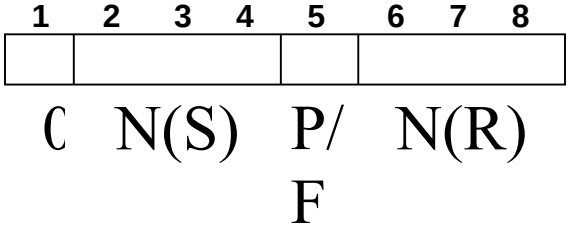
- Use depends on context
- Command frame
  - P bit : used for poll from primary
  - 1 to solicit (poll) response from peer
- Response frame
  - F bit : used for response from secondary
  - 1 indicates response to soliciting command

I-Frames:

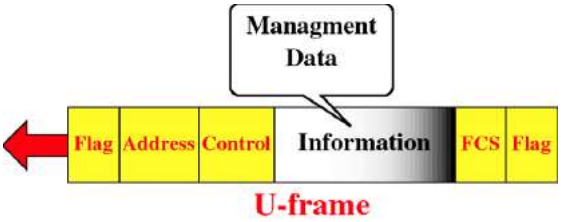


I-frame

- Contains the sequence number of transmitted frames and a piggybacked ACK

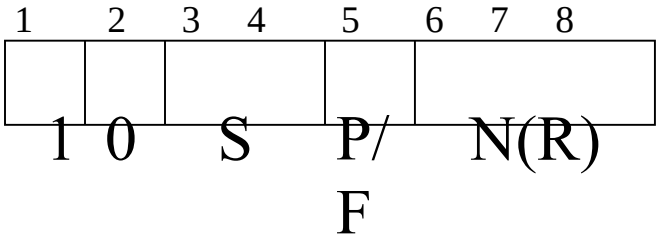


U-Frames:



U-frame

- Used for flow and error control



P/F:

- RR --- receive ready
- RNR --- receive not ready
- REJ --- reject on frame N(R)
- SREJ --- selective reject on N(R)

S-Frames:

- S-frames are similar to unnumbered frames, the main difference being that they do carry sequence information.
- Some supervisory frames function as positive and negative acknowledgements, they therefore play a very important role in error and flow control.
- Two bits indicate the frame type, so that there are four possibilities.
- Receiver Ready -RR(Positive Acknowledgement)
- Receiver Not Ready -RNR
- Reject -REJ(NAK go-back-N)
- Selective Reject -SREJ(NAK selective retransmit)

HDLC Operation

- Initialization: S-frames specify mode and sequence numbers, U-frames acknowledge
- Data Transfer: I-frames exchange user data, S-frames acknowledge and provide flow/error control
- Disconnect: U-frames initiate and acknowledge

Point-to –Point Protocol

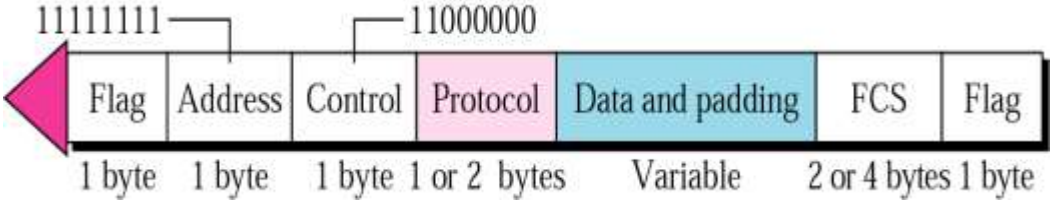
PPP

- In a network, two devices can be connected by a dedicated link or a shared link. In the first case, the link can be used by the two devices at any time. We refer to this type of access as point-to-point access. In the second case, the link is shared between pairs of devices that need to use the link. We refer to this type of access as multiple access.
- One of the most common protocols for point-to-point access is the Point-to-Point Protocol (PPP).

PPP services

- It defines the format of the frame to be exchanged between devices.
- It defines how two devices can negotiate the establishment of the link and the exchanged of data.
- It defines how network layer data are encapsulated in the data link frame.
- It defines how two devices can authenticate each other.

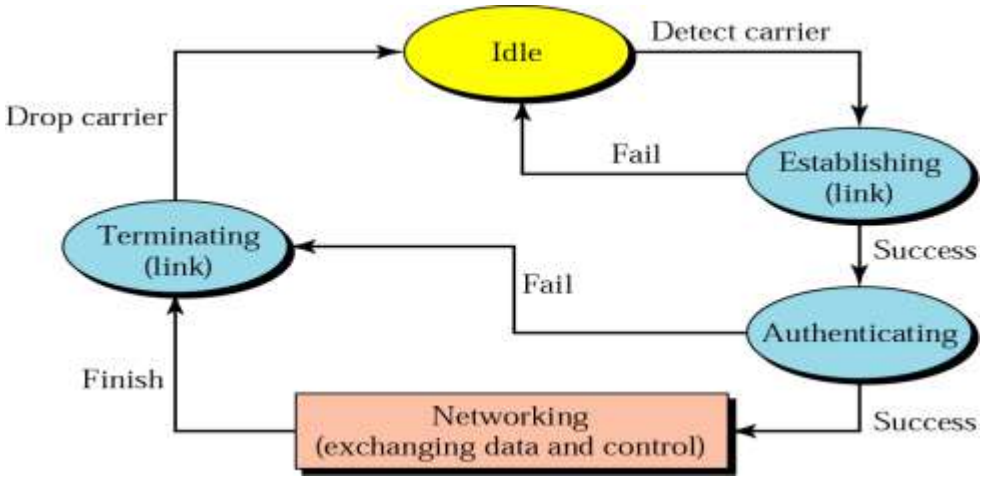
PPP FRAME FORMAT



- Flag field. The flag fields identify the boundaries of a PPP frame. Its value is 01111110.
- Address field. Because PPP is used for a point-to-point connection, it uses the broadcast address of HDCL, 11111111, to avoid a data link address in the protocol.
- Control field. The control field uses the format of the U-frame in HDCL. See pages 285-286.
- Protocol field. The protocol field defines what is being carried in the data field: user data or other information.
- Data field. This field carries either the user data or other information.
- Frame check sequence (FCS) field. This field is used for error detection**.**

Transition states

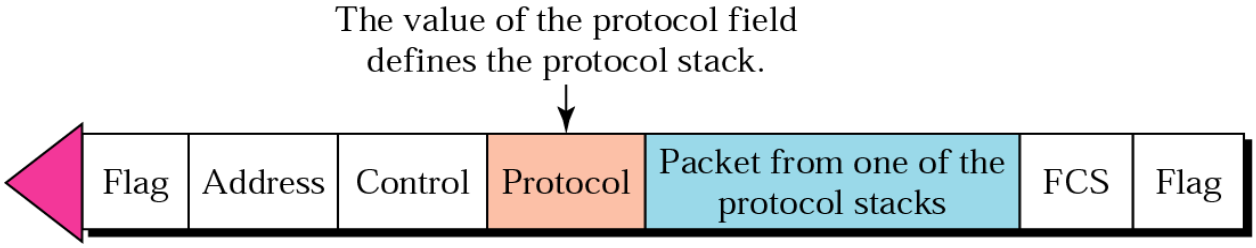
A PPP connection goes through different phases called transition sates.



- Idle state. The idle state means that the link is not being used. There is no active carrier, and the line is quiet.
- Establishing link. When one of the end point starts the communication, the connection goes into the establishing state. In this state, options are negotiated between the two parties. If the negotiation is successful, the system goes to the authenticating state (if authentication is required) or directly to the networking state.
- Authenticating state. The authenticating state is optional. If the result is successful, the connection goes to the networking state; otherwise, it goes to the terminating state.
- Networking State. When a connection reaches this state, the exchange of user control and data packets can be started. The connection remains in this state until one of the endpoints wants to terminate the connection.
- Terminating state. When the connection is in the terminating state, several packets are exchanged between the two ends for house cleaning and closing the link.

PPP Stack

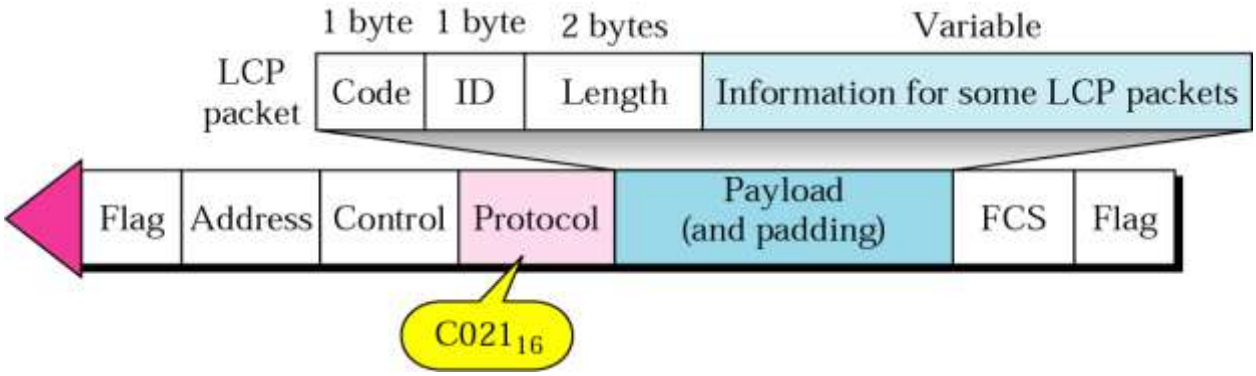
- Three sets of protocols are used by PPP: Link control protocol, authentication protocols, and network control protocol.



Link Control Protocol (LCP)

- It is responsible for establishing, maintaining, configuring, and terminating links.
- It also provides negotiation mechanisms to set options between endpoints. Both endpoints of the link must reach an agreement about the options before the link can be established.
- When PPP is carrying an LCP packet, it is either in the establishing state or in the terminating state.
- All LCP packets are carried in the data field of the PPP frame. What defines the frame as one carrying an LCP packet is the value of the protocol field, which is set to C021 (base 16).

LCP packet encapsulated in a frame



Link Control Protocol (LCP)

- Code. This field defines the type of LCP packet.
- ID. This field holds a value used to match a request with reply. One endpoint inserts a value in this field, which will be copied in the reply packet.
- Length. This field defines the length of the entire LCP packet.
- Information. This field contains extra information needed for some LCP packets.
- Configuration packets are used to negotiate the options between the two ends. There are four different types of packets for this purpose: configure-request, configure-ack, configure-nak, and configure-reject.
- Link termination packets. The link termination packets are used to disconnect the link between two endpoints.
- There are two types: **terminate-request** and **terminate-ack**.
- Link monitoring and debugging packets. These packets are used for monitoring and debugging the link. There are five types: **code-reject**, **protocol-reject**, **echo-reply**, **discard-request**.

Authentication Protocols

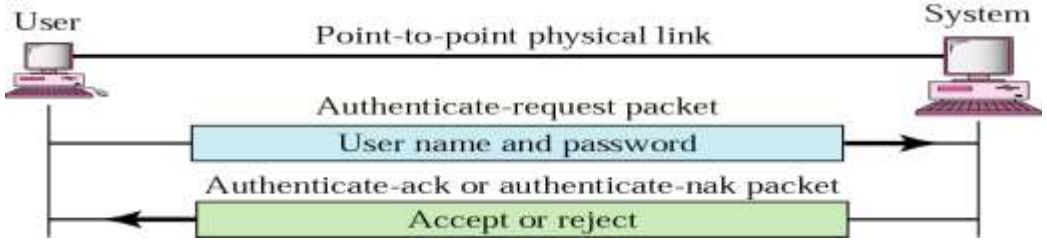
- Authentication plays a very important role in PPP because PPP is designed for use over dial-up links where verification of user identity is necessary.
- Authentication means validating the identity of a user who needs to access a set of resources.
- PPP uses two protocols for authentication: Password Authentication Protocol (PAP) and Challenge Handshake Authentication Protocol (CHAP)

PAP

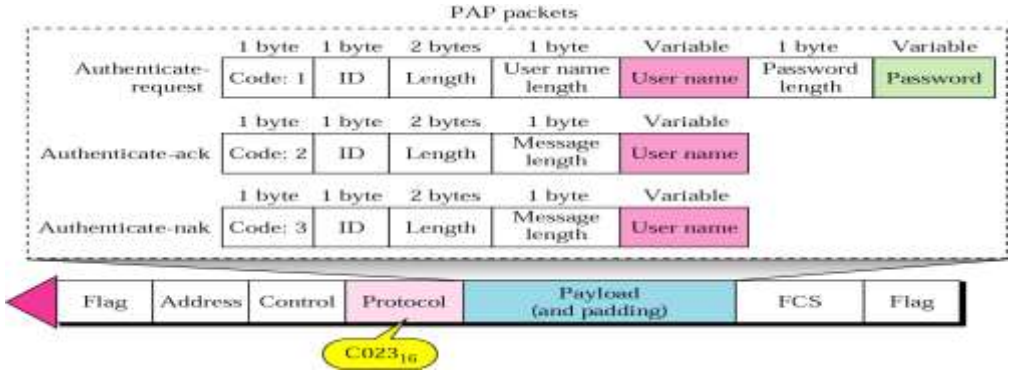
- The PAP is a simple authentication procedure with two steps:

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

- For those systems that require greater security, PAP is not enough. A third party with access to the link can easily pick up the password and access the system resources.



PAP Packets:

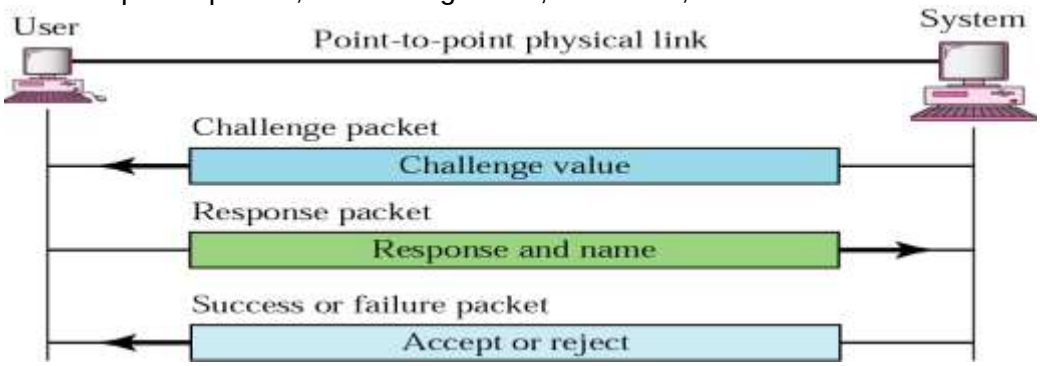


CHAP

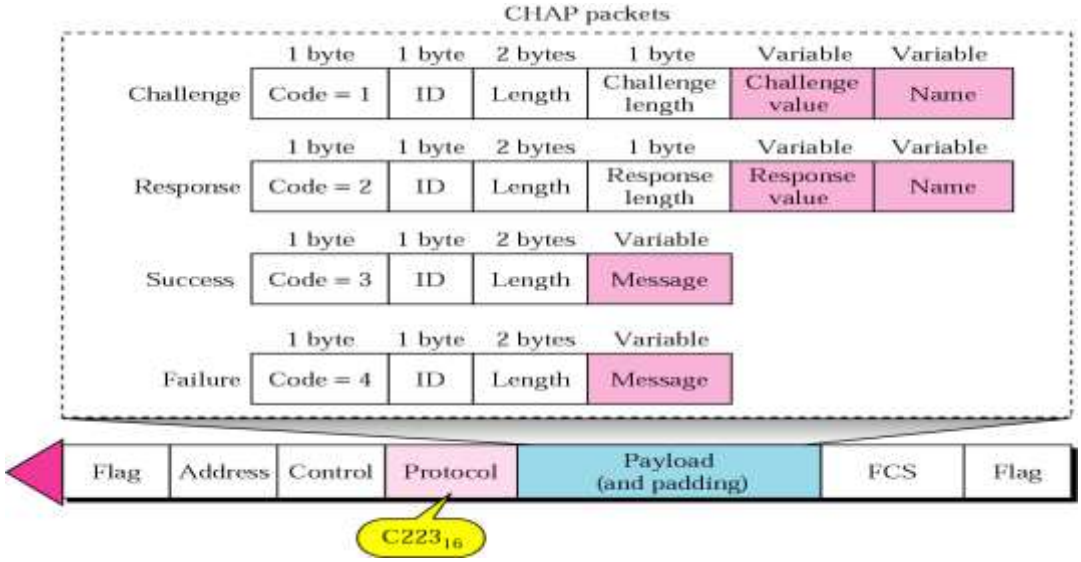
- The CHAP protocol is a three-way handshaking authentication protocol that provides greater security than PAP.
- In this method, the password is kept secret; it is never sent on-line.

Steps

- The system sends to the user a challenge packet containing a challenge value, usually a few bytes.
- The user applies a predefined function that takes the challenge value and the user's own password and creates a result. The user sends the result in the response packet to the system.
- The system does the same. It applies the same function to the password of the user and the challenge value to create a result. If the result created is the same as the result sent in the response packet, access is granted; otherwise, it is denied.



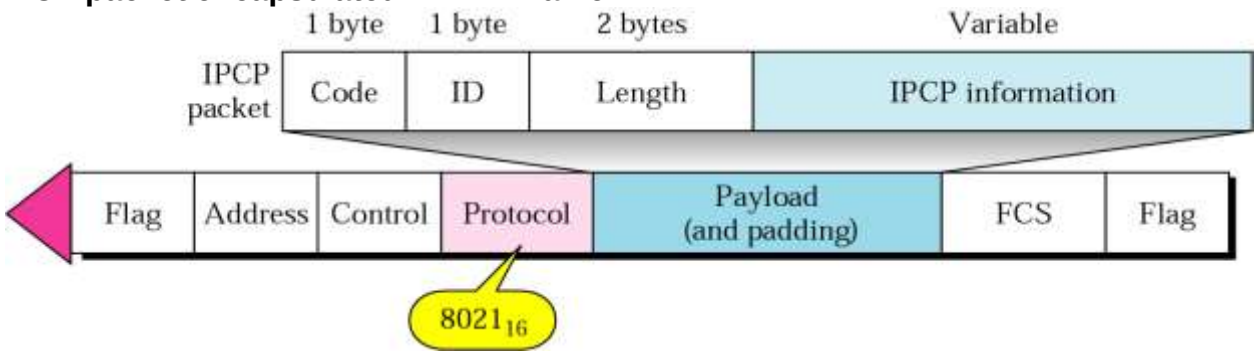
CHAP Packets:



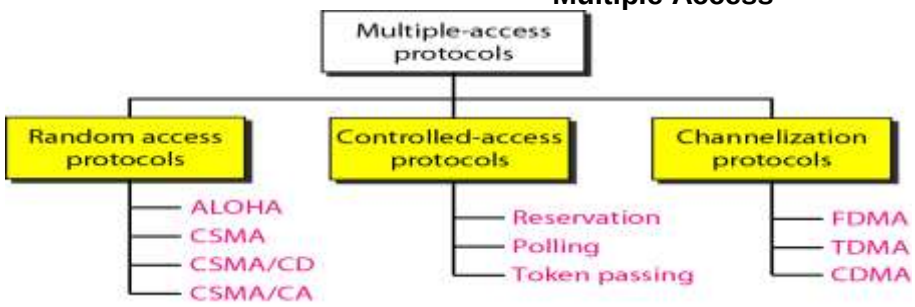
Network Control Protocol (NCP)

- After the link is established and authentication (if any) is successful, the connection goes on the networking state.
- NCP is a set of control protocols to allow the encapsulation of data coming from network layer protocols into the PPP frame.
- The set of packets that establish and terminate a network layer connection is called Internetwork Protocol Control Protocol (IPCP).

IPCP packet encapsulated in PPP frame



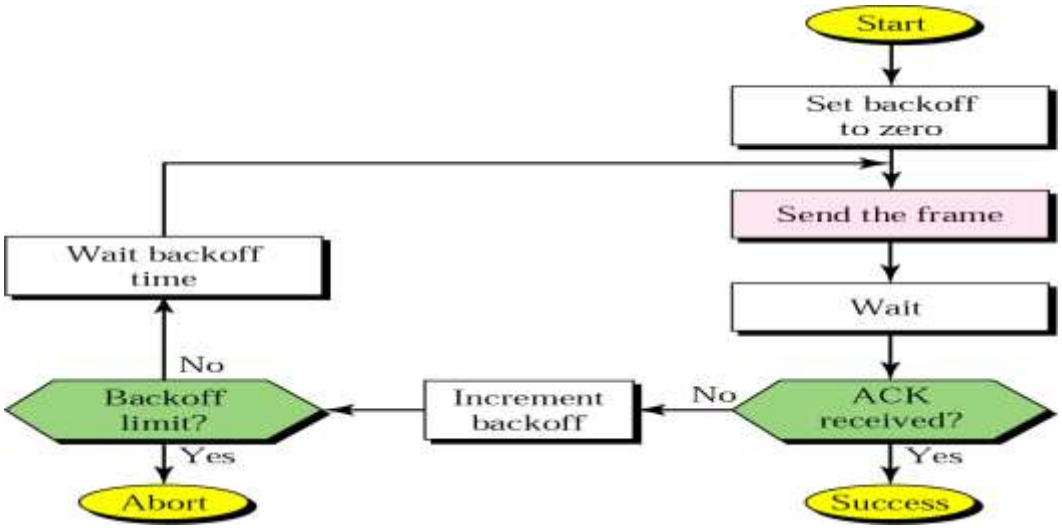
Multiple Access



RANDOM ACCESS:

- **Random Access (or contention) Protocols:**
  - No station is superior to another station and none is assigned the control over another.
  - A station with a frame to be transmitted **can use the link directly based** on a procedure defined by the protocol to make a decision on whether or not to send.
- **ALOHA Protocols**
- Was designed for **wireless LAN** and can be used for **<u>any shared medium</u>**
- **Pure ALOHA** Protocol Description
  - All frames from any station are of fixed length (**L bits**)
  - Stations transmit at equal **transmission time (*all stations produce frames with equal frame lengths*)**.
  - A station that has data **can transmit at any time**
  - **After transmitting a frame**, the sender **waits** for an **acknowledgment** for an amount of time (time out) equal to the **maximum round-trip propagation delay** = 2* t_{prop}(**see next slide**)
  - If **no ACK** was received, sender assumes that the **<u>frame or ACK</u>** has been destroyed and **resends** that frame after it **waits for a *random* amount of time**
  - If station fails to receive an ACK after repeated transmissions, **it gives up**
  - **Channel utilization or efficiency or Throughput** is the **percentage** of the transmitted frames that arrive **successfully** (without collisions) **or** the **percentage** of the **channel bandwidth** that will be used for transmitting frames without collisions
  - ALOHA Maximum channel utilization is **18%** (i.e, if the system produces **F frames/s**, then **0.18 * F** frames will arrive **successfully on average without the need of retransmission**).

Procedure for ALOHA protocol



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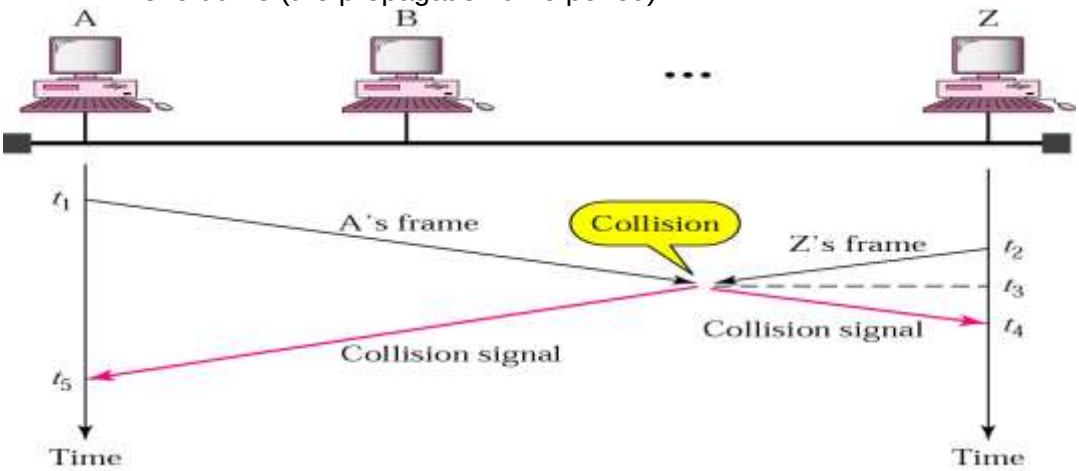
- Advantage of ALOHA protocols
  - A node that has frames to be transmitted can **transmit continuously** at the **full rate of channel (R bps)** if it is the only node with frames
  - Simple to be implemented
  - No master station is needed to control the medium
- Disadvantage
  - If (M) nodes want to transmit, many collisions can occur and the rate allocated for each node will **not be on average R/M bps**
  - This causes low channel utilization

Random Access – Carrier Sense Multiple Access (CSMA)

- To improve performance, avoid transmissions that are certain to cause collisions
- Based on the fact that in LAN propagation time is very small
- ➔ If a frame was sent by a station, All stations knows immediately so they can wait before start sending
  - ➔ A station with frames to be sent, should sense the medium for the presence of another transmission (carrier) before it starts its own transmission
- This can reduce the possibility of collision but it cannot eliminate it.
  - Collision can only happen when more than one station begin transmitting within a short time (the propagation time period)



Types of CSMA Protocols

Different CSMA protocols that determine:

- What a station should do when the medium is idle?
- What a station should do when the medium is busy?
    1. Non-Persistent CSMA
    2. 1-Persistent CSMA
    3. p-Persistent CSMA

Nonpersistent CSMA:

- A station with frames to be sent, should sense the medium
    1. If medium is idle, **transmit**; otherwise, go to 2
    2. If medium is busy, (**backoff**) wait a ***random* amount of time** and repeat **1**
- Non-persistent Stations are **deferential (respect others)**
- Performance:
    1. A random delay reduces probability of collisions because two stations with data to be transmitted will wait for different amount of times.
    2. Bandwidth is **wasted** if waiting time (backoff) is large because medium will remain idle following end of transmission even if one or more stations have frames to send

1-persistent CSMA

- To avoid idle channel time, 1-persistent protocol used
- Station wishing to transmit listens to the medium:
    1. If medium idle, transmit immediately;
    2. If medium busy, continuously listen until medium becomes idle; then transmit immediately with probability 1
- Performance
    - 1-persistent stations are selfish
    - If two or more stations becomes ready at the same time, collision guaranteed

P-persistent CSMA

- Time is divided to slots where each Time unit (slot) typically equals maximum propagation delay
- Station wishing to transmit listens to the medium:

1. If medium idle,
    - transmit with probability (p), OR
    - wait one time unit (slot) with probability (1 – p), then repeat 1.
2. If medium busy, continuously listen until idle and repeat step 1
3. Performance
    - Reduces the possibility of collisions like nonpersistent
    - Reduces channel idle time like 1-persistent

CSMA/CD (Collision Detection)

- *CSMA (all previous methods) has an inefficiency:*
    - If a collision has occurred, the channel is unstable until colliding packets have <u>been fully transmitted</u>
- *CSMA/CD (Carrier Sense Multiple Access with Collision*

Detection) overcomes this as follows:

- While transmitting, the sender is listening to medium for collisions.
- Sender stops transmission if collision has occurred reducing channel wastage .

CSMA/CD is Widely used for bus topology LANs (IEEE 802.3, Ethernet).

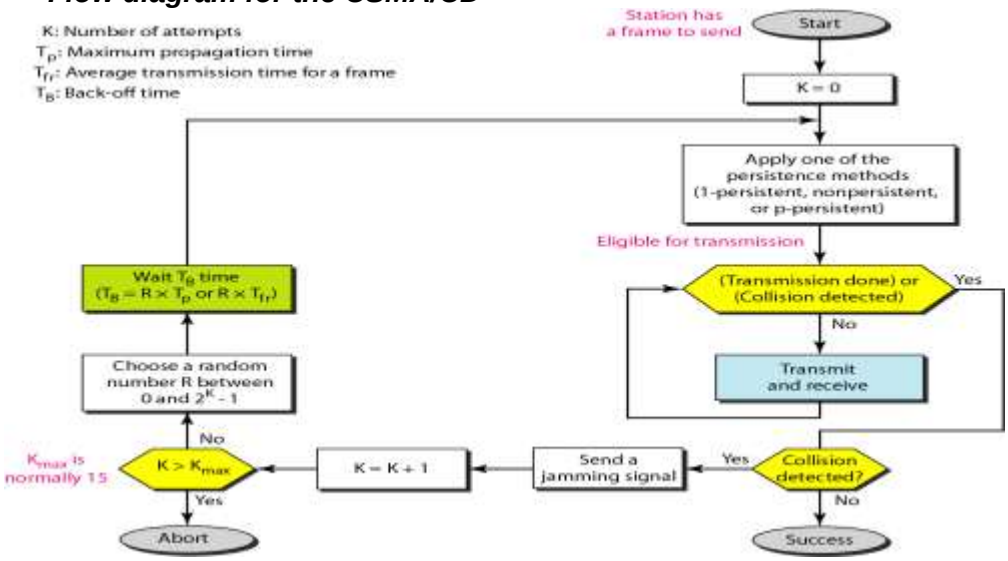
CSMA/CD Protocol

- Use one of the CSMA persistence algorithm ***(non-persistent, 1-persistent, p-persistent)*** for transmission
- If a collision is detected by a station during its transmission then it should do the following:
    - **Abort transmission** and
    - **Transmit** a ***jam signal*** (48 bit) to notify other stations of collision so that they will **discard the transmitted fra**me also to make sure that the collision signal will stay until detected by <u>the furthest station</u>

- After sending the ***jam signal***, **backoff (wait) for a *random*** amount of time, then
- Transmit the frame again

- Restrictions of CSMA / CD:
  - Packet **transmission time** should be **at least** as long as the time needed to detect a collision (2 * maximum propagation delay + *jam sequence* transmission time)
  - Otherwise, CSMA/CD does not have an advantage over CSMA

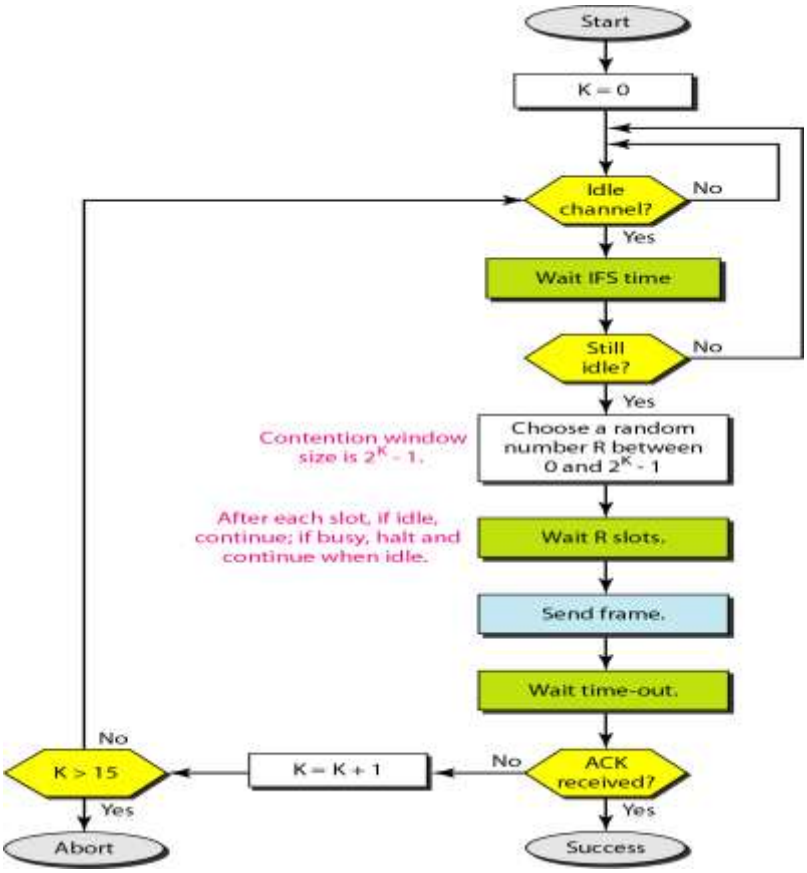
Flow diagram for the CSMA/CD



CSMA/CA (Collision Avoidance)

In CSMA/CA, the IFS can also be used to define the priority of a station or a frame. In CSMA/CA, if the station finds the channel busy, it does not restart the timer of the contention window; it stops the timer and restarts it when the channel becomes idle.

Flow diagram for CSMA/CA

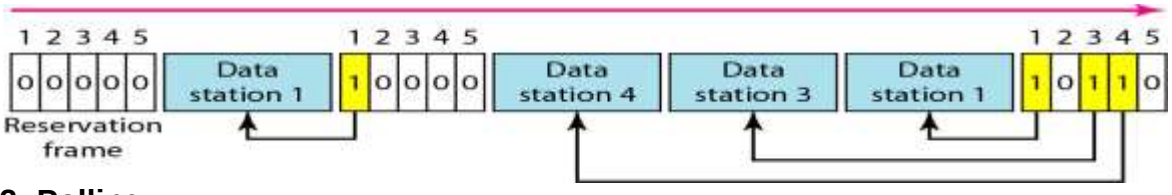


CONTROLLED ACCESS

- Provides **in order access** to shared medium so that every station has chance to transfer (**fair protocol**)
- ***Eliminates*** collision completely
- **Three methods** for controlled access:
  - Reservation
  - Polling
  - Token Passing

1-Reservation access method

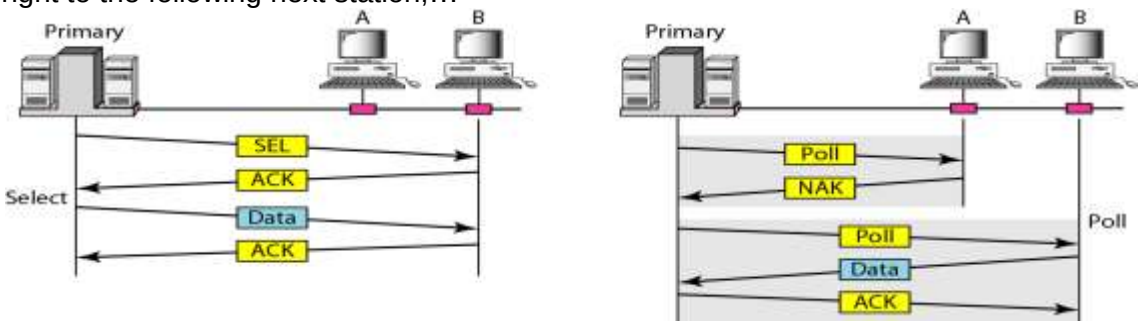
- Stations take turns transmitting a single frame at a full rate (R) bps
- Transmissions are organized into variable length cycles
- Each cycle begins with a reservation interval that consists of (N) minislots. One minislot for each of the N stations
- When a station needs to send a data frame, it makes a reservation in its own minislot.
- By listening to the reservation interval, every station knows which stations will transfer frames, and in which order.
- The stations that made reservations can send their data frames after the reservation frame.



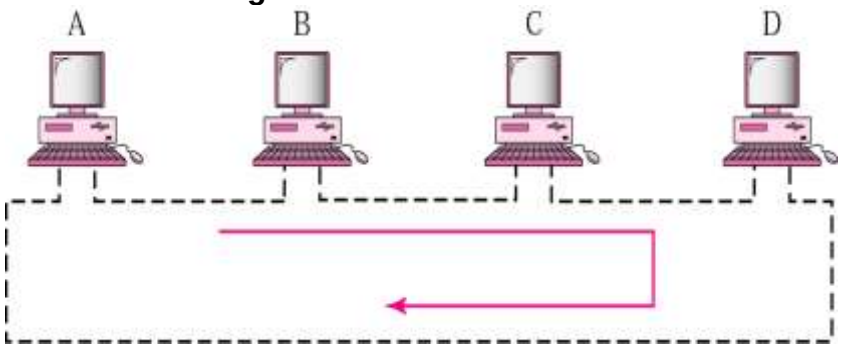
2- Polling

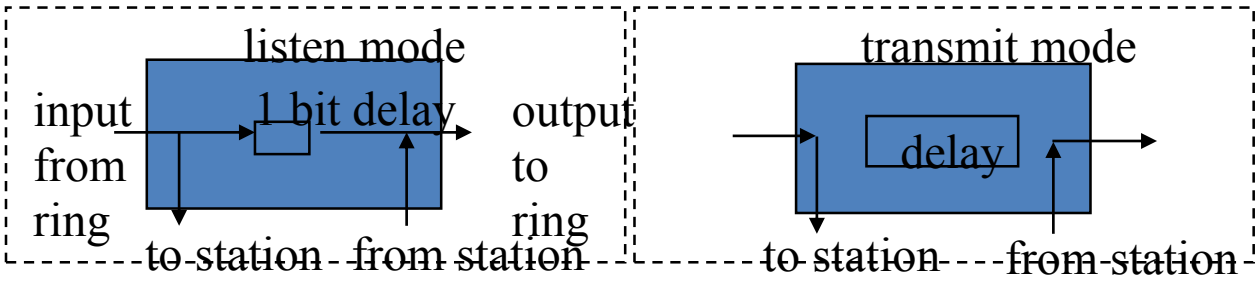
- Stations take turns accessing the medium
- Two models: Centralized and distributed polling
- Centralized polling
  - One device is assigned as primary station and the others as secondary stations
  - All data exchanges are done through the primary
  - When the primary has a frame to send it sends a select frame that includes the address of the intended secondary
  - When the primary is ready to receive data it send a Poll frame for each device to ask if it has data to send or not. If yes, data will be transmitted otherwise NAK is sent.
  - Polling can be done in order (Round-Robin) or based on predetermined order
- Distributed polling
  - No primary and secondary
  - Stations have a known polling order list which is made based on some protocol

station with the highest priority will have the access right first, then it passes the access right to the next station (it will send a pulling message to the next station in the pulling list), which will passes the access right to the following next station,…



3- Token-Passing network





Bits are copied to the output bits with a one bit delay **Bits are inserted by the station**

- Station Interface is in two states:

- Listen state: Listen to the arriving bits and check the destination address to see if it is its own address. If yes the frame is copied to the station otherwise it is passed through the output port to the next station.
- Transmit state: station captures a special frame called free token and transmits its frames. Sending station is responsible for reinserting the free token into the ring medium and for removing the transmitted frame from the medium.

CHANNELIZATION

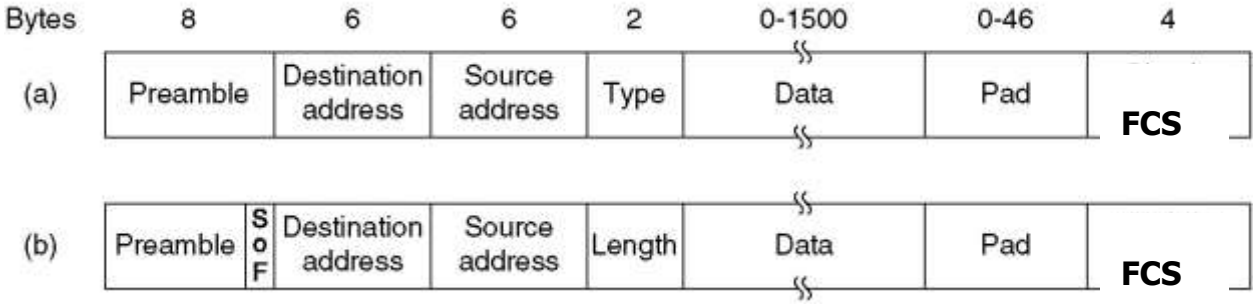
Channelization is a multiple-access method in which the available bandwidth of a link is shared in time, frequency, or through code, between different stations. In this section, we discuss three channelization protocols.

- **FDMA:** Frequency Division Multiple Access:
  - Transmission medium is divided into **M** separate frequency bands
  - Each station transmits **continuously** on the assigned band at an average rate of **R/M**
  - A node is **limited** to an average rate equal **R/M** (where M is number of nodes) even when it is **the only node with frame** to be sent
- **TDMA: Time Division Multiple Access**
  - The entire bandwidth capacity is a **single channel** with its capacity shared **in time** between **M** stations
  - A node must **always wait for its turn** until its slot time arrives even when it is the **only node** with frames to send
  - A node is limited to an average rate equal **R/M** (where M is number of nodes) even when it is the only node with frame to be sent
- **CDMA: Code Division Multiple Access**
  - In CDMA, **<u>one channel</u>** carries all transmissions **simultaneously**
  - Each station codes its data signal by a specific codes before transmission
  - The stations receivers use these codes to recover the data for the desired station

Local area Network: Ethernet.

Ethernet: It is a LAN protocol that is used in Bus and Star topologies and implements CSMA/CD as the medium access method

Ethernet Frame format:

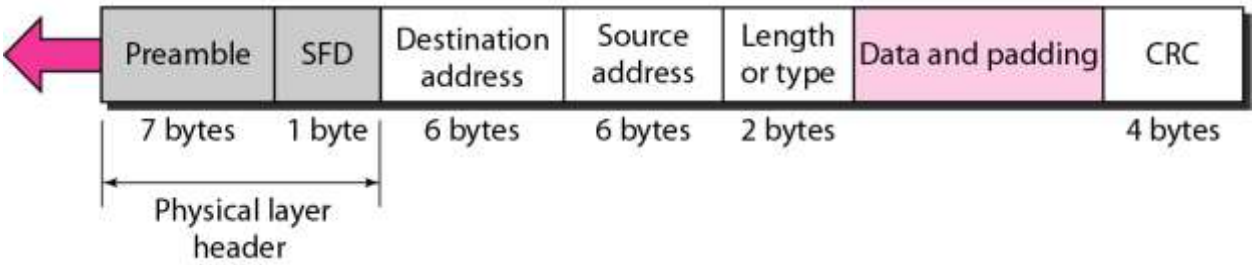


Frame formats. (a) DIX Ethernet , (b) IEEE 802.3.

802.3 MAC frame:

Preamble: 56 bits of alternating 1s and 0s.

SFD: Start frame delimiter, flag (10101011)



- Preamble:
  - 8 bytes with pattern 10101010 used to synchronize receiver, sender clock rates.
  - In IEEE 802.3, eighth byte is start of frame (10101011)
- Addresses: 6 bytes (explained latter)
- Type (DIX)
  - Indicates the type of the Network layer protocol being carried in the payload (data) field, mostly IP but others may be supported such as IP (0800), Novell IPX (8137) and AppleTalk (809B), ARP (0806))
  - Allow multiple network layer protocols to be supported on a single machine (multiplexing)
  - Its value starts at 0600h (=1536 in decimal)
- Length (IEEE 802.3): number of bytes in the data field.
  - Maximum 1500 bytes (= 05DCh)
- CRC: checked at receiver, if error is detected, the frame is discarded
  - CRC-32
- Data: carries data encapsulated from the upper-layer protocols
- Pad: Zeros are added to the data field to make the minimum data length = 46 bytes

- In IEEE 802.3 Ethernet Data link layer is split into two sublayers:
  - Bottom part: MAC
    - The frame is called IEEE 802.3
    - Handles framing, MAC addressing, Medium Access control
    - Specific implementation for each LAN protocol
      - Defines CSMA/CD as the access method for Ethernet LANs and Token passing method for Token Ring.
    - Implemented in hardware
  - Top part: LLC (Logical Link Control)
    - The sub frame is called IEEE 802.2
    - Provides error and flow control if needed
    - It makes the MAC sub layer transparent
      - Allows interconnectivity between different LANs data link layers
    - Used to multiplex multiple network layer protocols in the data link layer frame

Ethernet *address*:

- Six bytes = 48 bits
- Flat address not hierarchical
- Burned into the NIC ROM
- First three bytes from left specify the vendor. Cisco 00-00-0C, 3Com 02-60-8C and the last 24 bit should be created uniquely by the company
- Destination Address can be:
  - Unicast: second digit from left is even (one recipient)

- Multicast: Second digit from left is odd (group of stations to receive the frame – conferencing applications)
- Broadcast (ALL ones) (all stations receive the frame)

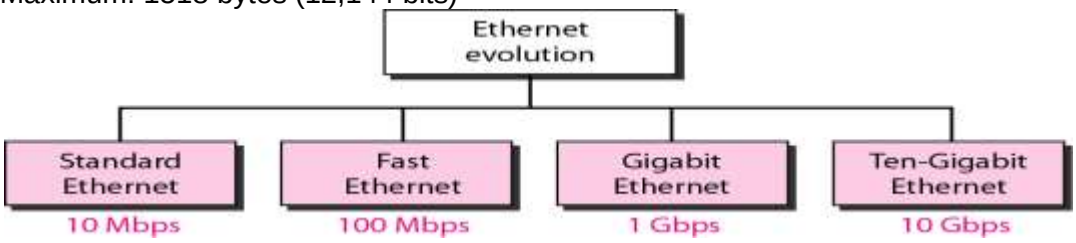
- Source address is always Unicast

The least significant bit of the first byte defines the type of address. If the bit is 0, the address is unicast; otherwise, it is multicast. The broadcast destination address is a special case of the multicast address in which all bits are 1s.

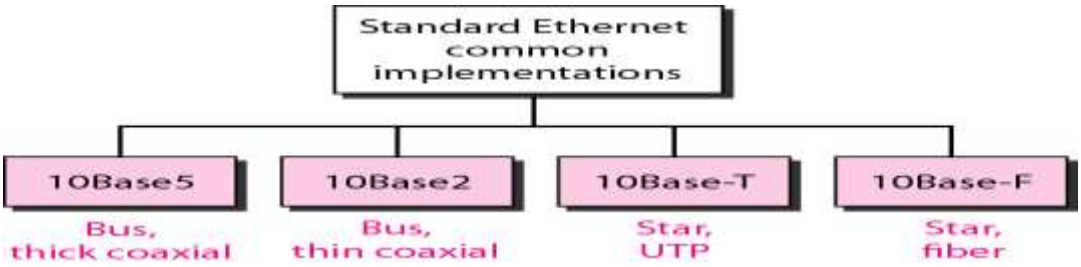
Frame length:

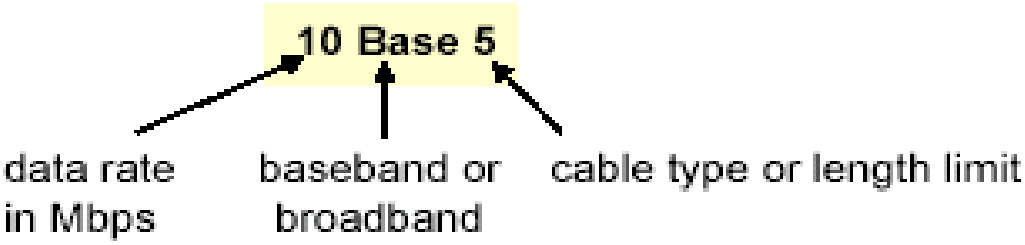
Minimum: 64 bytes (512 bits)

Maximum: 1518 bytes (12,144 bits)

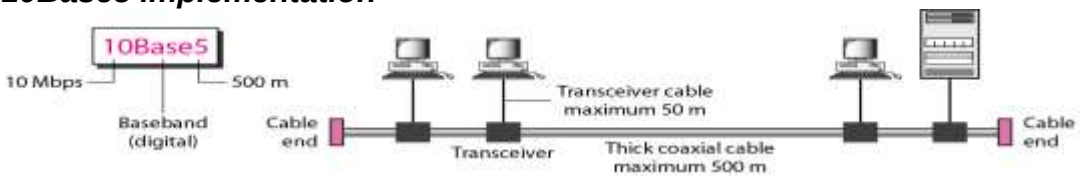


Standard Ethernet:

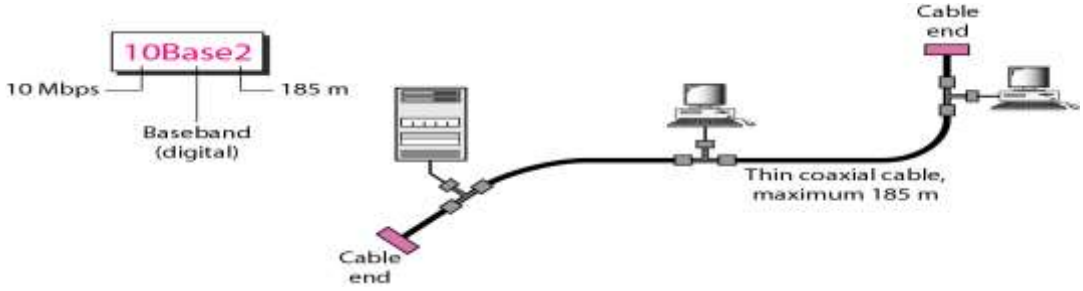




10Base5 implementation



10Base2 implementation



10BaseT

Uses twisted pair Cat3 cable

- Star-wire topology

• A hub functions as a repeater with additional functions
• Fewer cable problems, easier to troubleshoot than coax
• Cable length at most 100 meters

Summary of Standard Ethernet implementations

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

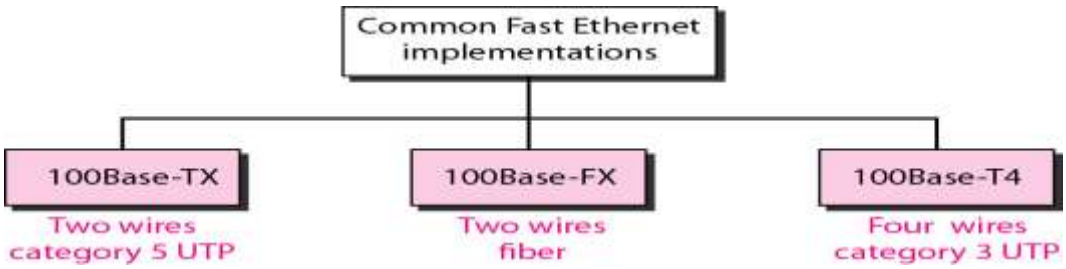
Switched Ethernet:

- Switches forward and filter frames based on LAN addresses
  - It's not a bus or a router (although simple forwarding tables are maintained)
- Very scalable
  - Options for many interfaces
  - Full duplex operation (send/receive frames simultaneously)
- Connect two or more "segments" by copying data frames between them
  - Switches only copy data when needed
    - key difference from repeaters
- Higher link bandwidth
  - Collisions are completely avoided
- Much greater aggregate bandwidth
  - Separate segments can send at once

Fast Ethernet

- 100 Mbps transmission rate
- same frame format, media access, and collision detection rules as 10 Mbps Ethernet
- can combine 10 Mbps Ethernet and Fast Ethernet on same network using a *switch*
- media: twisted pair (CAT 5) or fiber optic cable (no coax)
- Star-wire topology
- Similar to 10BASE-T

Name	Cable	Max. segment	
100Base-T4	Twisted pair	100 m	
100Base-TX	Twisted pair	100 m	**CAT 3**
100Base-FX	Fiber optics	2000 m	**CAT 5**



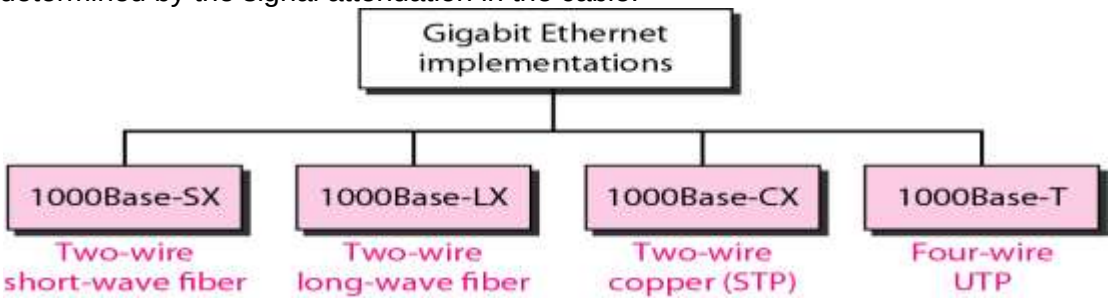
Characteristics	*100Base-TX*	*100Base-FX*	*100Base-T4*
Media	Cat 5 UTP or STP	Fiber	Cat 4 UTP
Number of wires	2	2	4
Maximum length	100 m	100 m	100 m
Block encoding	4B/5B	4B/5B	
Line encoding	MLT-3	NRZ-I	8B/6T

Gigabit Ethernet

- Speed 1Gpbs
- Minimum frame length is 512 bytes
- Operates in full/half duplex modes mostly full duplex

Name	Cable	Max. segment	Advantages
1000Base-SX	Fiber optics	550 m	Multimode fiber (50, 62.5 microns)
1000Base-LX	Fiber optics	5000 m	Single (10 μ) or multimode (50, 62.5 μ)
1000Base-CX	2 Pairs of STP	25 m	Shielded twisted pair
1000Base-T	4 Pairs of UTP	100 m	Standard category 5 UTP

In the full-duplex mode of Gigabit Ethernet, there is no collision; the maximum length of the cable is determined by the signal attenuation in the cable.



10Gbps Ethernet

- Maximum link distances cover 300 m to 40 km
- Full-duplex mode only
- No CSMA/CD
- Uses **optical fiber** only

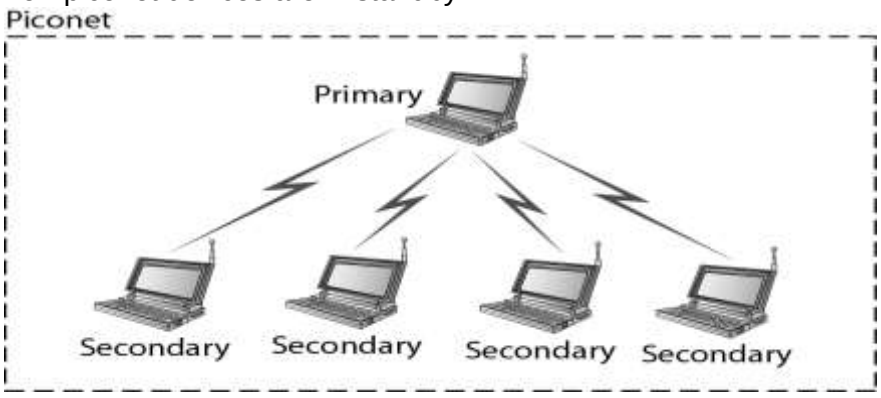
WIRELESS LAN

Blue Tooth:

- IEEE has defined the specifications for a wireless LAN, called IEEE 802.11, which covers the physical and data link layers.
- Bluetooth wireless technology is an **open** specification for a **low-cost**, **low-power**, short-range radio technology for **ad-hoc** wireless communication of **voice and data anywhere** in the world.
- Bluetooth is a wireless LAN technology designed to connect devices of different functions such as telephones, notebooks, computers, cameras, printers, coffee makers, and so on. A Bluetooth LAN is an ad hoc network, which means that the network is formed spontaneously.

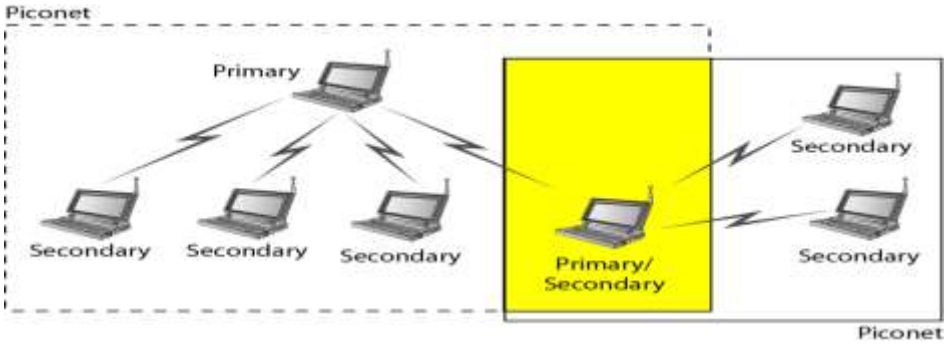
Piconet:

- All devices in a piconet hop together
  - Master gives slaves its clock and device ID
- Non-piconet devices are in standby



Scatternet:

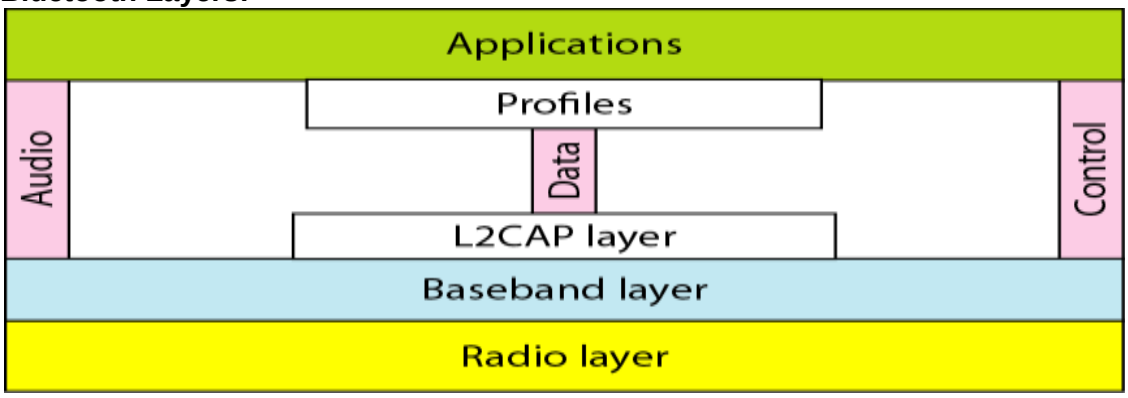
- Devices can be slave in one Piconet and master of another



Physical links:

- Synchronous Connection Oriented (SCO)
  - Support symmetrical, circuit-switched, point-to-point connections
  - Typically used for voice traffic.
  - Data rate is 64 kbit/s.
- Asynchronous Connection-Less (ACL)
  - Support symmetrical and asymmetrical, packet-switched, point-to-multipoint connections.
  - Typically used for data transmission .
  - Up to 433.9 kbit/s in symmetric or 723.2/57.6 kbit/s in asymmetric

Bluetooth Layers:



Bluetooth Radio: specifics details of the air interface, including frequency, frequency hopping, modulation scheme, and transmission power.

- the lowest defined layer of the Bluetooth specification
- operating in the 2,4 GHz ISM Band
- accomplishes spectrum spreading by frequency hopping (FHSS) from 2.402 GHz to 2.480 GHz
- 3 different power classes
- Power Class1: long range (100m,100mW)
- Power Class2: mid range (10m,1-2,5mW)
- Power Class3: short range (0.1-10m,1mW)
- signal strength adjustment

Baseband: concerned with connection establishment within a piconet, addressing, packet format, timing and power control.

- the physical layer of the Bluetooth that provides
- Error correction
- Flow control
- Hopping sequence
- Security
- hopping through 79 channels
- data is divided in packets
- access code: e.g. timing synchronization
- header: e.g. packet numbering, flow control, slave address
- payload: voice, data or both
- Connection Modes
- describes the set of rules by which all bluetooth devices must abide in order to establish a link a communicate with one another
- STANDBY : not connected in a piconet
- ACTIVE : active participation on the channel
- Power Saving Modes
- SNIFF : slave listens to the channel at a reduced rate (decreasing of duty cycle) least power efficient
- HOLD : data transfer is held for a specific time period, medium power efficient
- PARK : synchronized to the piconet but does not participate in traffic

Audio:

- two codecs: PCM and CVSD
- both at 64kbit/s
- synchronous connection oriented(SCO) links
- time-critical
- no retransmission
- errors appear as background noise

Link manager protocol (LMP): establishes the link setup between Bluetooth devices and manages ongoing links, including security aspects (e.g. authentication and encryption), and control and negotiation of baseband packet size.

- provides authentication, link setup and link configuration including power surveillance
- takes place as a service provider
- communication with LM PDUs (protocol data units)

Logical link control and adaptation protocol (L2CAP): adapts upper layer protocols to the baseband layer. Provides both connectionless and connection-oriented services.

- provides a connection-oriented and connectionless service to upper layer
- protocols with quality-of-service functions using multiplexing, segmentation and reassembly
- two link types defined in Baseband layer:
- 1. SCO (synchronous connection-oriented)
- 2. ACL (asynchronous connection-less)
- BUT ONLY ACL is supported by L2CAP (SCO not planned)

Service discovery protocol (SDP): handles device information, services, and queries for service characteristics between two or more Bluetooth devices.

- discovers which services are available
- identifies the characteristics of the services
- uses a request/response model where each transaction consists of one request protocol data unit (PDU) and one response PDU
- SDP is used with L2CAP
- is optimized for the dynamic nature of bluetooth
- SDP does not define methods for accessing services

Host Controller Interface (HCI): provides an interface method for accessing the Bluetooth hardware capabilities. It contains a command interface, which acts between the Baseband controller and link manager.

- provides a command interface to baseband controller and link manager
- also to hardware status, control and event register
- Bluetooth defined Host Controller Transport Layers:
- UART (HCI over serial interface)
- RS232(HCI over serial interface)
- USB(HCI over USB interface e.g. USB dongle)

RFCOMM (Radio Frequency Communication):

- Provides emulation of serial ports
- Supports up to 60 simultaneous connections
- Differentiates between two device types:
- Type 1: communication end points (e.g. printer or headsets)
- Type 2: devices which are part of communication (e.g. modems)

VIRTUAL CIRCUIT SWITCHING

Circuit-Switching:

- A circuit-switched network consists of a set of switches connected by physical links.
- A connection between two stations is a dedicated path made of one or more links
- each connection uses only one dedicated channel on each link
- Each link is normally divided into n channels by using FDM or TDM.
- The link can be permanent (leased line) or temporary (telephone)
- Switching take place at physical layer
- Before any data can be sent, an end-to-end circuit must be established
- This circuit is maintained for the duration of the transfer of all the data
- The data can be digital or analog and the signal can be either type as well
- Connection is usually full-duplex
- Is inefficient – channel capacity is dedicated for the duration of the connection

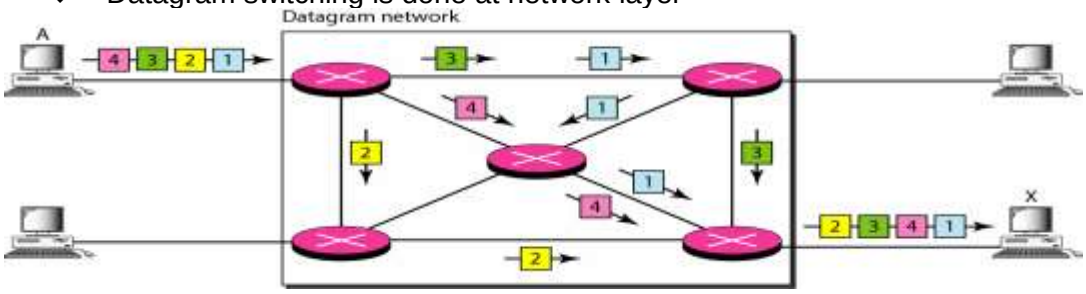
- Example – Pubic telephone system
- Resources
  - Such as bandwidth in FDM and time slot in TDM
  - Switch buffer
  - Switch processing time
  - Switch I/O ports
- Data transferred are not packetized, continuous flow
- No addressing involved during data transfer

Packet Switching:

- Station breaks long message into packets
- Packets sent one at a time to the network
- Very much like message switching
- Principal external difference is that the length of the message found internally has a maximum length
- A typical maximum length is several thousand bits
- Messages above the maximum length are divided up into smaller units and sent out one at a time
- These smaller units are called packets
- Packets, unlike messages, are typically not filed at the intermediate nodes
- Packets are handled in two ways
  - Datagram
  - Virtual circuit

Datagram Networks:

- In datagram approach each packet is treated independently with no reference to packets that have gone before. No connection is set up.
- The packets may take different paths to the destination
- The packets might arrive in a different sequence from the order in which they were sent
- The packets may have to be reordered at the destination
- Packets may go missing
- Up to receiver to re-order packets and recover from missing packets
- More processing time per packet per node
- Size of the packet depends on the protocol and network
- Packets switched networks are connectionless, hence no resource allocation
- Connectionless means the switch does not keep information about the connection state.
- Datagram switching is done at network layer

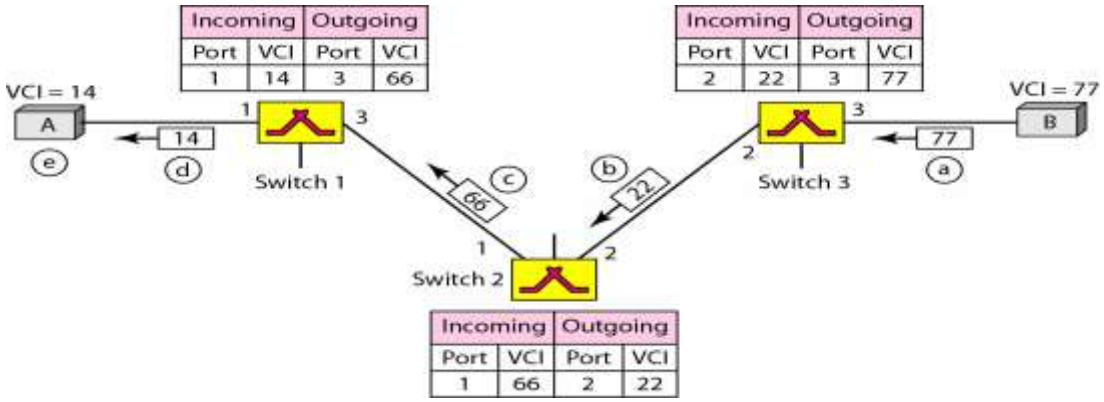


Virtual-Circuit Networks:

- In the Virtual Circuit approach a pre-planned route is established before any packets are sent.
- There is a call set up before the exchange of data (handshake).
- A logical connection is established before any packets are sent
- All packets follow the same path through the network
- This does not mean that there is a dedicated path, as in circuit switching
- All packets follow the same route and therefore arrive in sequence.

- Each packet contains a virtual circuit identifier instead of destination address
- More set up time
- No routing decisions required for each packet - Less routing or processing time.
- Susceptible to data loss in the face of link or node failure
- Clear request to drop circuit
- Not a dedicated path

Source-to-Destination data transfer



Virtual circuits can be either permanent, called Permanent virtual Circuits (PVC), or temporary, called Switched Virtual Circuits (SVCs).

Permanent Virtual Circuit (PVC)

A Permanent Virtual Circuit (PVC) is a virtual circuit that is permanently available to the user.

PVC is defined in advance by a network manager. The actual identifier used for data transfer is virtual circuit identifier (VCI). If permanent, an outgoing VCI is given to the source, and an incoming VCI is given to the destination.

The source always uses this out going VCI to send frames to this particular destination.

The destination knows that the frame is coming from that particular source if the frame carries the corresponding incoming VCI.

Once a communication session is complete, the virtual circuit is disabled.

Switched Virtual Circuit (SVC)

A switched virtual circuit is an automatically and temporarily created logical path with aid of some switch control for a communication session switch control for a communication session

Once a communication session is complete, the virtual circuit is disabled.

FRAME RELAY:

- **X.25**
  - Interface between attached station and link to node
  - Data terminal equipment DTE (user equipment)
  - Data circuit terminating equipment DCE (node)
  - Uses physical layer specification X.21
  - Interface between host and packet switched network
  - Almost universal on packet switched networks and packet switching in ISDN
  - Defines three layers
  - Physical
  - Link
  - Packet
- **Frame Relay:**
  - Frame Relay (FR) is a high-performance WAN protocol that operates at the physical and data link layers of the OSI reference model.

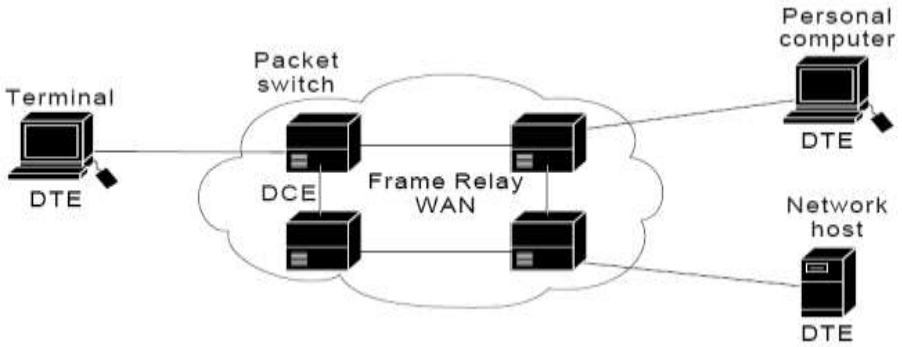
- ❖ FR originally was designed for use across Integrated Service Digital Network (ISDN) interfaces.
- ❖ Today, it is used over a variety of other network interfaces as well.
- ❖ FR is an example of a packet-switched technology.
- ❖ Packet-switched networks enable end stations to dynamically share the network medium and the available bandwidth.
- ❖ Packet switching system with low overhead
- ❖ Assumes very reliable high-quality physical network
- ❖ Developed for use in ISDN networks
- ❖ Used widely in a variety of private and public networks which are not ISDN
- ❖ No error checking and acknowledgment at the data link layer
- ❖ All error checking is left to the protocols at the network and transport layers
- ❖ operates at only the physical and data link layers

Frame Relay vs. X.25

- Frame Relay is a Layer 2 protocol suite, X.25 provides services at Layer 3
- Frame Relay offers higher performance and greater transmission efficiency than X.25

Frame Relay Devices

- data terminal equipment (DTE)
  - ☐ terminating equipment for a specific network
  - ☐ typically are located on the premises of a customer
  - ☐ Examples: terminals, personal computers, routers, and bridges
- data circuit-terminating equipment (DCE)
  - ☐ carrier-owned internetworking devices
  - ☐ to provide clocking and switching services in a network
  - ☐ actually transmit data through the WAN



Frame Relay Devices

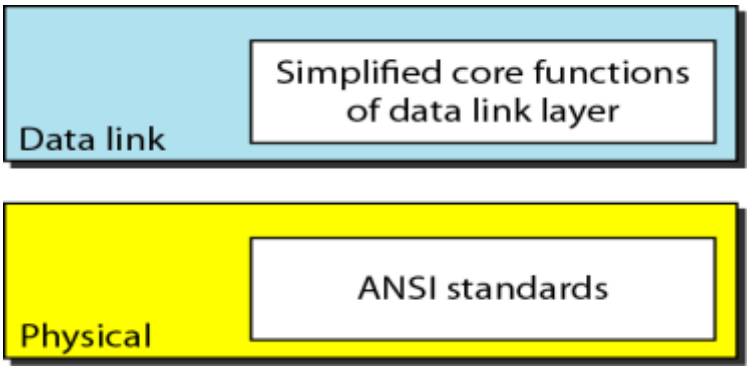
Frame Relay Virtual Circuits:

- provides connection-oriented data link layer communication
- a logical connection between two data terminal equipment across a Frame Relay packet-switched network
- provide a bi-directional communications path from one DTE device to another
- Switched virtual circuits (SVCs)
  - temporary connections requires sporadic data transfer between DTE devices across the Frame Relay network
    - *Call Setup*
    - *Data Transfer*
    - *Idle*
    - *Call Termination*
- Permanent Virtual Circuits (PVCs)
  - used for frequent and consistent data transfers between DTE devices across the Frame Relay network
    - *Data Transfer*

- *Idle*

Frame Relay Layers

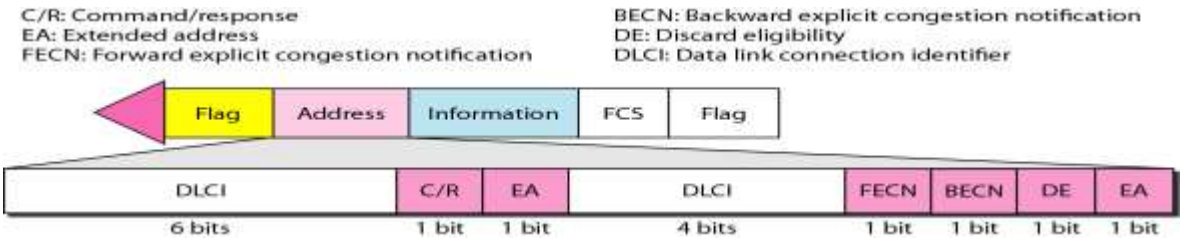
- Frame Relay operates only at the physical and data link layers.
- Eliminates all of the network layer functions and a portion of conventional data link layer functions.



- Physical Layers: no specific protocol is defined for the physical layer. It is left to the implementers.

Data Link Layer:

- employ a simplified version of HDLC - HDLC fields for extensive error checking and flow control not needed.
- Addressing and control fields combined into one field.



- Frame fields
  - Addressing(DLCI) fields: two parts (6bits, 4bits). A part of the 16-bit data link connection identifier.
  - Command/Response(C/R): allows upper layers to identify a frame as either a command or a response.
  - Extended address (EA):EA=0 another address byte follows, EA=1 the current byte is the final one.
  - Forward explicit congestion notification (FECN): initiated when a DTE device sends Frame Relay frames into the network. When the frames reach the destination DTE device, the frame experienced congestion in the path from source to destination. flow-control may be initiated, or the indication may be ignored.
  - Backward explicit congestion notification (BECN): DCE devices set the value of the BECN bit to 1 in frames traveling in the opposite direction, informs the receiving DTE device that a particular path through the network is congested. flow-control may be initiated, or the indication may be ignored.
  - Discard eligibility (DE) : (DE) bit is used to indicate that a frame has lower importance than other frames. When the network becomes congested, DCE devices will discard frames with the DE bit

Frame Relay Operation:

- Transmission is based on permanent virtual circuit
- DLCI identifies a permanent virtual circuit that is set up when the system is put in place

- Replay
  - Routing information is included in the destination information.
  - The path form point A to point D always passes through the same node.
  - The functions of routing and switching can be handled by the data link layer.
  - Frame relay (frame switching) occurs at the date link layer where the transmission unit is the frame
  - Packet switching occurs at the network layer where the transmission unit is the packet
- Switching
  - Two operations of a switch
    - checks a frame for errors using the FCS field: if an error, discard it
    - compares the DLCI to an entry in a switch table and find an outgoing port for the PVC identified by the DLCI

- **Congestion Control**
  - does not solve the problem, but does provide ways to lessen the probability of its occurrence
  - A switch in a PVC warns its downstream switches and destination by turning on the FECN bit
  - The receiver, in turn, set BECN to warn upstream switches and the sender that the link is congested and to send frames more slowly.
  - This option can not be used unless the channel is either full- or half-duplex and the receiver is sending its own data or acknowledgments to the sender

Frame Relay Implementation

- The most likely implementation:

  Used as a WAN backbone to connect a number of LANs using T-1 links
  - Frame relay assembler/disassembler(FRADs): assembles and disassembles packets coming from other protocols to allow them to be carried by frame relay frames

Asynchronous Transfer Mode (ATM)

- ATM is *Asynchronous Transfer Mode*.
- ATM is a connection-oriented, high-speed, low-delay switching and transmission technology that uses short and fixed-size packets, called cells, to transport information.
- ATM is originally the transfer mode for implementing Broadband ISDN (B-ISDN) but it is also implemented in non-ISDN environments where very high data rates are required
- a streamlined packet transfer interface
- similarities to packet switching
  - transfers data in discrete chunks
  - supports multiple logical connections over a single physical interface
- ATM uses fixed sized packets called cells
- with minimal error and flow control
- data rates of 25.6Mbps to 622.08Mbps

ATM OVERVIEW

- Used in both WAN and LAN settings
- Signaling (connection setup) Protocol:

- Packets are called *cells* (53 bytes)
  - 5-byte header + 48-byte payload
- Commonly transmitted over SONET
  - other physical layers possible
- Connections can be switched (SVC), or permanent (PVC).
- ATM operates on a best effort basis.
- ATM guarantees that cells will not be disordered.
- Two types of connections:
  - Point-to-point
  - Multipoint (Multicast)
- Four Types of Services:
  - CBR (Constant Bit Rate)
  - VBR (Variable Bit Rate)
  - ABR (Available Bit Rate) Flow Control, Rate-based, Credit- based
  - UBR (Unspecific Bit Rate) No Flow control.

ATM Characteristics

- No error protection or flow control on a link-by-link basis.
- ATM operates in a connection-oriented mode.
- The header functionality is reduced.
- The information field length is relatively small and fixed.
- All data types are the same

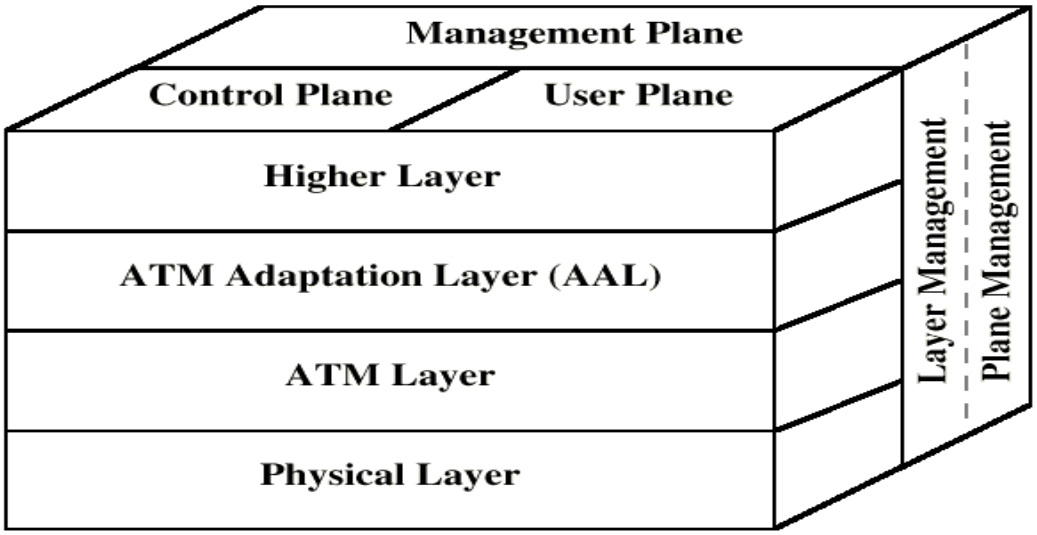
ATM NETWORKS

- Public ATM Network:
  - Provided by public telecommunications carriers (e.g., AT&T, MCI WorldCom, and Sprint)
  - Interconnects private ATM networks
  - Interconnects remote non-ATM LANs
  - Interconnects individual users
- Private ATM Network:
  - Owned by private organizations
  - Interconnects low speed/shared medium LANs (e.g., Ethernet, Token Ring, FDDI) as a backbone network
  - Interconnects individual users as the front-end LAN for high performance or multimedia applications

How ATM Works?

- ATM is connection-oriented -- an end-to-end connection must be established and routing tables setup prior to cell transmission
- Once a connection is established, the ATM network will provide end-to-end Quality of Service (QoS) to the end users
- All traffic, whether voice, video, image, or data is divided into 53-byte cells and routed in sequence across the ATM network
- Routing information is carried in the header of each cell
- Routing decisions and switching are performed by hardware in ATM switches
- Cells are reassembled into voice, video, image, or data at the destination

ATM Protocol Architecture



Reference Model Planes

- user plane
  - provides for user information transfer
- control plane
  - call and connection control
- management plane
  - plane management
    - whole system functions
  - layer management
    - Resources and parameters in protocol entities

ATM Logical Connections

- virtual channel connections (VCC)
  - analogous to virtual circuit in X.25
- basic unit of switching between two end users
  - full duplex
  - fixed size cells
- also for
  - user-network exchange (control)

ATM VIRTUAL CIRCUITS

- VC transport: cells carried on VC from source to destination
  - call setup, teardown for each call *before* data can flow
  - each packet carries VC identifier (not destination ID)
  - *every* switch on source-dest path maintain "state" for each passing connection
  - link,switch resources (bandwidth, buffers) may be *allocated* to VC: to get circuit-like perf.
  - Permanent VCs (PVCs)
  - long lasting connections
  - typically: "permanent" route between to IP routers
- Switched VCs (SVC):
  - dynamically set up on per-call basis
- The virtual channel (VC) is the fundamental unit of transport in a B-ISDN. Each ATM cell contains an explicit label in its header to identify the virtual channel.
  - a Virtual Channel Identifier (VCI)
  - a Virtual Path Identifier (VPI)

- A *virtual channel (VC)* is a communication channel that provides for the transport of ATM cells between two or more endpoints for information transfer.
- A Virtual Channel Identifier (VCI) identifies a particular VC within a particular VP over a UNI or NNI.
- A specific value of VCI has no end-to-end meaning.

ATM Protocol Layer

- **Physical Layer:** The lowest layer in the ATM protocol. It describes the physical transmission media. We can use shielded and unshielded twisted pair, coaxial cable, and fiber-optic cable.
- **ATM Layer:** It performs all functions relating to the routing and multiplexing of cells over VCs. It generates a header to the segment streams generated by the AAL. Similarly, on receipt of a cell streams, it removes the header from the cell and pass the cell contents to the AAL protocol. To perform all these functions, the ATM layer maintains a table which contains a list of VCIs.
- **ATM Adaptation Layer**: Top layer in the ATM protocol Model. It converts the submitted information into streams of 48-octet segments and transports these in the payload field of multiple ATM cells. Similarly, on receipt of the stream of cells relating to the same call, it converts the 48-octet information field into required form for delivery to the particular higher protocol layer. Currently five service types have been defined. They are referred to as AAL1-5. AAL1 and AAL2 are connection oriented. AAL1 provides a constant bit rate (CBR) service, where as AAL2 provides a variable bit rate (VBR) service. Initially, AAL 3 was defined to provide connection oriented and VBR service. Later, this service type was dropped and it is now merged with AAL 4. Both AAL ¾ and AAL 5 provide a similar connectionless VBR service.
- "adapts" upper layers (IP or native ATM applications) to ATM layer below
- AAL exists only in end systems, not in switches
- AAL layer segment (header/trailer fields, data) fragmented across multiple ATM cells
- AAL Services
  - Handle transmission errors
  - Segmentation/reassembly (SAR)
  - Handle lost and misinserted cell conditions
  - Flow control and timing control

AAL SUBLAYERS

- AAL layer has 2 sublayers:
  - Convergence Sublayer (CS)
  - Supports specific applications using AAL
  - manages the flow of data to and from SAR sublayer
  - Timing and cell loss recovery
- Segmentation and Reassembly Layer (SAR)
  - Packages data from CS into cells and unpacks at other end

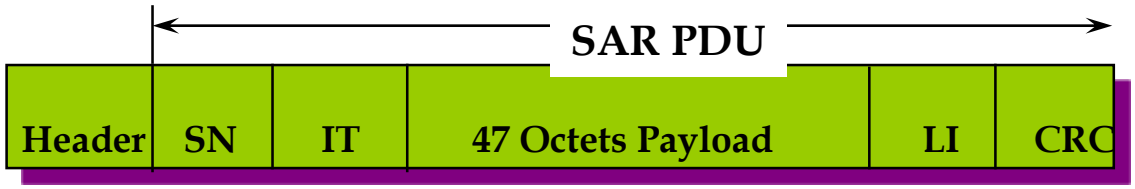
AAL Types:

AAL 1 (Constant Bit Rate) Functions

- Constant-bit-rate source

- SAR simply packs bits into cells and unpacks them at destination
- Emulation of DS1 and DS3 Circuits
- Distribution with forward error correction
- Handle cell delay for constant bit rate
- Transfer timing information between source and destination
- Transfer structure information (structure pointer)
- Provide indication of unrecoverable lost or error information

AAL 2 Protocol Data Unit (PDU)

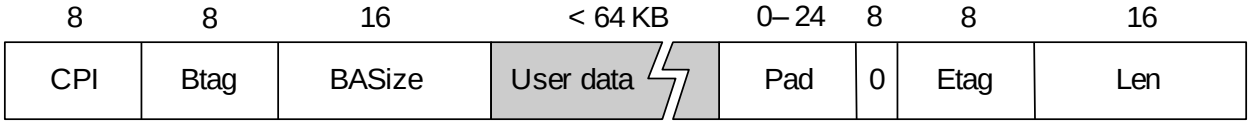


- SN: Sequence number
- IT: Information Type:BOM,COM,EOM,SSM
- Length Indicator

BOM: beginning of message
COM: continuation of message
EOM end of message

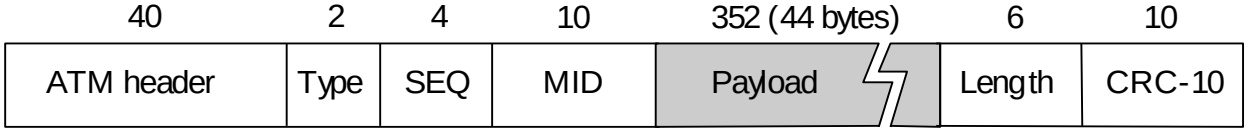
AAL 3/4

- Convergence Sublayer Protocol Data Unit (CS-PDU)



- CPI: commerce part indicator (version field)
- Btag/Etag:beginning and ending tag
- BAsize: hint on amount of buffer space to allocate
- Length: size of whole PDU

Cell Format



- Type
  - BOM: beginning of message
  - COM: continuation of message
  - EOM end of message
- SEQ: sequence of number
- MID: message id
- Length: number of bytes of PDU in this cell

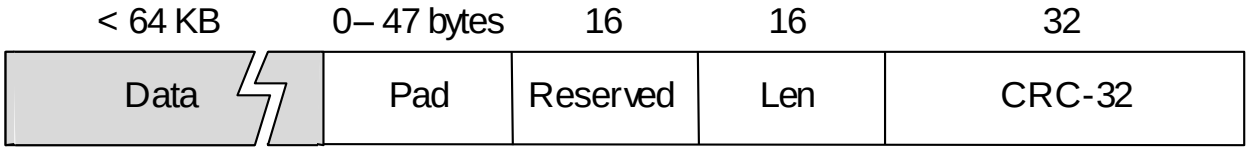
AAL 5 PDU Structure

- is used to transport IP datagrams over ATM networks.
- The Simple and Efficient Adaptation Layer (SEAL), attempts to reduce the complexity and overhead of AAL 3/4.
- It eliminates most of the overhead of AAL 3/4.
- AAL 5 comprise a convergence sublayer and a SAR sublayer, although the SAR is essentially null.
- Streamlined transport for connection oriented protocols
  - Reduce protocol processing overhead

- Reduce transmission overhead
- Ensure adaptability to existing transport protocols

AAL5

- CS-PDU Format



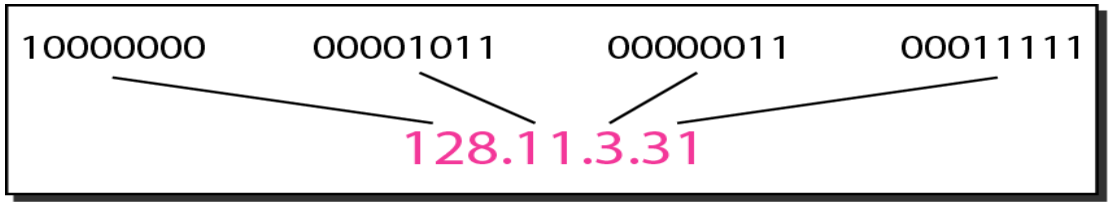
  - pad so trailer always falls at end of ATM cell
  - Length: size of PDU (data only)
  - CRC-32 (detects missing or misordered cells)
- Cell Format
  - end-of-PDU bit in Type field of ATM header

MODULE III

Internetworking:

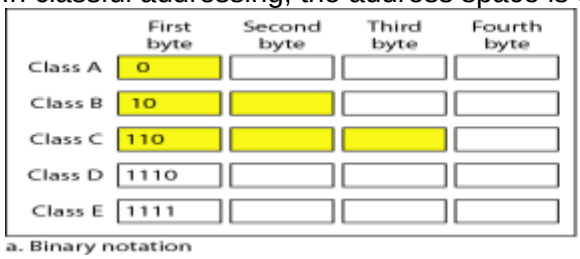
Each network interface on the Internet as a unique global address, called the IP address. An IP address- is 32 bits long. It encodes a network number and a host number. The address space of IPv4 is 2^{32} or 4,294,967,296.

IP addresses are written in a dotted decimal notation:

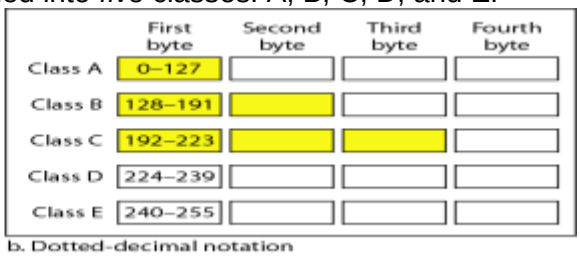


Classful Addressing:

In classful addressing, the address space is divided into five classes: A, B, C, D, and E.



a. Binary notation



b. Dotted-decimal notation

- **Class A networks**
  –First octet values range from 1 through 127
  –First octet starts with bit 0
  –Network mask is 8 bits, written /8 or 255.0.0.0
  –1.0.0.0 through 127.0.0.0 are class A networks with 16777214 hosts each.
- **Class B networks**
  -First octet values range from 128 through 191
  –First octet starts with binary pattern 10
  –Network mask is 16 bits , written /16 or 255.255.0.0
  –128.0.0.0 through 191.255.0.0 are class B networks, with 65534 hosts each.
- **Class C networks**
  First octet values range from 192 through 223
  –First octet starts with binary pattern 110
  –Network mask is 24 bits , written /24 or 255.255.255.0
  –192.0.0.0 through 223.255.255.0 are class C networks, with 254 hosts each.
- **Class D addresses**
  – First octet values range from 224 through 239
  – First octet starts with binary pattern 1110
  – Class D addresses are multicast addresses.
- **Class E addresses**
  – Reserved for future use.

The mask

•The network portion of the address is separated from the host portion of the address by a mask.
•The mask simply indicates how many bits are used for the network portion, leaving the remaining bits for the host portion.
•A 24-bit mask indicates that the first 24 bits of the address are network bits, and the remaining 8 bits are host bits.
•A 16-bit mask indicates that the first 16 bits of the address are network bits, and the remaining 16 bits are host bits.

Subnet Mask:

Subnet masks are used to make classful networks more manageable and efficient, by creating smaller subnets and reducing the number of host addresses per subnet to what is actually required.

• Subnet masks were first used on class boundaries.
• Example

– Take class A network 10.0.0.0 with network mask 255.0.0.0.
– Add additional 8 subnet bits to network mask.
– New subnet mask is 255.255.0.0.
– New subnets are 10.0.0.0, 10.1.0.0, 10.2.0.0, and so on with 65534 host addresses per subnet. Still too many hosts per subnet.
• Example
– Take class A network 10.0.0.0 with network mask 255.0.0.0.
– Add additional 16 subnet bits to network mask.
– New subnet mask is 255.255.255.0
– New subnets are 10.0.0.0, 10.0.1.0, 10.0.2.0, …, 10.1.0.0, 10.1.1.0, 10.1.2.0, …, 10.2.0.0, 10.2.1.0, 10.2.2.0, and so on with 254 host addresses per subnet.

ROUTING:

Static routing:

- Static routes are manually entered into a router or host.
- An administrator must know the internetwork layout and the paths that exist between networks.
- Then the administrator must program each router in the internetwork with the proper routes to get from any given network to any other network.
- The hosts obtain their routes manually or via DHCP.

Dynamic routing

- Dynamic routes are routes learned via one or more routing protocols.
- Routing protocols are used by routers to inform one another of the IP networks accessible to them.
- There are classful routing protocols, such as RIPv1, that do not transmit masks in their routing updates - the classful network mask is implied.
- There are also classless routing protocols, such as OSPF, that do transmit masks in their routing updates.
- Routing protocols typically do not apply to hosts. Hosts obtain their routes by manual configuration or by DHCP.

Static vs. dynamic routing

The main difference between static routes and dynamic routes is that first one is entered in manually and the other is learned and/or calculated dynamically.

- The big differentiator, however, is in how the routers adapt to sporadic changes in network topology caused by outages.
- A statically routed network has almost no way of adapting to temporary topology changes. But routing protocols are designed for this purpose.
- A key factor in designing networks and choosing a routing protocol is convergence time, which is the time it takes for the network as a whole to discover its topology and reach a steady state.
- In general, the shorter the convergences time the better. A network that converges quickly can better compensate for unexpected outages.

Routing Protocol Category:

- Exterior Routing Protocols – used for use between different organizations such as ISPs or ISPs and their customers.
  - Ex: Border Gateway Protocol (BGP)
- Interior Routing Protocols – used to distribute routing information inside a single organization.
  - Ex: RIP, IGRP, EIGRP, OSPF, IS-IS

Interior vs. Exterior Gateway Protocols

- IGPs are used to exchange routing information with routers in the same autonomous system (AS).

- An AS is a collection of networks under a common administrative domain, which basically means that all routers sharing the same routing table information are in the same AS.
- EGPs are used to communicate between Ass such as in WAN links.

Three Classes of Routing Protocols:

- Distance Vector – finds the best path to a remote network using **hop count**. (RIP, IGRP)
- Link State – (also called shortest-path-first protocols) – the routers each create three separate tables. 1) keeps track of directly attached neighbors, 2) topology of network, 3) the routing table. (OSPF, IS-IS)
- Hybrid – uses aspects of both distance vector and link state. (EIGRP)

Approaches to Routing – Distance-vector

- Each node (router or host) exchange information with neighboring nodes
  - Neighbors are both directly connected to same network
- Node maintains vector of link costs for each directly attached network and distance and next-hop vectors for each destination
- Used by Routing Information Protocol (RIP)
- Requires transmission of lots of information by each router
  - Distance vector to all neighbors
  - Contains estimated path cost to all networks in configuration
  - Changes take long time to propagate
  - In practice, Bellman Ford algorithm

Approaches to Routing – Link-state

- Designed to overcome drawbacks of distance-vector
- When router initialized, it determines link cost on each interface
- Advertises set of link costs to all other routers in topology
  - Not just neighboring routers
- From then on, monitor link costs
  - If significant change, router advertises new set of link costs
- Each router can construct topology of entire configuration
  - Can calculate shortest path to each destination network
- Router constructs routing table, listing first hop to each destination
- Router does not use distributed routing algorithm
  - Use any routing algorithm to determine shortest paths
  - In practice, Dijkstra's algorithm
- Open shortest path first (OSPF) protocol uses link-state routing.

Exterior Router Protocols –Path-vector

- Dispense with routing metrics
- Provide information about which networks can be reached by a given router and ASs crossed to get there
  - Does not include distance or cost estimate
- Each block of information lists all ASs visited on this route
  - Enables router to perform policy routing
  - E.g. avoid path to avoid transiting particular AS
  - E.g. link speed, capacity, tendency to become congested, and overall quality of operation, security
  - E.g. minimizing number of transit Ass

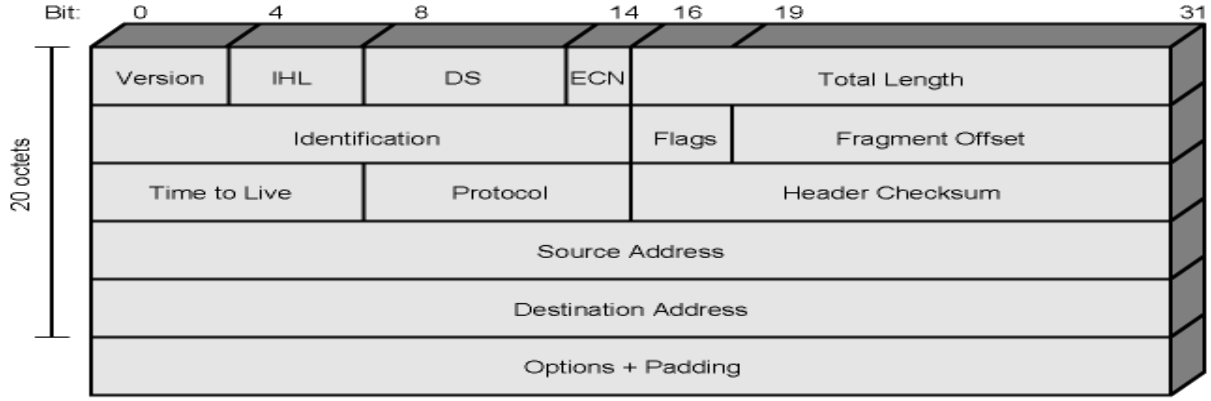
NETWORK LAYER PROTOCOLS

Internet Protocol (IP)

- IP is a connectionless, unreliable, best-effort delivery protocol.
- IP accepts whatever data is passed down to it from the upper layers and forwards the data in the form of IP Packets.

- ❖ All the nodes are identified using an IP address.
- ❖ Packets are delivered from the source to the destination using IP address

IPV4 Header:



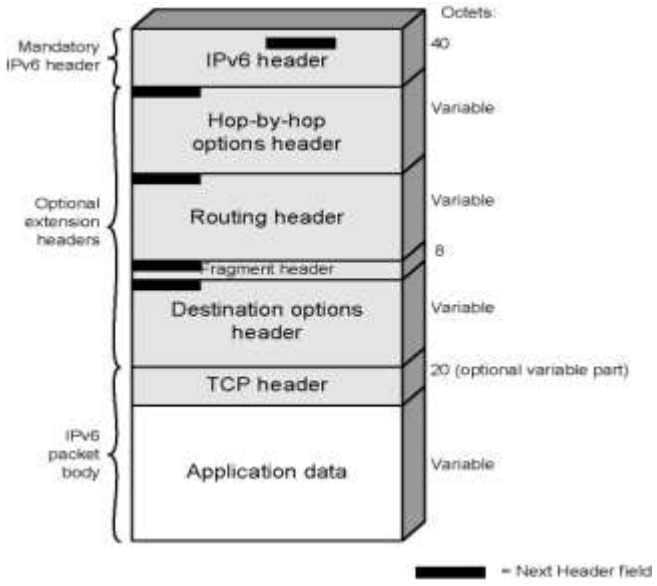
- Version
  - — Currently 4
  - — IP v6 - see later
- Internet header length
  - — In 32 bit words
  - — Including options
  - — minimum 5
- DS (Differentiated Services) and ECN (Explicit Congestion Notification)
  - — previously used for “Type of Service”
  - — now used by (interpreted as) DS and ECN
  - — DS is for QoS support (that we will not cover)
  - — we will see the concept of Explicit Congestion Notification later
- Total length
  - — of datagram (header + data), in octets
- Identification
  - — Sequence number
  - — Used with addresses and user protocol to identify datagram uniquely
- Flags
  - — More bit
  - — Don’t fragment
- Fragmentation offset
- Time to live
- Protocol
  - — Next higher layer to receive data field at destination
- Header checksum
  - — Verified and recomputed at each router
- Source address
- Destination address
- Options
- Padding
  - — To fill to multiple of 32 bits long

Advantages of IPV6 over IPV4

- Expanded address space
  - — 128 bit
  - — $6*10^{23}$ addresses per square meter on earth!
- Improved option mechanism
  - — Separate optional headers between IPv6 header and transport layer PDU
  - — Some are not examined by intermediate routers

        - Improved speed and simplified router processing
    - Easier to extend with new options
        - Flexible protocol
- Support for resource allocation
    - Labeling of packets for particular traffic flow
    - Allows special handling
        - e.g. real time video

IPv6 Packet with Extension Headers



- Hop-by-Hop Options
    - special options that require hop-by-hop processing
- Routing
    - Similar to source routing
- Fragment
    - fragmentation and reassembly information
- Authentication
    - Integrity and Authentication
- Encapsulating security payload
    - Privacy and Confidentiality (plus optional authentication)
- Destination options
    - Optional info to be processed at destination node

IPV6 Header Field:

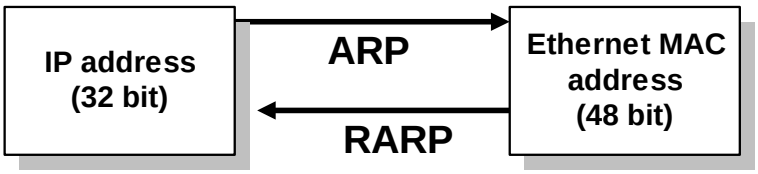
- Version
    - 6
- DS/ECN
    - Previously, Traffic Class (Types of Service)
        - Classes or priorities of packet
    - Now interpretation is different as discussed in v4
- Flow Label
    - Identifies a sequence of packets (a flow) that has special handling requirements
- Payload length
    - Length of all extension headers plus user data
- Next Header
    - Identifies type of header

- Extension or next layer up
- Hop Limit
  - Remaining number of hops
  - As in TTL of IPv4, decremented by one at each router
  - Packet discarded if reaches zero
- Source Address
- Destination address
- Longer header but less number of fields
  - simplified processing

Address Resolution Protocol(ARP)

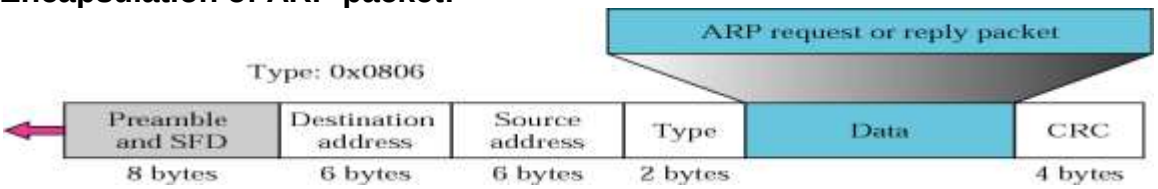
- The Internet is based on IP addresses
- Data link protocols (Ethernet, FDDI, ATM) may have different (MAC) addresses

- The ARP and RARP protocols perform the translation between IP addresses and MAC layer addresses



An ARP request is broadcast where as an ARP reply is unicast.

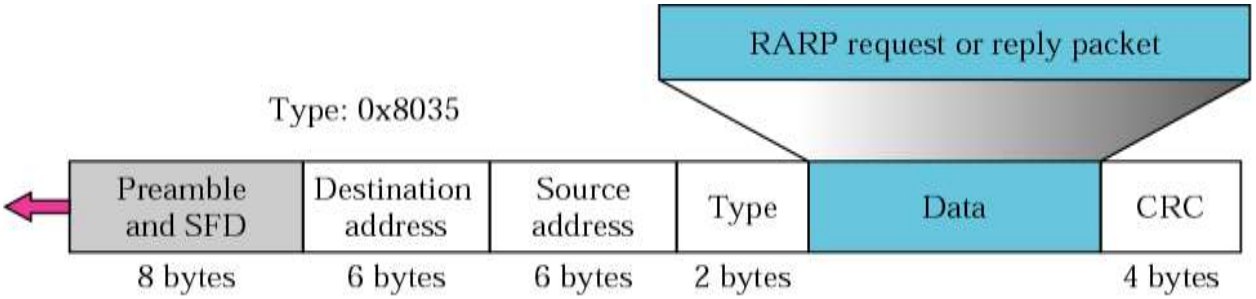
Encapsulation of ARP packet:



RARP

RARP finds the logical address for a machine that only knows its physical address.
The RARP request packets are broadcast; the RARP reply packets are unicast.

Encapsulation of RARP packet:



Internet Control Message Protocol

- used to report information and errors
- IP uses ICMP when it sends an error message
- ICMP uses IP to transport messages

ICMP Error Messages

- Source Quench
  - used by router when it has discarded datagram due to unavailable buffer memory.

- sent to source computer requesting it to slow down rate of data transmission.

- Time Exceeded
  - whenever the TIME TO LIVE field in a datagram is decremented to 0, the router discards the datagram and sends a time exceeded message
  - eg. Traceroute sends datagrams to a destination host with TIME TO LIVE field set to 1,2,3… so that routers along the path will send time exceeded message back, allowing source to know of path or routes taken by datagram.
- Destination Unreachable
  - sent by router to source host whenever router determines that a datagram cannot be delivered to its final destination. A "host unreachable" or "network unreachable" message is sent.
- Redirect message
  - sent by router to source host whenever router determines that a host has incorrectly sent a datagram that should be sent to a different router.
- Fragmentation required message
  - sent by a router to a source host if the router needs to fragment a datagram but the DO NOT FRAGMENT bit had been set in the datagram header. Such a datagram is discarded by the router.

ICMP Informational Messages

- Echo Request message
  - sent to a computer by setting the *PROTOCOL TYPE* field in a IP header to 1, corresponding to *ICMP*
  - Used by **Ping** to test for reachability.
- Echo Reply message
  - Sent by ICMP software on destination computer in response to echo request.
- address mask request & reply message
  - can be broadcasted by host during bootup, asking for router to send an address mask reply that contains the correct 32-bit subnet mask for the network.

Multicast Routing

- Unicast: one source to one destination
- Multicast: one source to many destinations

Internet Group Management Protocol

- The Internet Group Management Protocol (IGMP) is a simple protocol for the support of IP multicast.
- IGMP is defined in RFC 1112.
- IGMP operates on a physical network (e.g., single Ethernet Segment.
- IGMP is used by multicast routers to keep track of membership in a multicast group.
- Support for:
  - Joining a multicast group
  - Query membership
  - Send membership reports
- A host sends an IGMP report when it joins a multicast group (Note: multiple processes on a host can join. A report is sent only for the first process).
- No report is sent when a process leaves a group
- A multicast router regularly multicasts an IGMP query to all hosts (group address is set to zero).

TRANSPORT LAYER

- The transport layer is an end-to-end layer – this means that nodes within the subnet do not participate in transport layer protocols – only the end hosts.
- As with other layers, transport layer protocols send data as a sequence of packets (segments).
- The network layer provides communication between two hosts.

- The transport layer provides communication between two *processes* running on different hosts.
- A process is an instance of a program that is running on a host.
- There may be multiple processes communicating between two hosts – for example, there could be a FTP session and a Telnet session between the same two hosts.

Transport services and protocols

- provide *logical communication* between app processes running on different hosts
- transport protocols run in end systems
  - send side: breaks app messages into segments, passes to network layer
  - rcv side: reassembles segments into messages, passes to app layer
- more than one transport protocol available to apps
  - Internet: TCP and UDP

Transport Layer Protocols:

- reliable, in-order delivery (TCP)
  - congestion control
  - flow control
  - connection setup
- unreliable, unordered delivery: UDP
  - no-frills extension of "best-effort" IP
- services not available:
  - delay guarantees
  - bandwidth guarantees

UDP: User Datagram Protocol

- "no frills," "bare bones" Internet transport protocol
- "best effort" service, UDP segments may be:
  - lost
  - delivered out of order to app
- *connectionless:*
  - no handshaking between UDP sender, receiver
  - each UDP segment handled independently of others
- often used for streaming multimedia apps
  - loss tolerant
  - rate sensitive
- other UDP uses
  - DNS
  - SNMP
- reliable transfer over UDP: add reliability at application layer
  - application-specific error recovery!

TCP Service Model

- TCP Service is obtained by having both the sender and receiver create end points, called sockets. Each socket has a socket number (address) consisting of the IP address of the host and a 16-bit number local to that host, called a port.
- To obtain TCP service, a connection must be explicitly established between a socket on the sending machine and a socket on the receiving machine.
- A socket may be used for multiple connections at the same time. In other words, two or more connections may terminate at the same socket.
- Port numbers below 1024 are called well-known ports and are reserved for standard services. For example, any process wishing to establish a connection to a host to transfer a file using FTP can connect to the destination host's port 21 to contact its FTP

daemon/service. Similarly, to establish a remote login session using TELNET, port 23 is used. Port 80 is used for HTTP, port 443 is used for SSL, etc.

- Ports between 1024 and 5000 are called ephemeral and are free to use (not reserved). The client's socket would use such port.
- All TCP connections are full-duplex and point-to-point. Full duplex means that traffic can go in both directions at the same time. Point-to-point means that each connection has exactly two end points. TCP does not support multicasting or broadcasting.
- A TCP connection is byte stream, not a message stream. Message boundaries are not preserved end to end.
- For example, if the sending process does four 512-byte writes to a TCP stream, these data may be delivered to the receiving process as four 512-byte chunks, or two 1024-byte chunks, or one 2048-byte chunk, or some other way.
- When an application passes data to TCP, TCP may send it immediately or buffer it (in order to collect a larger amount to send at once), at its discretion.
- Every byte on a TCP connection has its own 32-bit sequence number.
- The sending and receiving TCP entities exchange data in the form of segments. A segment consists of a fixed 20-byte header (plus an optional part) followed by 0 or more data bytes. The TCP software decides how big segments should be. It can accumulate data from several writes into one segment or split data from one write over multiple segments.
- Two limits restrict the segment size:
- Each segment, including the TCP header, must fit in the 64K byte IP payload
- Each network has a maximum transfer unit or MTU, and each segment must fit in the MTU.
- TCP uses a sliding window mechanism for flow control
- Sender maintains 3 pointers for each connection
- Pointer to bytes sent and acknowledged
- Pointer to bytes sent, but not yet acknowledged
- *Sender window includes bytes sent but not acknowledged*
- Pointer to bytes that cannot yet be sent
-

Port Numbers:

- Port numbers are 16-bit integers (0 → 65,535)
  - Servers use *well know ports*, 0-1023 are privileged
  - Clients use *ephemeral* (short-lived) ports
- *Internet Assigned Numbers Authority* (IANA) maintains a list of port number assignment
  - Well-known ports (0-1023) → controlled and assigned by IANA
  - Registered ports (1024-49151) → IANA registers and lists use of ports as a convenience (49151 is ¾ of 65536)
  - Dynamic ports (49152-65535) → ephemeral ports

Socket Addressing

- Process-to-process delivery needs *two* identifiers
  - IP address and Port number
  - Combination of IP address and port number is called a socket address (a socket is a communication endpoint)
  - Client socket address uniquely identifies client process
  - Server socket address uniquely identifies server process
- Transport-layer protocol needs a *pair* of socket addresses
  - Client socket address
  - Server socket address
  - For example, socket pair for a TCP connection is a 4-tuple
    - ✓ Local IP address, local port, and
    - ✓ foreign IP address, foreign port

Multiplexing and Demultiplexing:

Multiplexing

Sender side may have several processes that need to send packets (albeit only 1 transport-layer protocol)

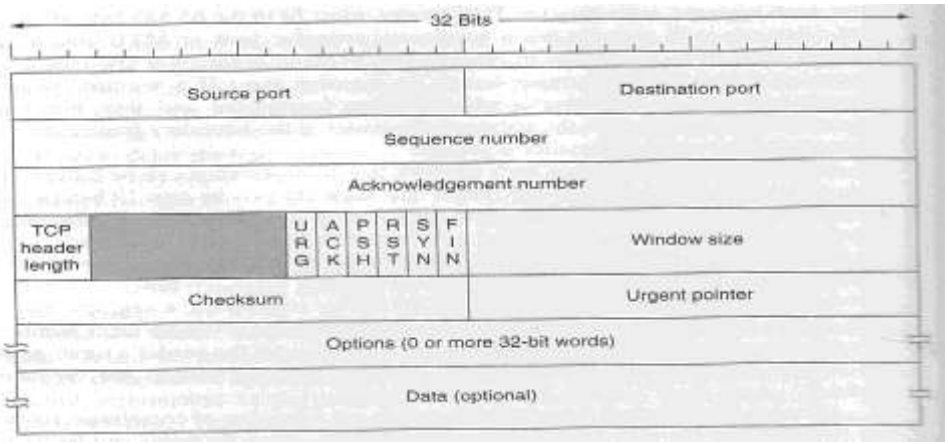
Demultiplexing

At receiver side, after error checking and header dropping, transport-layer delivers each message to appropriate process

- **Flow Control**
  - Tell peer exactly how many bytes it is willing to accept (advertised window → sender can not overflow receiver buffer)
    - ✓ *Sender window includes bytes sent but not acknowledged*
    - ✓ *Receiver window (number of empty locations in receiver buffer)*
    - ✓ Receiver advertises window size in ACKs
  - Sender window <= receiver window (flow control)
    - ✓ Sliding sender window (without a change in receiver's advertised window)
    - ✓ Expanding sender window (receiving process consumes data faster than it receives → receiver window size increases)
    - ✓ Shrinking sender window (receiving process consumes data more slowly than it receives → receiver window size reduces)
    - ✓ Closing sender window (receiver advertises a window of zero)
- **Error Control**
  - Mechanisms for detecting corrupted segments, lost segments, out-of-order segments, and duplicated segments
  - Tools: checksum (corruption), ACK, and time-out (one time-out counter per segment)
    - ✓ *Lost segment or corrupted segment* are the same situation: segment will be retransmitted after time-out (no NACK in TCP)
    - ✓ *Duplicate segment* (destination discards)
    - ✓ *Out-of-order segment* (destination does not acknowledge, until it receives all segments that precede it)
    - ✓ *Lost ACK* (loss of an ACK is irrelevant, since ACK mechanism is cumulative)

- **Congestion Control**
  - TCP assumes the cause of a lost segment is due to congestion in the network
  - If the cause of the lost segment is congestion, retransmission of the segment does not remove the problem, it actually aggravates it
  - The network needs to tell the sender to slow down (affects the sender window size in TCP)
  - Actual window size = Min (receiver window size, congestion window size)
    - ✓ The congestion window is flow control imposed by the sender
    - ✓ The advertised window is flow control imposed by the receiver

TCP segment Header:



- Source port and Destination port – identify the local end points of the connection.
- Sequence number and acknowledgement number (specifies the next sequence number expected)
- TCP header length – tells now many 32-bit words are contained in the TCP header (because Options field is of variable length)
- Next comes a 6-bit field that is not used.
- Next come 6 1-bit flags:
  - URG is set to 1 if the Urgent pointer is in use. The Urgent Pointer is used to indicate a byte offset (from the current sequence number) at which urgent data is located
  - ACK is set to 1 to indicate that the acknowledgement number field is valid. Otherwise, if set to 0, then this segment does not contain an acknowledgment
  - PSH bit indicates PUSHed data. The receiver hereby kindly requested to deliver the data to the application upon arrival and not buffer it (done for efficiency)
  - RST bit is used to reset a connection that has become confused due to a host crash or some other reason. It is also used to reject an invalid segment or refuse an attempt to open a connection.
  - SYN bit is used to establish connections. SYN=1 and ACK=0 – connection request, SYN=1 and ACK=1 – connection accepted.
  - FIN but is used to release a connection. It specifies that the sender has no more data to transmit.
- Window size field tells how many bytes may be sent starting at the byte acknowledged.
- A Checksum is also provided for extreme reliability – it checksums the header and the data.
- Options field was designed to provide a way to add extra facilities not covered by the regular header. For example, allow each host to specify the maximum TCP payload it is willing to accept. (using large segments is more efficient than using small ones)

TCP Connection Establishment

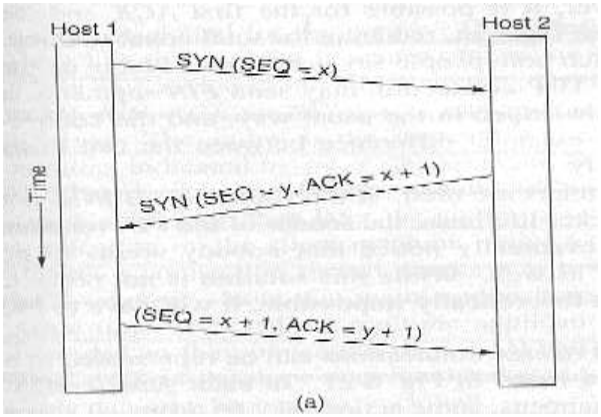
- TCP uses a **three-way handshake** to open a connection:
  (1) ACTIVE OPEN: Client sends a segment with
    - SYN bit set *
    - port number of client
    - initial sequence number (ISN) of client

  (2) PASSIVE OPEN: Server responds with a segment with
    - SYN bit set *
    - initial sequence number of server

- ACK for ISN of client

(3) Client acknowledges by sending a segment with:

- ACK ISN of server (* counts as one byte)



TCP connection Establishment

SYN: Synchronize
ACK: Acknowledge

TCP Connection Termination

- Each end of the data flow must be shut down independently **("half-close")**
- If one end is done it sends a FIN segment. This means that no more data will be sent
- Four steps involved:
  (1) X sends a FIN to Y **(active close)**
  (2) Y ACKs the FIN,
  (at this time: Y can still send data to X)
  (3) and Y sends a FIN to X **(passive close)**
  (4) X ACKs the FIN.

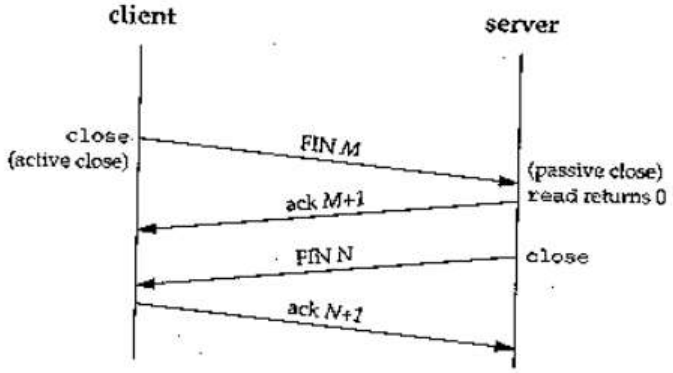
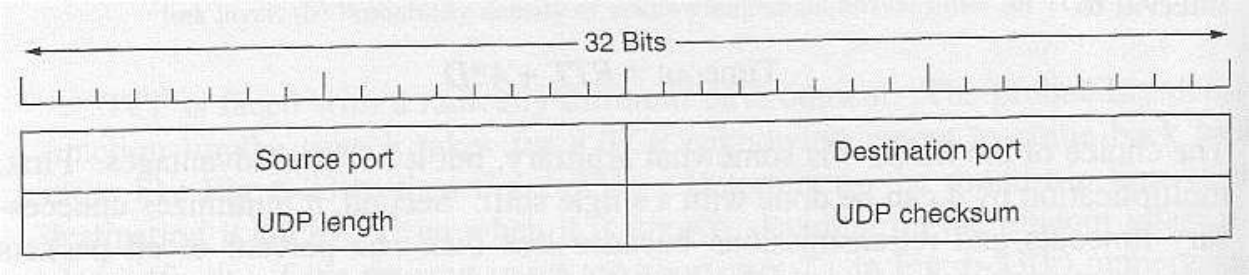


Figure 2.3 Packets exchanged when a TCP connection is closed.

- FIN: Finish
- Step 1 can be sent with data
- Steps 2 and 3 can be combined into 1 segment

UDP

- The Internet protocol suite also supports a connectionless transport protocol, UDP (User Data Protocol)
- UDP provides a way for applications to send encapsulated raw IP datagrams and send them without having to establish a connection.
- Many client-server applications that have 1 request and 1 response use UDP rather than go to the trouble of establishing and later releasing a connection.
- A UDP segment consists of an 8-byte header followed by the data.



UDP Header

- The two ports serve the same function as they do in TCP: to identify the end points within the source and destination machines.
- The UDP length field includes the 8-byte header and the data.
- The UDP checksum is used to verify the size of header and data.

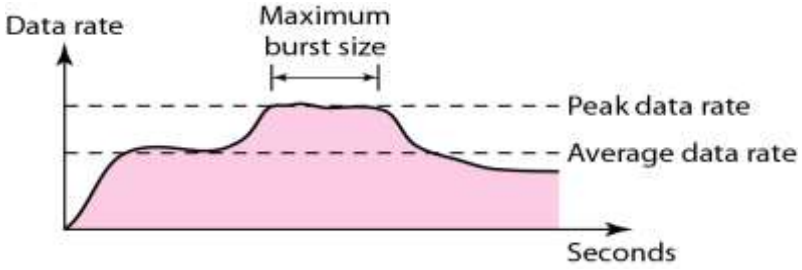
Congestion Control and Quality of service

DATA TRAFFIC

The main focus of congestion control and quality of service is data traffic. In congestion control we try to avoid traffic congestion. In quality of service, we try to create an appropriate environment for the traffic.

Traffic Descriptor

Traffic descriptors are qualitative values that represent a data flow. Figure shows a traffic flow with some of these values.



Average Data Rate

The average data rate is the number of bits sent during a period of time, divided by the number of seconds in that period. We use the following equation:

Average data rate =amount of data/time

Peak Data Rate

The peak data rate defines the maximum data rate of the traffic. The peak data rate is a very important measurement because it indicates the peak bandwidth that the network needs for traffic to pass through without changing its data flow.

Maximum Burst Size

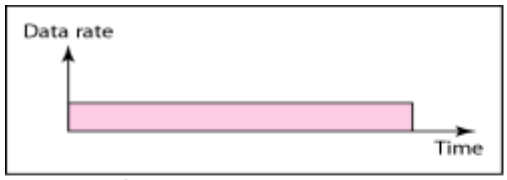
Although the peak data rate is a critical value for the network, it can usually be ignored if the duration of the peak value is very short.

Effective Bandwidth

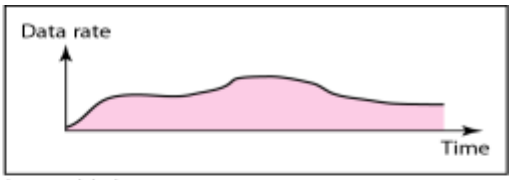
The effective bandwidth is the bandwidth that the network needs to allocate for the flow of traffic. The effective bandwidth is a function of three values: average data rate, peak data rate, and maximum burst size.

Traffic Profiles

For our purposes, a data flow can have one of the following traffic profiles: constant bit rate, variable bit rate, or bursty as shown in Figure below.



a. Constant bit rate



b. Variable bit rate

c. Bursty

Constant Bit Rate

A constant-bit-rate (CBR), or a fixed-rate, traffic model has a data rate that does not change. In this type of flow, the average data ratc and thc peak data rate are the same.

Variable Bit Rate

In the variable-bit-rate (VBR) category, the rate of the data flow changes in time, with the changes smooth instead of sudden and sharp.

Bursty

In the **bursty data** category, the data rate changes suddenly in a very short time. It may jump from zero, for example, to 1 Mbps in a few microseconds and vice versa. Bursty traffic is one of the main causes of congestion in a network.

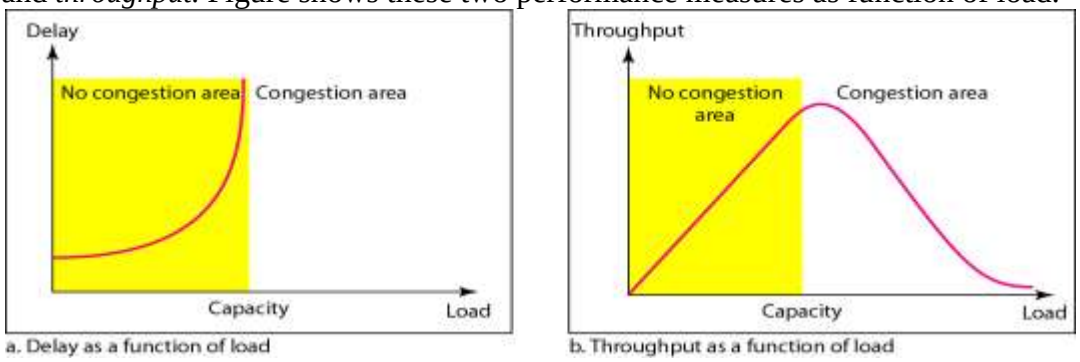
CONGESTION

Congestion in a network may occur if the **load** on the network-the number of packets sent to the network-is greater than the *capacity* of the network-the number of packets a network can handle. **Congestion control** refers to the mechanisms and techniques to control the congestion and keep the load below the capacity.

Congestion in a network or internetwork occurs because routers and switches have queues-buffers that hold the packets before and after processing.

Network Performance

Congestion control involves two factors that measure the performance of a network: *delay* and *throughput.* Figure shows these two performance measures as function of load.



a. Delay as a function of load

b. Throughput as a function of load

Delay versus Load

When the load is much less than the capacity of the network, the delay is at a minimum. This minimum delay is composed of propagation delay and processing delay, both of which are negligible. However, when the load reaches the network capacity, the delay increases sharply.

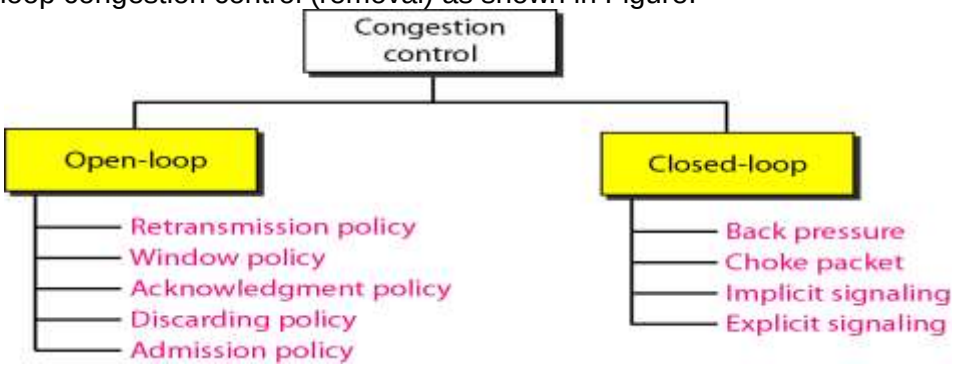
Throughput versus Load

We can define throughput in a network as the number of packets passing through the network in a unit of time. From the above figure it can be found that when the load is below the capacity of the network, the throughput increases proportionally with the *load.* We expect the throughput to remain constant after the load reaches the capacity, but instead the throughput declines sharply.

CONGESTION CONTROL

Congestion control refers to techniques and mechanisms that can either prevent congestion,

before it happens, or remove congestion, after it has happened. we can divide congestion control mechanisms into two broad categories: open-loop congestion control (prevention) and closed-loop congestion control (removal) as shown in Figure.



Open-Loop Congestion Control

In open-loop congestion control, policies are applied to prevent congestion before it happens.

Retransmission Policy

Retransmission is sometimes unavoidable. If the sender feels that a sent packet is lost or corrupted, the packet needs to be retransmitted. Retransmission in general may increase congestion in the network. However, a good retransmission policy can prevent congestion.

Window Policy

The type of window at the sender may also affect congestion. The Selective Repeat window is better than the Go-Back-N window for congestion control. The Selective Repeat window tries to send the specific packets that have been lost or corrupted instead of sending several packets.

Acknowledgment Policy

The acknowledgment policy imposed by the receiver may also affect congestion. If the receiver does not acknowledge every packet it receives, it may slow down the sender and help prevent congestion. Sending fewer acknowledgments means imposing less load on the network.

Discarding Policy

A good discarding policy by the routers may prevent congestion and at the same time may not harm the integrity of the transmission.

Admission Policy

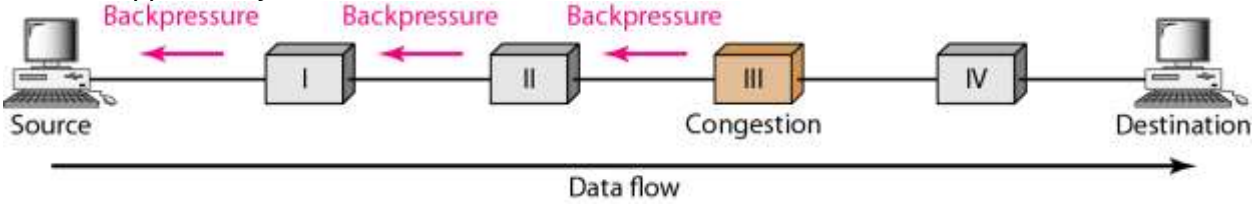
An admission policy, which is a quality-of-service mechanism, can also prevent congestion in virtual-circuit networks.

Closed-Loop Congestion Control

Closed-loop congestion control mechanisms try to alleviate congestion after it happens. Several policies are as follows:

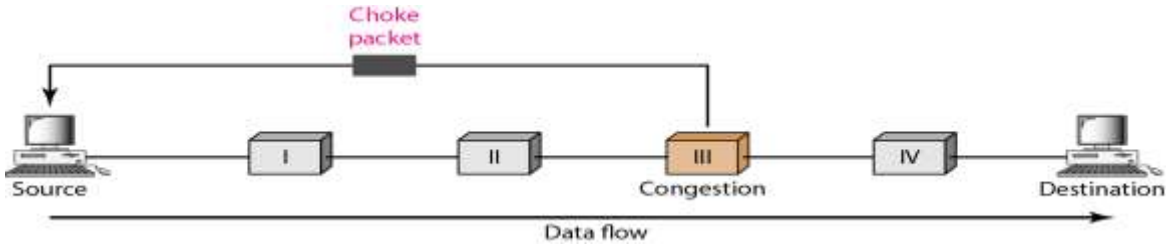
Backpressure

The technique of *backpressure* refers to a congestion control mechanism in which a congested node stops receiving data from the immediate upstream node or nodes. This may cause the upstream node or nodes to become congested, and they, in turn, reject data from their upstream nodes or nodes. Backpressure is a node-to-node congestion control that starts with a node and propagates, in the opposite direction of data flow, to the source. The backpressure technique can be applied only to virtual circuit networks



Choke Packet

A choke packet is a packet sent by a node to the source to inform it of congestion. In backpressure, the warning is from one node to its upstream node, although the warning may eventually reach the source station. In the choke packet method, the warning is from the router, which has encountered congestion, to the source station directly. The intermediate nodes through which the packet has travelled are not warned.



Implicit Signaling

In implicit signaling, there is no communication between the congested node or nodes and the source. The source guesses that there is a congestion somewhere in the network from other symptoms.

Explicit Signaling

The node that experiences congestion can explicitly send a signal to the source or destination. The explicit signaling method, however, is different from the choke packet method. In the choke packet method, a separate packet is used for this purpose; in the explicit signaling method, the signal is included in the packets that carry data. Explicit signalling can occur in either the forward or the backward direction.

Backward Signaling A bit can be set in a packet moving in the direction opposite to the congestion. This bit can warn the source that there is congestion and that it needs to slow down to avoid the discarding of packets.

Forward Signaling A bit can be set in a packet moving in the direction of the congestion. This bit can warn the destination that there is congestion. The receiver in this case can use policies, such as slowing down the acknowledgments, to alleviate the congestion.

Congestion Control in TCP

- TCP's general policy for handling congestion is based on three phases: slow start, congestion avoidance, and congestion detection.
- In the slow-start phase, the sender starts with a very slow rate of transmission, but increases the rate rapidly to reach a threshold.
- When the threshold is reached, the data rate is reduced to avoid congestion.
- Finally if congestion is detected, the sender goes back to the slow-start or congestion avoidance phase based on how the congestion is detected.

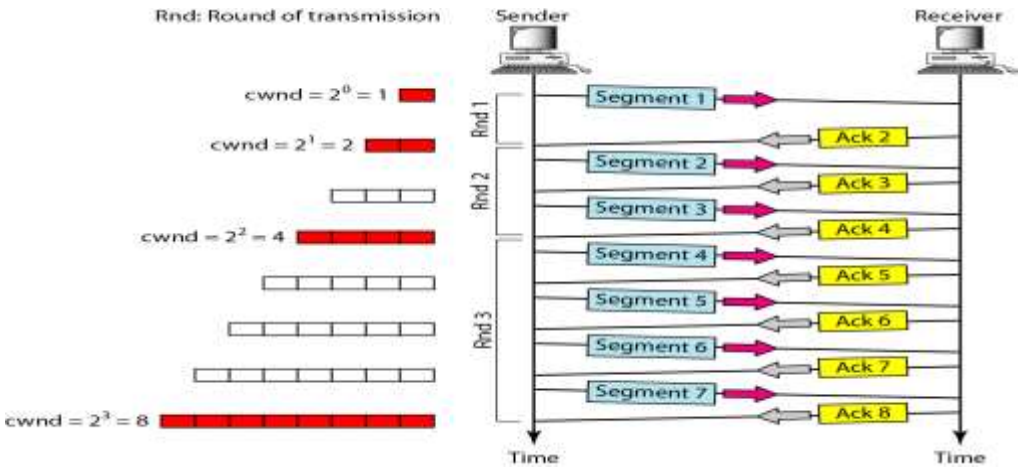
Slow Start: Exponential Increase

- The source starts with cwnd = 1.
- Every time an ACK arrives, cwnd is incremented.

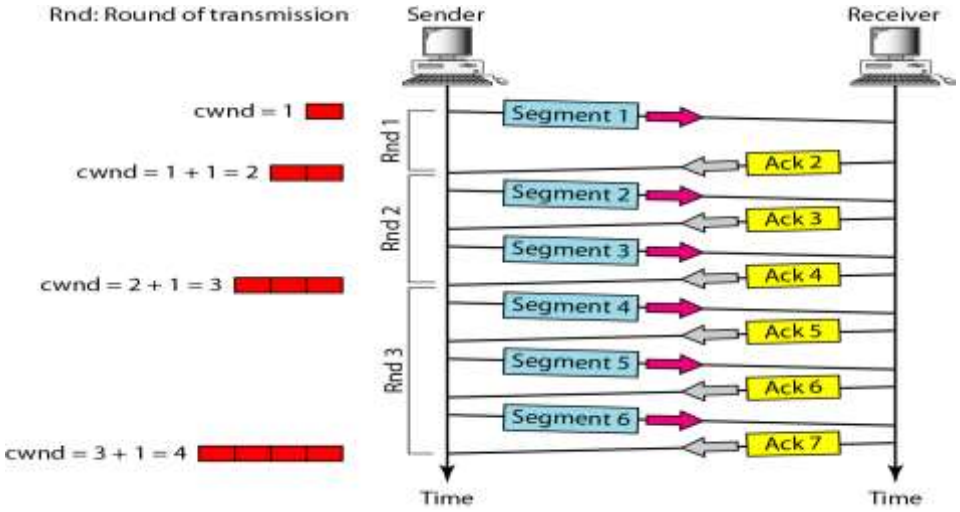
➔ cwnd is effectively doubled per RTT “epoch”.

- Two slow start situations:
  - At the very beginning of a connection **{cold start}**.
  - When the connection goes dead waiting for a timeout to occur (i.e, the advertized window goes to zero!)
  - However, in the second case the source has more information. The current value of cwnd can be saved as a **congestion threshold.**
  - This is also known as the “slow start threshold” **ssthresh**.

- In the slow-start algorithm, the size of the congestion window increases exponentially until it reaches a threshold.



Congestion Avoidance: Additive Increase If we start with the slow-start algorithm, the size of the congestion window increases exponentially. To avoid congestion before it happens, one must slow down this exponential growth. TCP defines another algorithm called congestion avoidance, which undergoes an additive increase instead of an exponential one. When the size of the congestion window reaches the slow-start threshold, the slow-start phase stops and the additive phase begins. In this algorithm, each time the whole window of segments is acknowledged (one round), the size of the congestion window is increased by 1.



In the congestion avoidance algorithm, the size of the congestion window increases additively until congestion is detected.

Congestion Detection: Multiplicative Decrease If congestion occurs, the congestion window size must be decreased. The only way the sender can guess that congestion has occurred is by the need to retransmit a segment. However, retransmission can occur in one of two cases: when a timer times out or when three ACKs are received. In both cases, the size of the threshold is dropped to one-half, a multiplicative decrease.

An implementation reacts to congestion detection in one of the following ways:

❑ If detection is by time-out, a new slow start phase starts.

❑ If detection is by three ACKs, a new congestion avoidance phase starts.

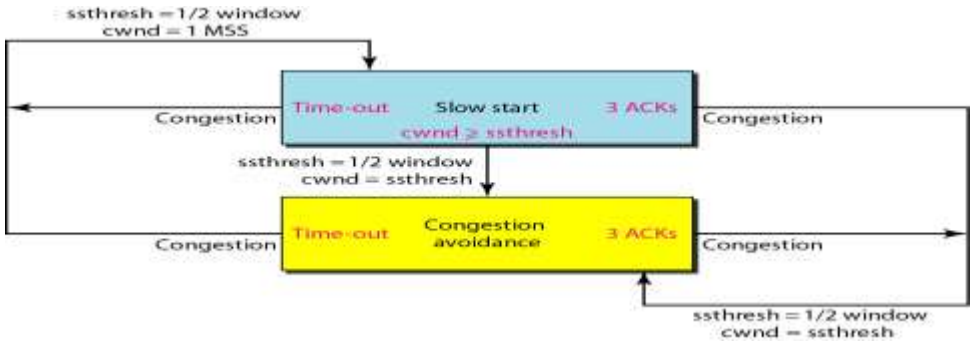


Fig.TCP congestion policy summary

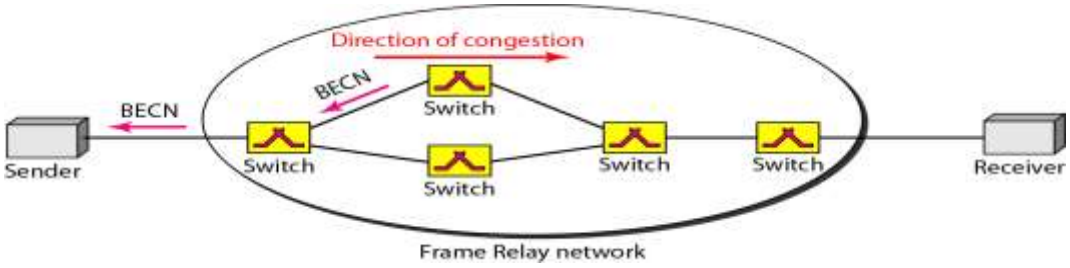
Congestion Control in Frame Relay

Congestion in a Frame Relay network decreases throughput and increases delay. A high throughput and low delay are the main goals of the Frame Relay protocol. Frame Relay does not have flow control. In addition, Frame Relay allows the user to transmit bursty data. This means that a Frame Relay network has the potential to be really congested with traffic, thus requiring congestion control.

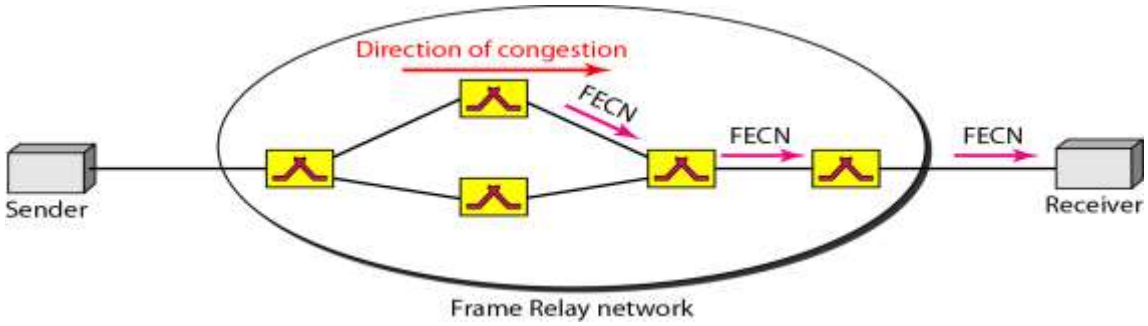
Congestion Avoidance

For congestion avoidance, the Frame Relay protocol uses 2 bits in the frame to explicitly warn the source and the destination of the presence of congestion.

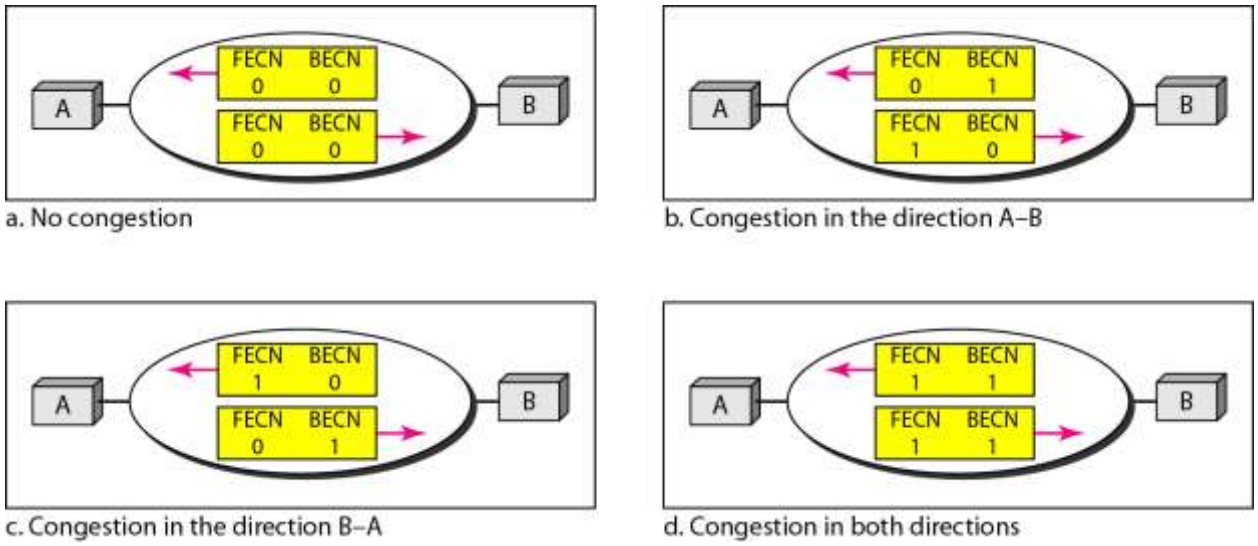
BECN The backward explicit congestion notification (BECN) bit warns the sender of congestion in the network. One might ask how this is accomplished since the frames are traveling away from the sender. In fact, there are two methods: The switch can use response frames from the receiver (full-duplex mode), or else the switch can use a predefined connection (DLCI =1023) to send special frames for this specific purpose. The sender can respond to this warning by simply reducing the data rate.



FECN The forward explicit congestion notification (FECN) bit is used to warn the receiver of congestion in the network. It might appear that the receiver cannot do anything to relieve the congestion. However, the Frame Relay protocol assumes that the sender and receiver are communicating with each other and are using some type of flow control at a higher level.



When two endpoints are communicating using a Frame Relay network, four situations may occur with regard to congestion. Figure shows these four situations and the values of FECN and BECN.



a. No congestion

b. Congestion in the direction A–B

c. Congestion in the direction B–A

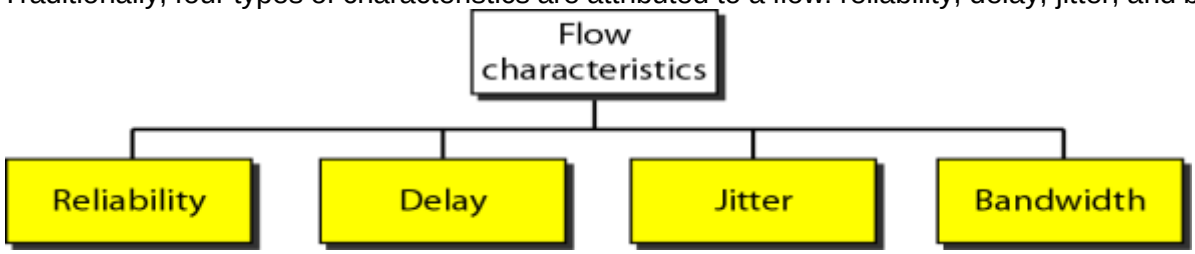
d. Congestion in both directions

QUALITY OF SERVICE

Quality of service (QoS) is an internetworking issue that has been discussed more than defined. We can informally define quality of service as something a flow seeks to attain.

Flow Characteristics

Traditionally, four types of characteristics are attributed to a flow: reliability, delay, jitter, and bandwidth.



Reliability

Reliability is a characteristic that a flow needs. Lack of reliability means losing a packet or acknowledgment, which entails retransmission.

Delay

Source-to-destination delay is another flow characteristic.

Jitter

Jitter is the variation in delay for packets belonging to the same flow.

Bandwidth

Different applications need different bandwidths.

TECHNIQUES TO IMPROVE QoS

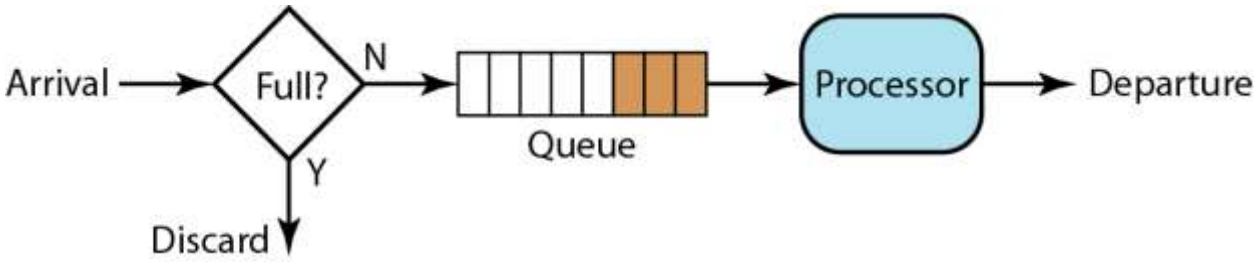
We briefly discuss four common methods: scheduling, traffic shaping, admission control, and resource reservation.

Scheduling

Packets from different flows arrive at a switch or router for processing. A good scheduling technique treats the different flows in a fair and appropriate manner. Several scheduling techniques are designed to improve the quality of service. We discuss three of them here: FIFO queuing, priority queuing, and weighted fair queuing.

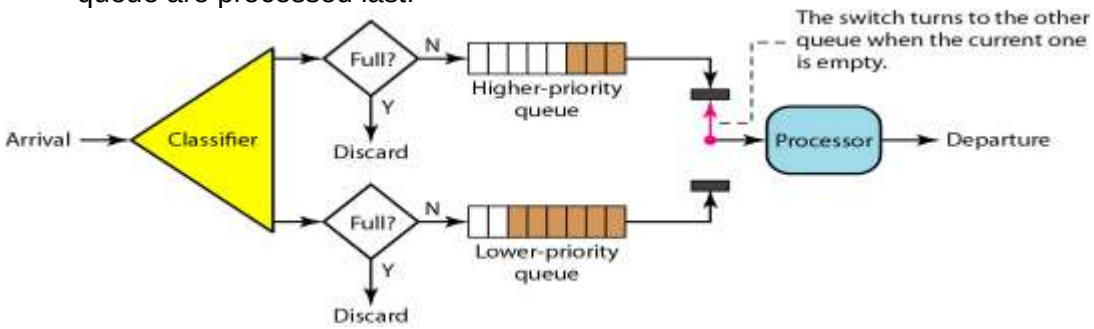
FIFO Queuing

In first-in, first-out (FIFO) queuing, packets wait in a buffer (queue) until the node (router or switch) is ready to process them. If the average arrival rate is higher than the average processing rate, the queue will fill up and new packets will be discarded.



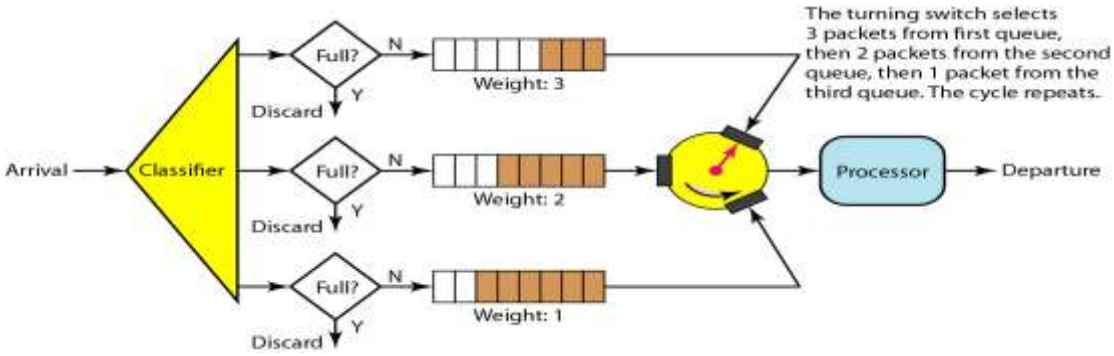
Priority Queuing

In priority queuing, packets are first assigned to a priority class. Each priority class has its own queue. The packets in the highest-priority queue are processed first. Packets in the lowest-priority queue are processed last.



Weighted Fair Queuing

A better scheduling method is weighted fair queuing. In this technique, the packets are still assigned to different classes and admitted to different queues. The queues, however, are weighted based on the priority of the queues; higher priority means a higher weight. The system processes packets in each queue in a round-robin fashion with the number of packets selected from each queue based on the corresponding weight.

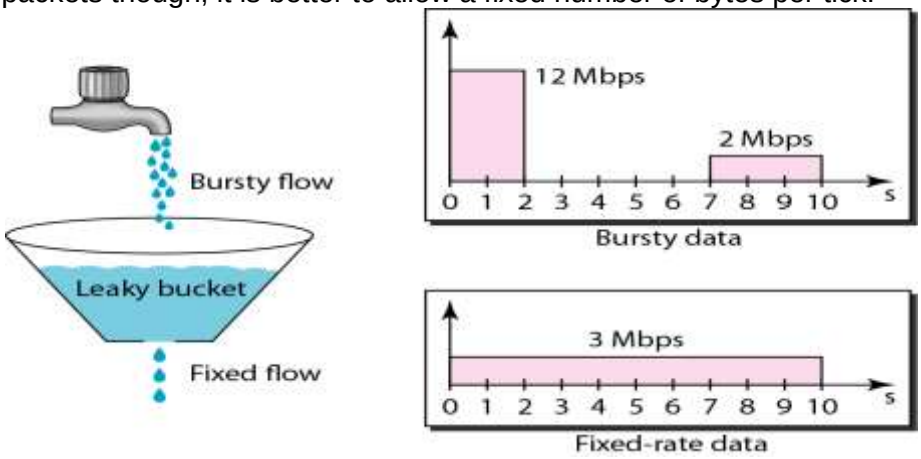


Traffic Shaping

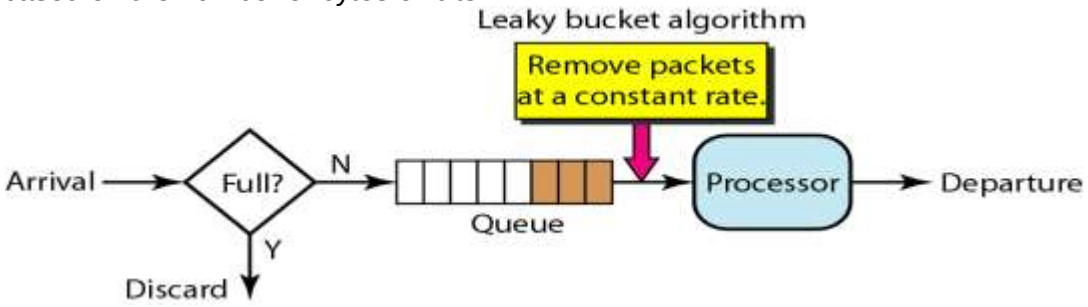
- Traffic shaping controls the *rate* at which packets are sent (not just how many)
- At connection set-up time, the sender and carrier negotiate a traffic pattern (shape)
- Two traffic shaping algorithms are:
  - Leaky Bucket
  - Token Bucket

The Leaky Bucket Algorithm

- The **Leaky Bucket Algorithm** used to control rate in a network. It is implemented as a single-server queue with constant service time. If the bucket (buffer) overflows then packets are discarded.
- The leaky bucket enforces a constant output rate regardless of the burstiness of the input. Does nothing when input is idle.
- The host injects one packet per clock tick onto the network. This results in a uniform flow of packets, smoothing out bursts and reducing congestion.
- When packets are the same size (as in ATM cells), the one packet per tick is okay. For variable length packets though, it is better to allow a fixed number of bytes per tick.



A simple leaky bucket implementation is shown in Figure below. A FIFO queue holds the packets. If the traffic consists of fixed-size packets the process removes a fixed number of packets from the queue at each tick of the clock. If the traffic consists of variable-length packets, the fixed output rate must be based on the number of bytes or bits.



Token Bucket Algorithm

- In contrast to the LB, the Token Bucket (TB) algorithm, allows the output rate to vary, depending on the size of the burst.
- In the TB algorithm, the bucket holds tokens. To transmit a packet, the host must capture and destroy one token.
- Tokens are generated by a clock at the rate of one token every Δt sec.
- Idle hosts can capture and save up tokens (up to the max. size of the bucket) in order to send larger bursts later.
- The token bucket allows bursty traffic at a regulated maximum rate.

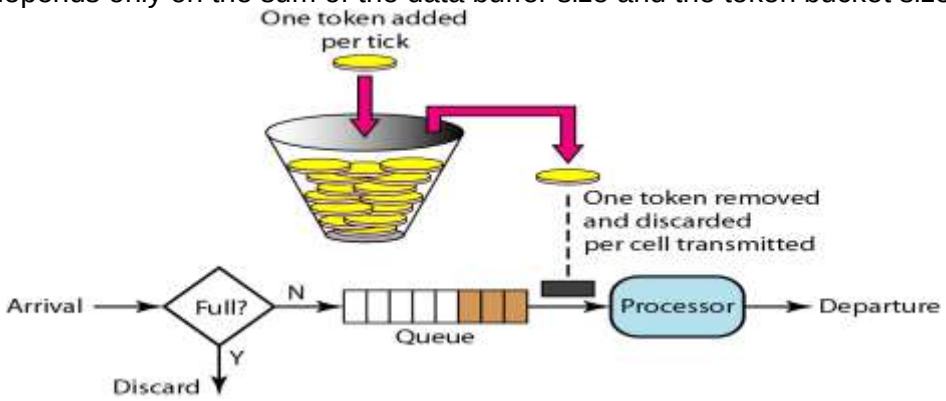
Token bucket operation

- TB accumulates fixed size tokens in a token bucket
- Transmits a packet (from data buffer, if any are there) or arriving packet if the sum of the token sizes in the bucket add up to packet size
- More tokens are periodically added to the bucket (at rate Δt). If tokens are to be added when the bucket is full, they are discarded

Token bucket properties

- Does not bound the peak rate of small bursts, because bucket may contain enough token to cover a complete burst size

- Performance depends only on the sum of the data buffer size and the token bucket size



Name servers

The DNS name space is divided up into nonoverlapping **zones**, A zone normally has one primary name server, which gets its information from a file on its disk, and one or more secondary name servers, which get their information from the primary name server

When a resolver has a query about a domain name, it passes the query to one of the local name servers.

- If the domain being sought falls under the jurisdiction of the name server, it returns the **authoritative records** (always correct).
- Once these records get back to the local name server, they will be entered into a cache there (timer controlled).

SNMP - Simple Network Management Protocol

The SNMP model

The SNMP model of a managed network consists of four components

1. Managed nodes.
2. Management stations.
3. Management information
4. A management protocol. Network management is done from **management stations**: general-purpose computers with a graphical user interface.

ASN.1 - Abstract Syntax Notation 1

The heart of the SNMP model is the set of objects managed by the agents and read and written by the management station.

To make multivendor communication possible, it is essential that these objects be defined in a standard and vendor-neutral way.

Furthermore, a standard way is needed to encode them for transfer over a network.

A standard object definition language, along with encoding rules, is needed. The one used by SNMP is taken from OSI and called **ASN.1 (Abstract Syntax Notation One)**, defined in International Standard 8824.

The rules for encoding ASN.1 data structures to a bit stream for transmission are given in International Standard 8825. The format of the bit stream is called the **transfer syntax.**

The basic idea:

- The users first define the data structure types in their applications in ASN.1 notation.
- When an application wants to transmit a data structure, it passes the data structure to the presentation layer (in the OSI model), along with the ASN.1 definition of the data structure.
- Using the ASN.1 definition as a guide, the presentation layer then knows what the types and sizes of the fields in the data structure are, and thus knows how to encode them for transmission according to the ASN.1 transfer syntax.
- Using the ASN.1 transfer syntax as a guide, the receiving presentation layer is able to do any necessary conversions from the external format used on the wire to the internal format used by the receiving computer, and pass a semantically equivalent data structure to the application layer.