Data Communication and Computer Network

CHAPTER 1

INTRODUCTION TO DATA

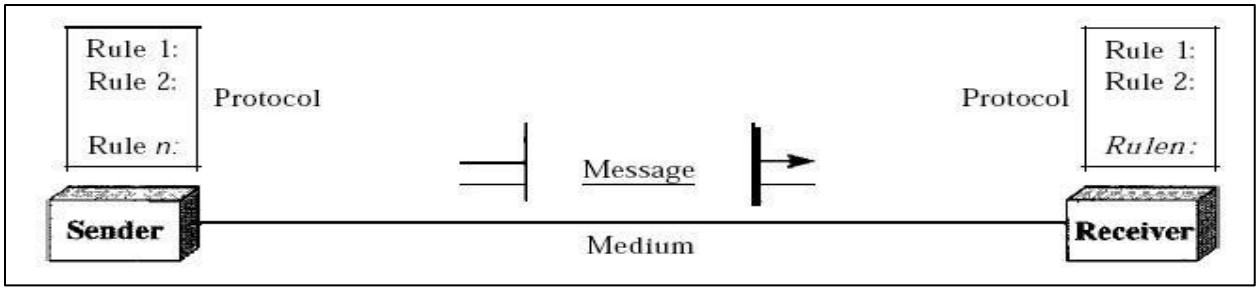
Data communications refers to the exchange of data between two devices via some form of transmission medium such as a wire cable. For data communications to occur, the communicating devices must be part of a communication system made up of a combination of hardware (physical equipment) and software (programs). The effectiveness of a data communications system depends on four fundamental characteristics: delivery, accuracy, timeliness, and jitter.

1. **Delivery:** The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.
2. **Accuracy:** The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.
3. **Timeliness:** The system must deliver data in a timely manner. Data delivered late are useless. In the case of video and audio, 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.
4. **Jitter:** Jitter refers to the variation in the packet arrival time. It is the uneven delay in the delivery of audio or video packets. For example, let us assume that video packets are sent every 30ms. If some of the packets arrive with 30ms delay and others with 40ms delay, an uneven quality in the video is the result.

Components of a data communications system

A data communications system has five components:

- **Message:** The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.
- **Sender:** The sender is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.
- **Receiver:** The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.
- **Transmission medium:** The transmission medium is the physical path by which a message travels from sender to receiver. Some examples of transmission media include twisted-pair wire, coaxial cable, fiber-optic cable, and radio waves.
- **Protocol:** A protocol is a set of rules that govern data communications. It represents an agreement between the communicating devices. Without a protocol, two devices may be connected but not communicating, just as a person speaking French cannot be understood by a person who speaks only Japanese.



Data Flow

Communication between two devices can be simplex, half-duplex, or full-duplex.

Simplex

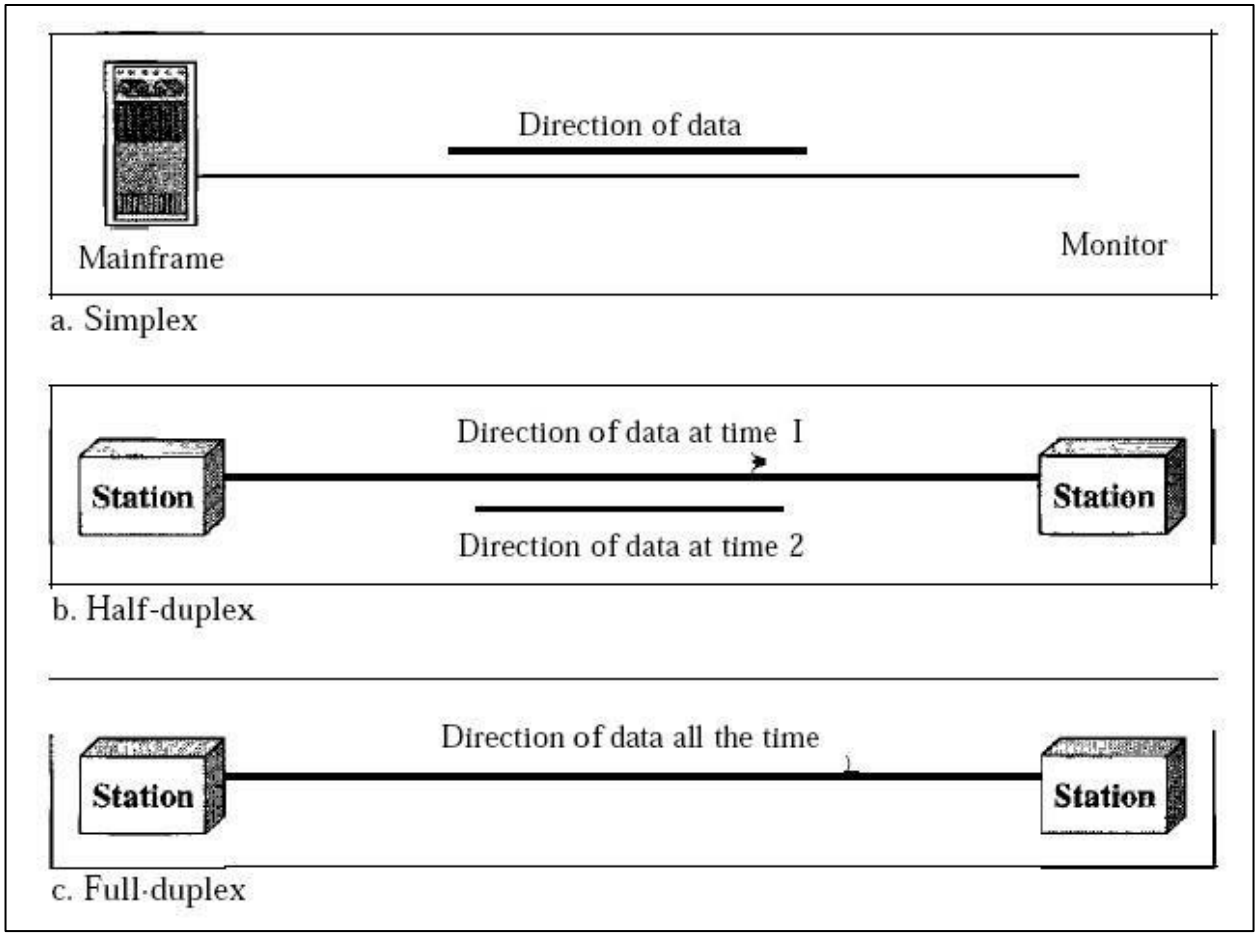
- In simplex mode, the communication is unidirectional, as on a one-way street. Only one of the two devices on a link can transmit; the other can only receive.
- Keyboards and traditional monitors are examples of simplex devices. The keyboard can only introduce input; the monitor can only accept output.
- The simplex mode can use the entire capacity of the channel to send data in one direction.

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, and vice versa.
- The half-duplex mode is like a one-lane road with traffic allowed in both directions. When cars are traveling in one direction, cars going the other way must wait.
- In a half-duplex transmission, the entire capacity of a channel is taken over by whichever of the two devices is transmitting at the time.

Full Duplex

- In full-duplex mode, both stations can transmit and receive simultaneously.
- The full-duplex mode is like a two way street with traffic flowing in both directions at the same time.
- One common example of full-duplex communication is the telephone network. When two people are communicating by a telephone line, both can talk and listen at the same time.



Network

A network is a set of devices (often referred to as nodes) connected by communication links. A node can be a computer, printer, or any other device capable of sending and/or receiving data generated by other nodes on the network.

Distributed Processing

Most networks use distributed processing, in which a task is divided among multiple computers. Instead of one single large machine being responsible for all aspects of a process, separate computer (usually a personal computer or workstation) handle a subset.

Network Criteria

A network must be able to meet a certain number of criteria. The most important of these are performance, reliability, and security.

Performance

- Performance can be measured in many ways, including transmit time and response time.
- Transmit time is the amount of time required for a message to travel from one device to another. Response time is the elapsed time between an inquiry and a response.

- The performance of a network depends on a number of factors, including the number of users, the type of transmission medium, the capabilities of the connected hardware, and the efficiency of the software.
- Performance is often evaluated by two networking metrics: throughput and delay. We often need more throughputs and less delay.

Reliability

In addition to accuracy of delivery, network reliability is measured by the frequency of failure, the time it takes a link to recover from a failure.

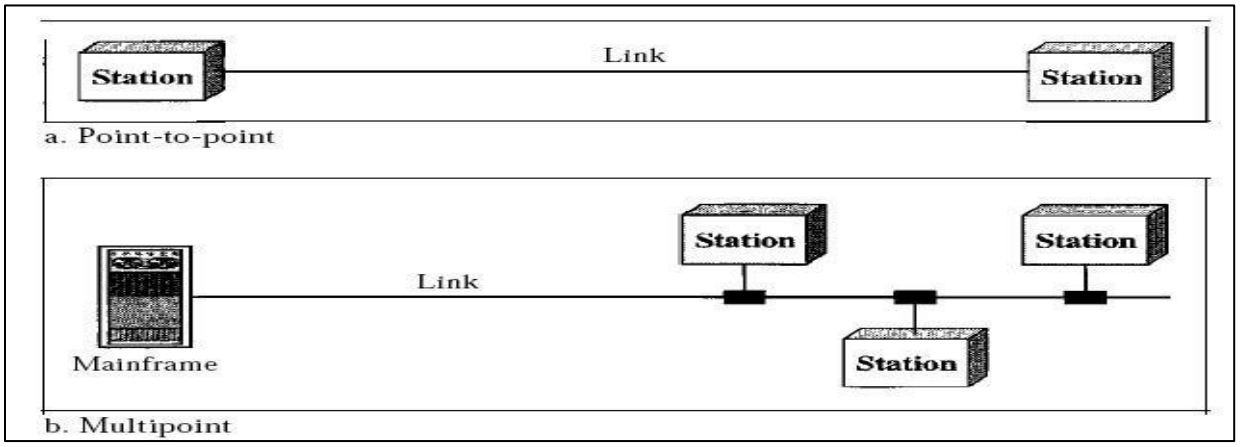
Security

Network security issues include protecting data from unauthorized access, protecting data from damage and development, and implementing policies and procedures for recovery from breaches and data losses.

Type of Connection

A network is two or more devices connected through links. A link is a communications pathway that transfers data from one device to another. For visualization purposes, it is simplest to imagine any link as a line drawn between two points.

Point-to-Point: A point-to-point connection provides a dedicated link between two devices. The entire capacity of the link is reserved for transmission between those two devices.



Multipoint: A multipoint (also called multi drop) connection is one in which more than two specific devices share a single link. In a multipoint environment, the capacity of the channel is shared, either spatially or temporally.

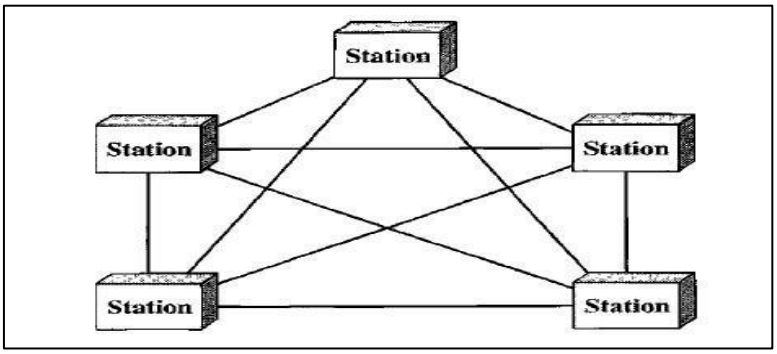
Physical Topology

The term physical topology refers to the way in which a network is laid out physically: I/O or more devices connect to a link; two or more links form a topology.

There are four basic topologies possible: mesh, star, bus, and ring.

Mesh Topology

- Mesh In a mesh topology, every device has a dedicated point-to-point link to every other device. The term dedicated means that the link carries traffic only between the two devices it connects.
- Node 1 must be connected to n - 1nodes, node 2 must be connected to n – 1 nodes, and finally node n must be connected to n - 1 nodes. We need n (n - 1) physical links.



Advantages

- The use of dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problems that can occur when links must be shared by multiple devices.
- A mesh topology is robust. If one link becomes unusable, it does not incapacitate the entire system.
- There is the advantage of privacy or security. When every message travels along a dedicated line, only the intended recipient sees it.
- Point-to-point links make fault identification and fault isolation easy.

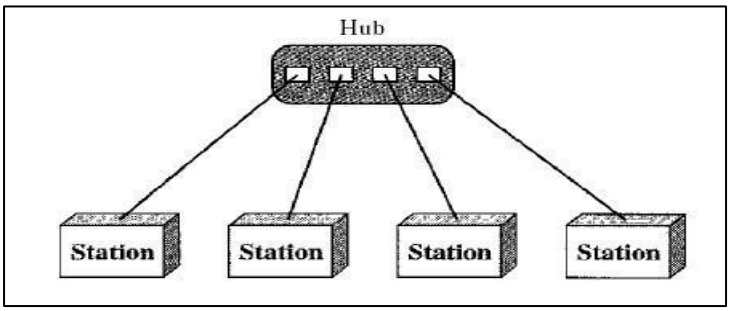
Disadvantages

- Amount of cabling and the number of I/O ports required.
- The sheer bulk of the wiring can be greater than the available space (in walls, ceilings, or floors) can accommodate.
- The hardware required to connect each link (I/O ports and cable) can be prohibitively expensive.

Star Topology

- In a star topology, each device has a dedicated point-to-point link only to a central controller, usually called a hub.
- The devices are not directly linked to one another. Unlike a mesh topology, a star topology does not allow direct traffic between devices.

- The controller acts as an exchange: If one device wants to send data to another, it sends the data to the controller, which then relays the data to the other connected device.



Advantages

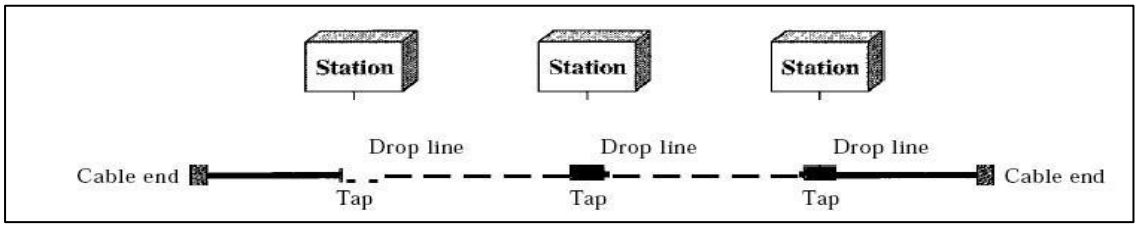
- A star topology is less expensive than a mesh topology.
- In a star, each device needs only one link and one I/O port to connect it to any number of others. This factor also makes it easy to install and reconfigure.
- Far less cabling needs to be housed, and additions, moves, and deletions involve only one connection: between that device and the hub.
- Other advantages include robustness. If one link fails, only that link is affected. All other links remain active.

Disadvantages

- The dependency of the whole topology bon one single point, the hub. If the hub goes down, the whole system is dead.
- Although a star requires far less cable than a mesh, each node must be linked to a central hub.

Bus Topology

The preceding examples all describe point-to-point connections. A **bus topology,** on the other hand, is multipoint. One long cable acts as a **backbone** to link all the devices in a network.



Advantages

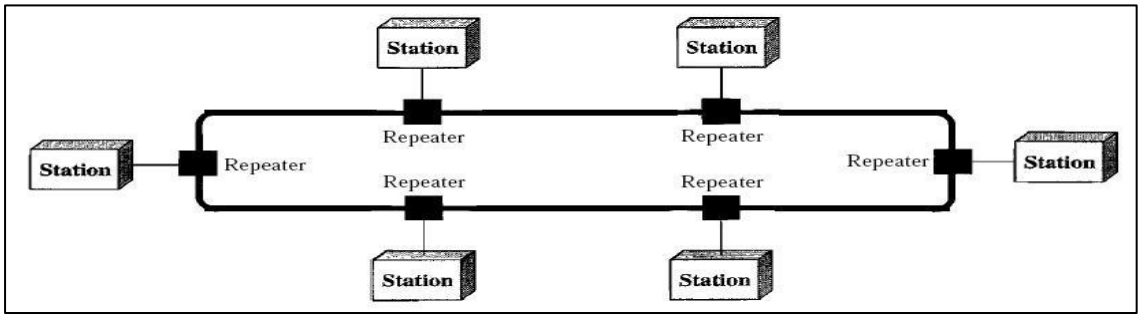
- Bus topology includes ease of installation.
- A bus uses less cabling than mesh or star topologies.

Disadvantages

- Difficult reconnection and fault isolation.
- A fault or break in the bus cable stops all transmission, even between devices on the same side of the problem.

Ring topology

Ring Topology In a ring topology, each device has a dedicated point-to-point connection with only the two devices on either side of it. A signal is passed along the ring in one direction, from device to device, until it reaches its destination.



Advantages

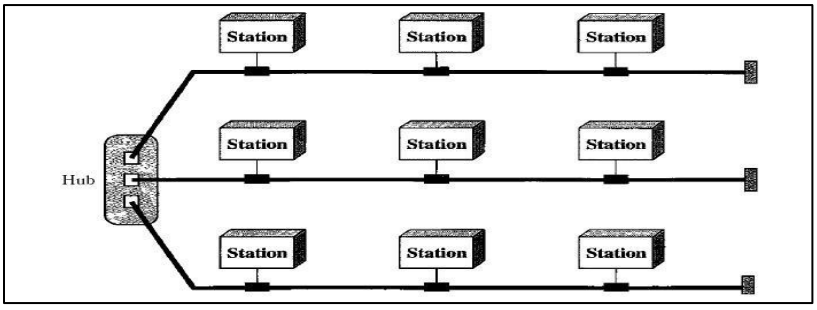
- A ring is relatively easy to install and reconfigure.
- Fault isolation is simplified.

Disadvantages

- Unidirectional traffic can be a disadvantage.
- In a simple ring, a break in the ring (such as a disabled station) can disable the entire network.

Hybrid Topology

A network can be hybrid. For example, we can have a main star topology with each branch connecting several stations in a bus topology.



Categories of networks

Today when we speak of networks, we are generally referring to two primary categories:

- Local area networks (**LAN**) and wide-area networks (**WAN**).
- The category into which a network falls is determined by its size.
- A LAN normally covers an area less than 2 mi; a WAN can be worldwide. Networks of a size in between are normally referred to as metropolitan area networks and span tens of miles.

A **local area network** (LAN) is usually privately owned and links the devices in a single office, building, or campus. LANs are designed to allow resources to be shared between personal computers or workstations. The resources to be shared can include hardware (e.g., a printer), software (e.g., an application program), or data. A common example of a LAN, found in many business environments, links a workgroup of task-related computers, for example, engineering workstations or accounting PCs.

A **wide area network** (WAN) provides long-distance transmission of data, image, audio, and video information over large geographic areas that may comprise a country, a continent, or even the whole world.

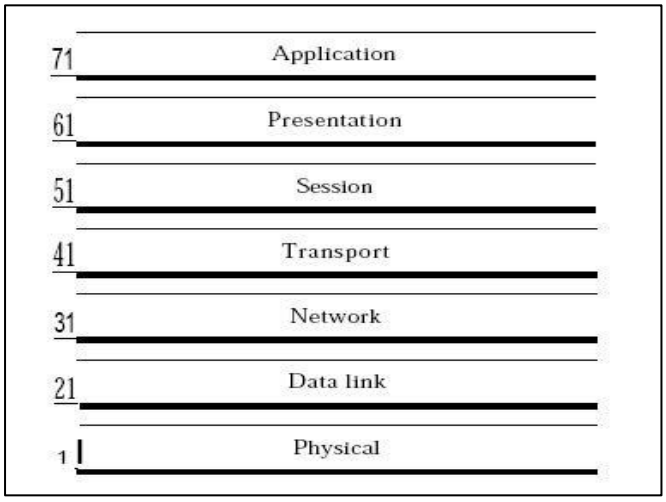
A **metropolitan area network** (MAN) is a network with a size between a LAN and a WAN. It normally covers the area inside a town or a city. It is designed for customers who need a high-speed connectivity, normally to the Internet, and have endpoints spread over a city or part of city. A good example of a MAN is the part of the telephone company network that can provide a high-speed DSL line to the customer.

CHAPTER 2

The OSI Model

An open system is a set of protocols that allows any two different systems to communicate regardless of their underlying architecture.

The purpose of the OSI model is to show how to facilitate communication between different systems without requiring changes to the logic of the underlying hardware and software. The OSI model is not a protocol; it is a model for understanding and designing a network architecture that is flexible, robust, and interoperable.



Physical Layer

The physical layer coordinates the functions required to carry a bit stream over a physical medium. It deals with the mechanical and electrical specifications of the interface and transmission medium.

The physical layer is also concerned with the following:

- **Physical characteristics of interfaces and medium:** The physical layer defines the characteristics of the interface between the devices and the transmission medium. It also defines the type of transmission medium.
- **Representation of bits:** The physical layer data consists of a stream of bits (sequence of Os or 1s) with no interpretation. To be transmitted, bits must be encoded into signals--electrical or optical. The physical layer defines the type of encoding (how Os and I s are changed to signals).
- **Data rate:** The transmission rate-the number of bits sent each second-is also defined by the physical layer. In other words, the physical layer defines the duration of a bit, which is how long it lasts.

- **Synchronization of bits:** The sender and receiver not only must use the same bit rate but also must be synchronized at the bit level. In other words, the sender and the receiver clocks must be synchronized.
- **Line configuration:** The physical layer is concerned with the connection of devices to the media. In a point-to-point configuration, two devices are connected through a dedicated link. In a multipoint configuration, a link is shared among several devices.
- **Physical topology:** The physical topology defines how devices are connected to make a network. Devices can be connected by using a mesh topology (every device is connected to every other device), a star topology (devices are connected through a central device), a ring topology (each device is connected to the next, forming a ring), a bus topology (every device is on a common link), or a hybrid topology (this is a combination of two or more topologies).
- **Transmission mode:** The physical layer also defines the direction of transmission between two devices: simplex, half-duplex, or full-duplex. In simplex mode, only one device can send; the other can only receive. The simplex mode is a one-way communication. In the half-duplex mode, two devices can send and receive, but not at the same time. In a full-duplex (or simply duplex) mode, two devices can send and receive at the same time.

Data Link Layer

The data link layer transforms the physical layer, a raw transmission facility, to a reliable link. It makes the physical layer appear error-free to the upper layer (network layer).
Other responsibilities of the data link layer include the following:

- **Framing:** The data link layer divides the stream of bits received from the network layer into manageable data units called frames.
- **Physical addressing:** If frames are to be distributed to different systems on the network, the data link layer adds a header to the frame to define the sender and/or receiver of the frame. If the frame is intended for a system outside the sender's network, the receiver address is the address of the device that connects the network to the next one.
- **Flow control:** If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender, the data link layer imposes a flow control mechanism to avoid overwhelming the receiver.
- **Error control:** The data link layer adds reliability to the physical layer by adding mechanisms to detect and retransmit damaged or lost frames. It also uses a mechanism to recognize duplicate frames. Error control is normally achieved through a trailer added to the end of the frame.

- **Access control:** When two or more devices are connected to the same link, data link layer protocols are necessary to determine which device has control over the link at any given time.

Network Layer

The network layer is responsible for the source-to-destination delivery of a packet, possibly across multiple networks (links). Whereas the data link layer oversees the delivery of the packet between two systems on the same network (links), the network layer ensures that each packet gets from its point of origin to its final destination.

Other responsibilities of the network layer include the following:

- **Logical addressing.** The physical addressing implemented by the data link layer handles the addressing problem locally. If a packet passes the network boundary, we need another addressing system to help distinguish the source and destination systems. The network layer adds a header to the packet coming from the upper layer that, among other things, includes the logical addresses of the sender and receiver. We discuss logical addresses later in this chapter.
- **Routing.** When independent networks or links are connected to create internetworks (network of networks) or a large network, the connecting devices (called routers or switches) route or switch the packets to their final destination. One of the functions of the network layer is to provide this mechanism.

Transport Layer

The transport layer is responsible for process-to-process delivery of the entire message. A process is an application program running on a host. Whereas the network layer oversees source-to-destination delivery of individual packets, it does not recognize any relationship between those packets.

Other responsibilities of the transport layer include the following:

- **Service-point addressing:** Computers often run several programs at the same time. For this reason, source-to-destination delivery means delivery not only from one computer to the next but also from a specific process (running program) on one computer to a specific process (running program) on the other. The transport layer header must therefore include a type of address called a service-point address (or port address). The network layer gets each packet to the correct computer; the transport layer gets the entire message to the correct process on that computer.
- **Segmentation and reassembly:** A message is divided into transmittable segments, with each segment containing a sequence number. These numbers enable the transport layer to reassemble the message correctly upon arriving at the destination and to identify and replace packets that were lost in transmission.

- **Connection control:** The transport layer can be either connectionless or connection oriented. A connectionless transport layer treats each segment as an independent packet and delivers it to the transport layer at the destination machine. A connection oriented transport layer makes a connection with the transport layer at the destination machine first before delivering the packets. After all the data are transferred, the connection is terminated.
    - **Flow control:** Like the data link layer, the transport layer is responsible for flow control.
    - However, flow control at this layer is performed end to end rather than across a single link.
- **Error control:** Like the data link layer, the transport layer is responsible for error control. However, error control at this layer is performed process-to process rather than across a single link. The sending transport layer makes sure that the entire message arrives at the receiving transport layer without error (damage, loss, or duplication). Error correction is usually achieved through retransmission.

Session Layer

The services provided by the first three layers (physical, data link, and network) are not sufficient for some processes. The session layer is the network dialog controller. It establishes, maintains, and synchronizes the interaction among communicating systems. The session layer is responsible for dialog control and synchronization.
Specific responsibilities of the session layer include the following:

- **Dialog control:** The session layer allows two systems to enter into a dialog. It allows the communication between two processes to take place in either half duplex (one way at a time) or full-duplex (two ways at a time) mode.
- **Synchronization:** The session layer allows a process to add checkpoints, or synchronization points, to a stream of data. For example, if a system is sending a file of 2000 pages, it is advisable to insert checkpoints after every 100 pages to ensure that each 100-page unit is received and acknowledged independently. In this case, if a crash happens during the transmission of page 523, the only pages that need to be resent after system recovery are pages 501 to 523. Pages previous to 501 need not be resent.

Presentation Layer

The presentation layer is concerned with the syntax and semantics of the information exchanged between two systems.
Specific responsibilities of the presentation layer include the following:

- **Translation:** The processes (running programs) in two systems are usually exchanging information in the form of character strings, numbers, and so on. The

information must be changed to bit streams before being transmitted. Because different computers use different encoding systems, the presentation layer is responsible for interoperability between these different encoding methods. The presentation layer at the sender 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:** To carry sensitive information, a system must be able to ensure privacy. Encryption means that the sender transforms the original information another form and sends the resulting message out over the network. Decryption reverses the original process to transform the message back to its original form.
- **Compression:** Data compression reduces the number of bits contained in the information. Data compression becomes particularly important in the transmission of multimedia such as text, audio, and video.

Application Layer

The application layer enables the user, whether human or 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.

CHAPTER 3

ANALOG AND DIGITAL

- One of the major functions of the physical layer is to move data in the form of electromagnetic signals across a transmission medium.
- Both data and the signals that represent them can be either **analog or digital** in form.

Analog and Digital Data

- 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.
- For example, an analog clock that has hour, minute, and second hands gives information in a continuous form; the movements of the hands are continuous. On the other hand, a digital clock that reports the hours and the minutes will change suddenly from 8:05 to 8:06.
- Analog data, such as the sounds made by a human voice, take on continuous values. When someone speaks, an analog wave is created in the air. This can be captured by a microphone and converted to an analog signal or sampled and converted to a digital signal.
- Digital data take on discrete values. For example, data are stored in computer memory in the form of Os and 1s. They can be converted to a digital signal or modulated into an analog signal for transmission across a medium.

Analog and Digital Signals

- Like the data they represent, signals can be either analog or digital. An analog signal has infinitely many levels of intensity over a period of time.
- A digital signal, on the other hand, can have only a limited number of defined values. Although each value can be any number, it is often as simple as 1 and O.

Periodic and Non Periodic Signals

- Both analog and digital signals can take one of two forms: periodic or non periodic.
- A periodic signal completes a pattern within a measurable time frame, called a period, and repeats that pattern over subsequent identical periods. The completion of one full pattern is called a cycle.
- A non periodic signal changes without exhibiting a pattern or cycle that repeats over time.
- Both analog and digital signals can be periodic or non periodic. In data communications, we commonly use periodic analog signals (because they need less

bandwidth) and non periodic digital signals (because they can represent variation in data).

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.

Sine Wave

- The sine wave is the most fundamental form of a periodic analog signal. When we visualize it as a simple oscillating curve, its change over the course of a cycle is smooth and consistent, a continuous, rolling flow.
- A sine wave can be represented by three parameters: the peak amplitude, the frequency, and the phase. These three parameters fully describe a sine wave.

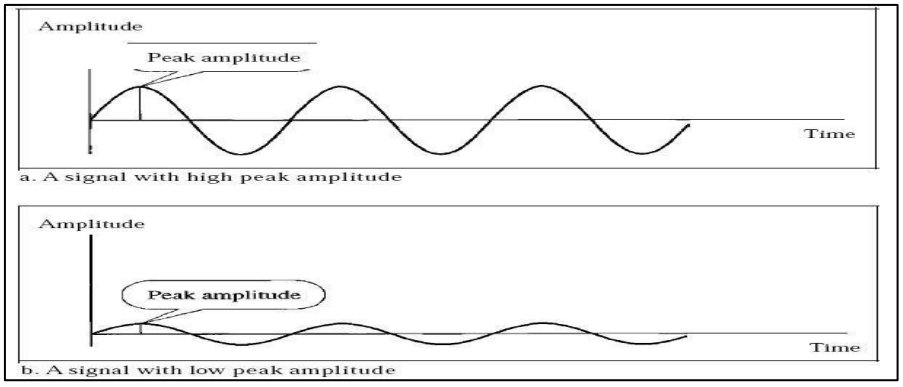
Peak Amplitude

The peak amplitude of a signal is the absolute value of its highest intensity, proportional to the energy it carries. For electric signals, peak amplitude is normally measured in volts.

Period and Frequency

- Period refers to the amount of time, in seconds, a signal needs to complete 1 cycle. Frequency refers to the number of periods in 1s.
- Period is the inverse of frequency, and frequency is the inverse of period, as the following formulas show.
  $f = 1/T$ and $T = 1/f$

Unit	Equivalent	Unit	Equivalent
Seconds (s)	1 s	Hertz (Hz)	1 Hz
Milliseconds (ms)	10^{-3} s	Kilohertz (kHz)	10^{3} Hz
Microseconds (μs)	10^{-6} s	Megahertz (MHz)	10^{6} Hz
Nanoseconds (ns)	10^{-9} s	Gigahertz (GHz)	10^{9} Hz
Picoseconds (ps)	10^{-12} s	Terahertz (THz)	10^{12} Hz

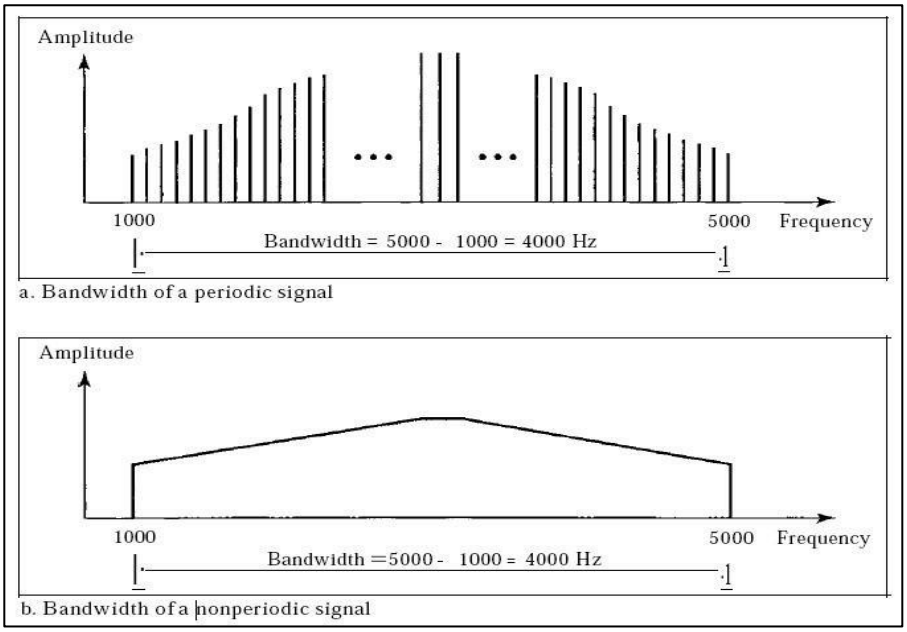


Composite Signals

- Simple sine waves have many applications in daily life. We can send a single sine wave to carry electric energy from one place to another. For example, the power company sends a single sine wave with a frequency of 60 Hz to distribute electric energy to houses and businesses.
- 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.
- According to Fourier analysis, any composite signal is a combination of simple sine waves with different frequencies, amplitudes, and phases.

Bandwidth

- The range of frequencies contained in a composite signal is its **bandwidth**. The bandwidth is normally a difference between two numbers. For example, if a composite signal contains frequencies between 1000 and 5000, its bandwidth is 5000 - 1000, or 4000.
- 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 :

A periodic signal has a bandwidth of 20 Hz. The highest frequency is 60 Hz. What is the lowest frequency? Draw the spectrum if the signal contains all frequencies of the same amplitude.

Solution:

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

$B = f_h - f_z$

20 =60 - f_z or f_z =60 - 20=40 Hz

Digital Signal

In addition to being represented by an analog signal, information can also be represented by a digital signal. For example, 1 can be encoded as a positive voltage and a 0 as zero voltage. A digital signal can have more than two levels.
In general, if a signal has L levels, each level needs **$\log_2 L$** bits.

Example:

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

Solution:

We calculate the number of bits from the formula:

Number of bits per level =$\log_2 8$ =3

Each signal level is represented by 3 bits.

Bit Rate

- Most digital signals are non periodic, and thus period and frequency are not appropriate characteristics. Another term-bit rate is used to describe digital signals.
- The bit rate is the number of bits sent in 1s, expressed in bits per second (**bps**).

Example :

Assume we need to download text documents at the rate of 100 pages per minute. What is the required bit rate of the channel?

Solution:

A page is an average of 24 lines with 80 characters in each line. If we assume that one character requires

8 bits, the bit rate is

100 x 24 x 80 x 8 =1,636,000 bps =1.636 Mbps

Bit Length

We discussed the concept of the wavelength for an analog signal: the distance one cycle occupies on the transmission medium. We can define something similar for a digital signal: the bit length. The bit length is the distance one bit occupies on the transmission medium.

Bit length=propagation speed x bit duration

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)

Two theoretical formulas were developed to calculate the data rate: one by Nyquist for a noiseless channel, another by Shannon for a noisy channel.

Noiseless Channel: Nyquist Bit Rate

For a noiseless channel, the Nyquist bit rate formula defines the theoretical maximum bit rate

BitRate = 2 x bandwidth x $l0g_2$ L

In this formula, bandwidth is the bandwidth of the channel, L is the number of signal levels used to represent data, and Bit Rate is the bit rate in bits per second.

Example:

Consider a noiseless channel with a bandwidth of 3000 Hz transmitting a signal with two signal levels.

The maximum bit rate can be calculated as

BitRate =2 x 3000 x $\log_2$ 2 =6000 bps

Noisy Channel: Shannon Capacity

In reality, we cannot have a noiseless channel; the channel is always noisy. In 1944, Claude Shannon introduced a formula, called the Shannon capacity, to determine the theoretical highest data rate for a noisy channel:

Capacity =bandwidth X $\log_2$ (1 +SNR)

In this formula, bandwidth is the bandwidth of the channel, SNR is the signal-to-noise ratio, and capacity is the capacity of the channel in bits per second. Note that in the Shannon formula there is no indication of the signal level, which means that no matter how many levels we have, we cannot achieve a data rate higher than the capacity of the channel. In other words, the formula defines a characteristic of the channel, not the method of transmission.

Example:

Consider an extremely noisy channel in which the value of the signal-to-noise ratio is almost zero. In other words, the noise is so strong that the signal is faint. For this channel the capacity C is calculated as

C=B $\log_2$ (1 + SNR) =B $l0g_2$ (1 + 0) =B $\log_2$ 1 = B x 0 =0

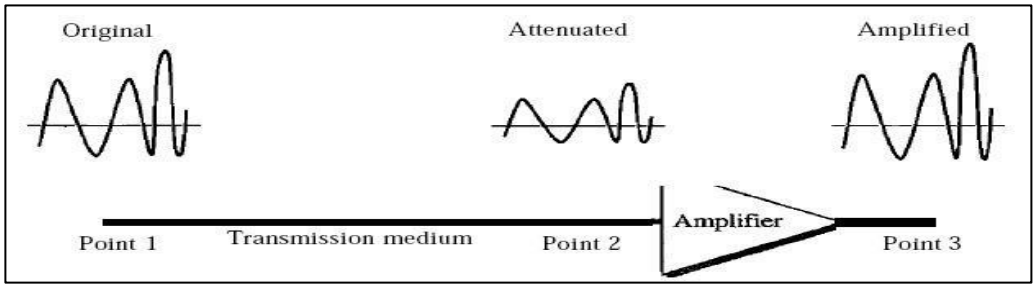
This means that the capacity of this channel is zero regardless of the bandwidth. In other words, we cannot receive any data through this channel.

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

Attenuation means a loss of energy. When a signal, simple or composite, travels through a medium, it loses some of its energy in overcoming the resistance of the medium. That is why a wire carrying electric signals gets warm, if not hot, after a while. Some of the electrical energy in the signal is converted to heat. To compensate for this loss, amplifiers are used to amplify the signal.



Decibel

- To show that a signal has lost or gained strength, engineers use the unit of the decibel.
- The decibel (dB) measures the relative strengths of two signals or one signal at two different points.
- Note that the decibel is negative if a signal is attenuated and positive if a signal is amplified.

$$\mathbf{dB = 10 \log_{10} \frac{P_2}{P_1}}$$

Variables PI and P2 are the powers of a signal at points 1 and 2, respectively.

Example:

Suppose a signal travels through a transmission medium and its power is reduced to one-half. Find the attenuation (loss of power).

Solution:

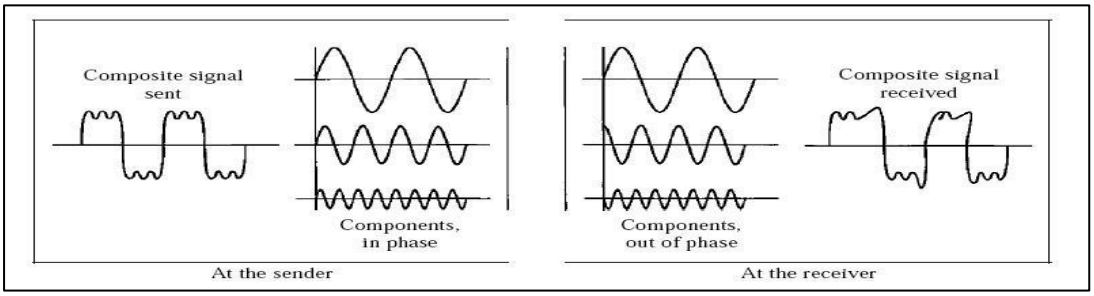
dB=10 log (P/2P)= -3 dB

Example:

A signal travels through an amplifier, and its power is increased 10 times Find the amplification (gain of power). **Solution:**
dB=10 log (10P/P)= 10 dB

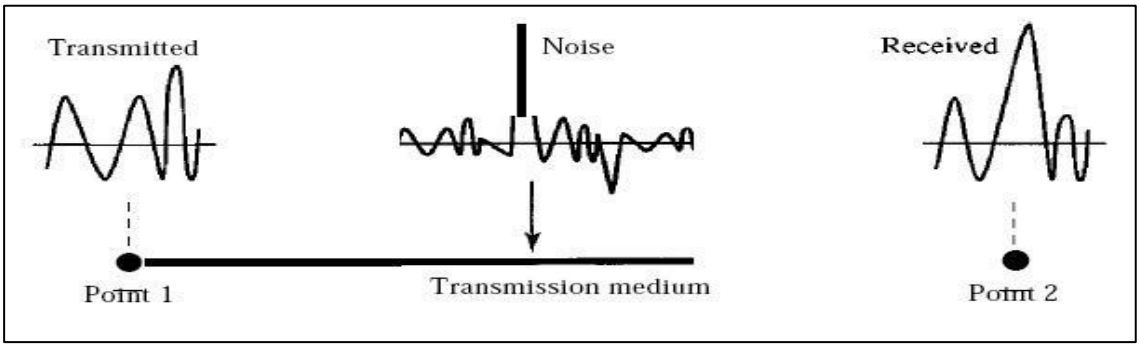
Distortion

- **Distortion** means that the signal changes its form or shape.
- Distortion can occur in a composite signal made of different frequencies. Each signal component has its own propagation speed (see the next section) through a medium and, therefore, its own delay in arriving at the final destination. Differences in delay may create a difference in phase if the delay is not exactly the same as the period duration.
- In other words, signal components at the receiver have phases different from what they had at the sender. The shape of the composite signal is therefore not the same.



Noise

Noise is another cause of impairment. Several types of noise, such as thermal noise, induced noise, crosstalk, and impulse noise, may corrupt the signal. Thermal noise is the random motion of electrons in a wire which creates an extra signal not originally sent by the transmitter. Induced noise comes from sources such as motors and appliances.



To find the theoretical bit rate limit, we need to know the ratio of the signal power to the noise power.

The signal-to-noise ratio is defined as:

SNR = average signal power/ average noise power

Because SNR is the ratio of two powers, it is often described in decibel units, $\mathbf{SNR_{dB}}$, defined as

$$\mathbf{SNR_{dB}} = 10\log_{10} SNR$$

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 SNRdB?

Solution:

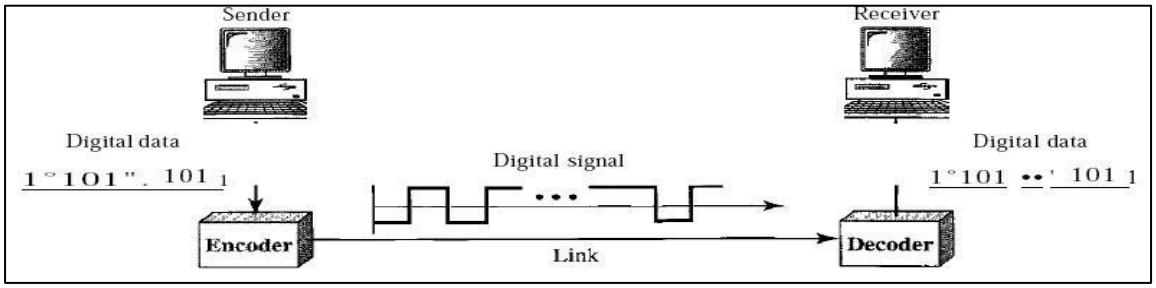
The values of SNR and SNR_{dB} can be calculated as follows: $SNR = 10^{-2}/10^{-6} = 10,000$

$SNR_{dB} = 10 \log_{10} 10^4 = 10 \times 4 = 40$

CHAPTER 4

DIGITAL TRANSMISSION

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 mayor may not be needed.



Line Coding

Line coding is the process of converting digital data to digital signals.

We assume that data, in the form of text, numbers, graphical images, audio, or video, are stored in computer memory as sequences of bits.

Line coding converts a sequence of bits to a digital signal. At the sender, digital data are encoded into a digital signal; at the receiver, the digital data are recreated by decoding the digital signal.

We can formulate the relationship between data rate and signal rate as: **S =c xNx 1/r baud**

where N is the data rate (bps); c is the case factor, which varies for each case; S is the number of signal elements; and r is the previously defined factor.

Example

A signal is carrying data in which one data element is encoded as one signal element (r = 1). If the bit rate is 100 kbps, what is the average value of the baud rate if c is between 0 and 1?

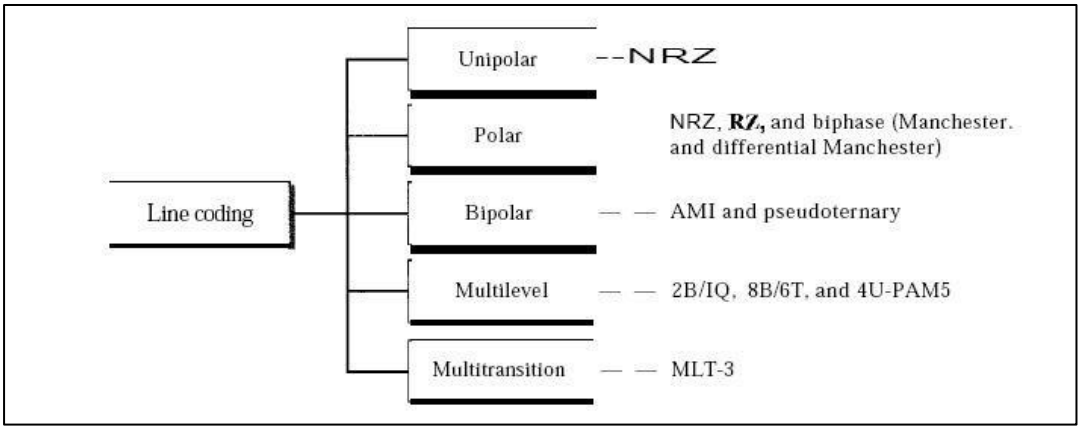
Solution:

We assume that the average value of c is ½. The baud rate is then

S =c x N x1/r = ½ x 100, 000 x 1= 50, 000 = 50 Kbaud

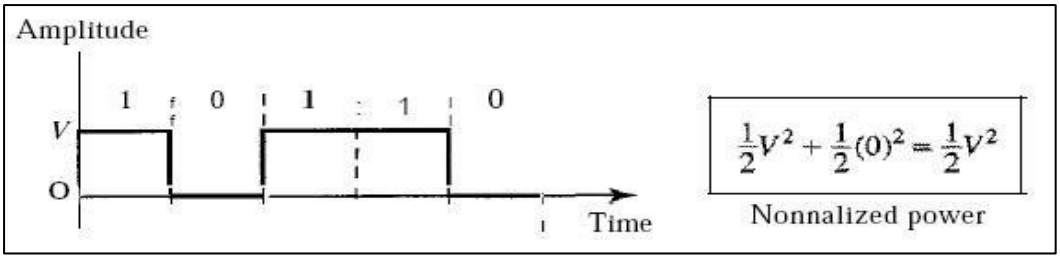
Line Coding Schemes

We can roughly divide line coding schemes into five broad categories, as shown in below:



Unipolar Scheme

In a unipolar scheme, all the signal levels are on one side of the time axis, either above or below. **NRZ** (Non-Return-to-Zero): Traditionally, a unipolar scheme was designed as a non-return-to-zero (NRZ) scheme in which the positive voltage defines bit 1 and the zero voltage defines bit 0. ***It is called NRZ because the signal does not return to zero at the middle of the bit.***



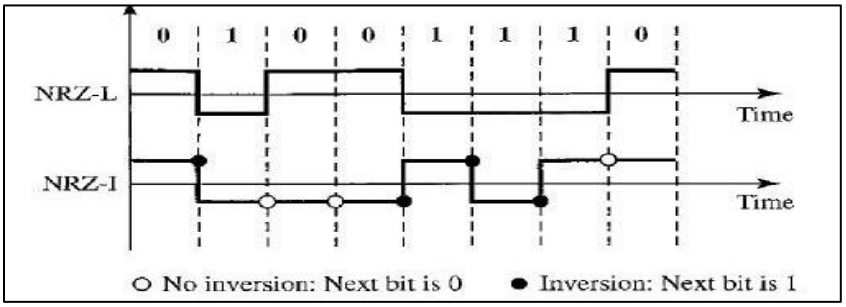
Polar Schemes

In polar schemes, the voltages are on the both sides of the time axis. For example, the voltage level for 0 can be positive and the voltage level for 1 can be negative.

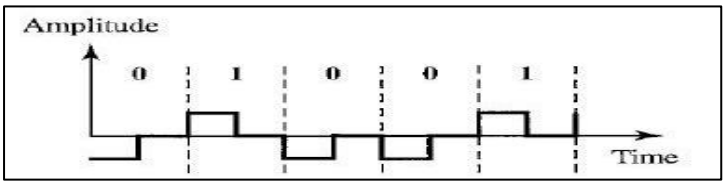
Non-Return-to-Zero (NRZ):

- In polar NRZ encoding, we use two levels of voltage amplitude. We can have two versions of polar NRZ: NRZ-Land NRZ-I.
- In the first variation, **NRZ-L** (NRZ-Level), the level of the voltage determines the value of the bit.
- In the second variation, **NRZ-I** (NRZ-Invert), the change or lack of change in the level of the voltage determines the value of the bit**. If there is no change, the bit is 0; if there is a change, the bit is 1.**
- The synchronization problem (sender and receiver clocks are not synchronized) also exists in both schemes. Again, this problem is more serious in **NRZ-L** than in **NRZ-I**. While a long sequence of 0's can cause a problem in both schemes, a long sequence of 1s affects only **NRZ-L**.
- Another problem with NRZ-L occurs when there is a sudden change of polarity in the system.

NRZ-I does not have this problem. Both schemes have an average signal rate of **N/2 Bd**.

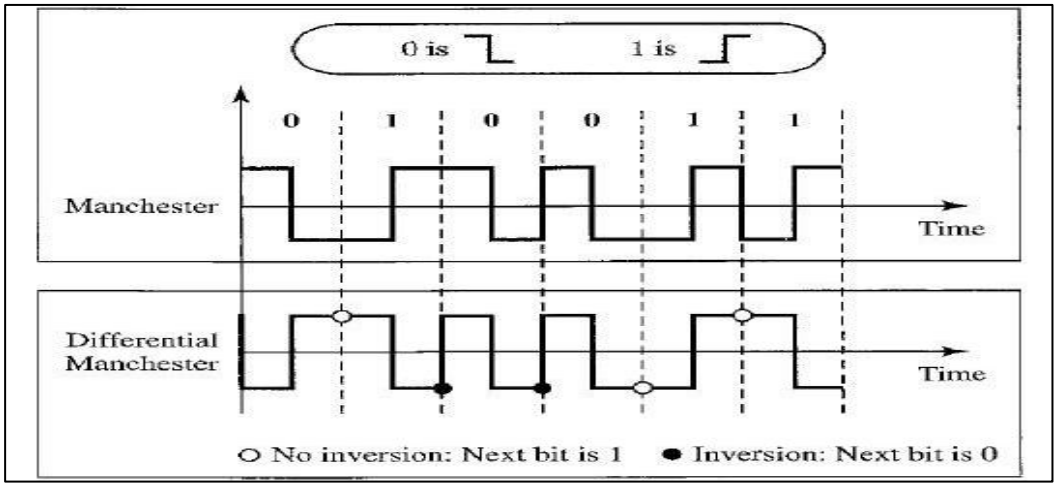


Return to Zero (RZ) The main problem with NRZ encoding occurs when the sender and receiver clocks are not synchronized. The receiver does not know when one bit has ended and the next bit is starting. One solution is the return-to-zero (RZ) scheme, which uses three values: positive, negative, and zero**. In RZ, the signal changes not between bits but during the bit**.



Biphase: Manchester and Differential Manchester

- The idea of RZ (transition at the middle of the bit) and the idea of NRZ-L are combined into the Manchester scheme.
- In Manchester encoding, the duration of the bit is divided into two halves. The voltage remains at one level during the first half and moves to the other level in the second half. The transition at the middle of the bit provides synchronization.
- Differential Manchester, on the other hand, combines the ideas of RZ and NRZ-I. There is always a transition at the middle of the bit, but the bit values are determined at the beginning of the bit. If the next bit is 0, there is a transition; if the next bit is 1, there is none.



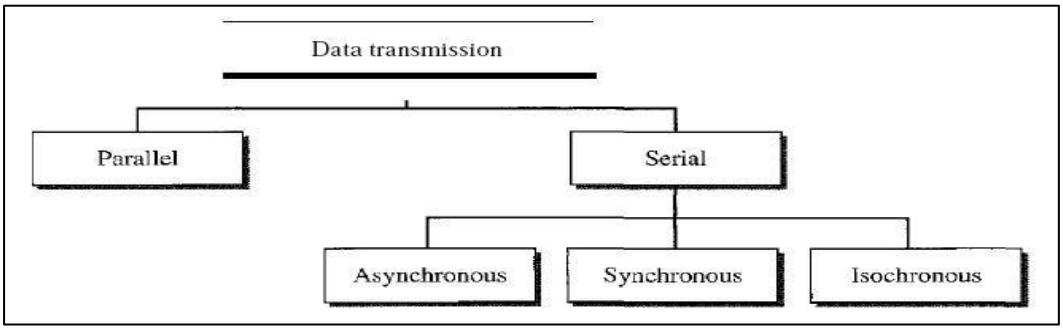
The minimum bandwidth of Manchester and differential Manchester is **2** times that of **NRZ**.

Block Coding

We need redundancy to ensure synchronization and to provide some kind of inherent error detecting. Block coding can give us this redundancy and improve the performance of line coding. In general, block coding changes a block of m bits into a block of n bits, where n is larger than m. Block coding is referred to as an mB/ nB encoding technique.

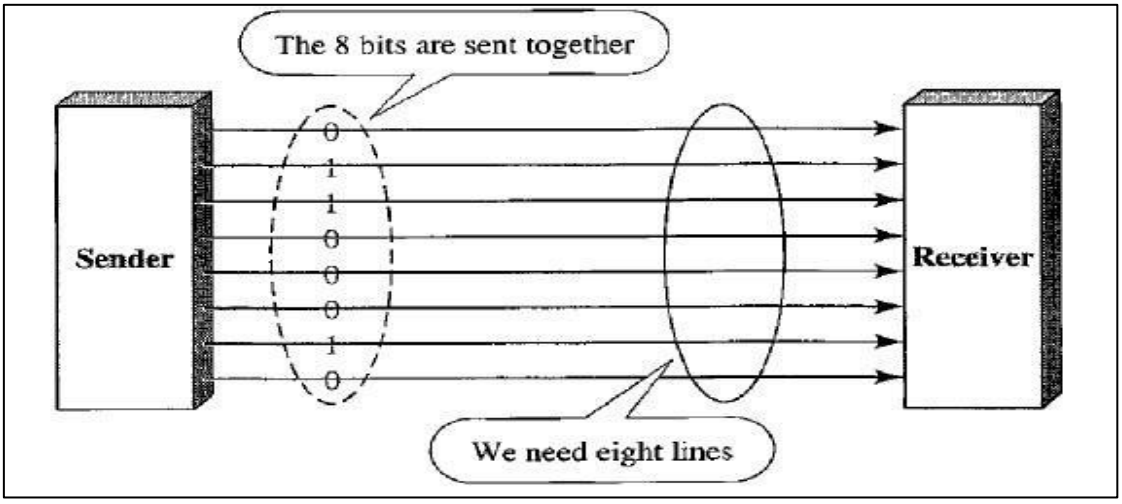
Transmission Modes

Of primary concern when we are considering the transmission of data from one device to another is the wiring, and of primary concern when we are considering the wiring is the data stream. Do we send 1 bit at a time; or do we group bits into larger groups and, if so, how? 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.



Parallel Transmission

Binary data, consisting of 1s and 0s, may be organized into groups of n bits each. Computers produce and consume data in groups of bits much as we conceive of and use spoken language in the form of words rather than letters. By grouping, we can send data n bits at a time instead of 1. This is called parallel transmission.



Advantage:

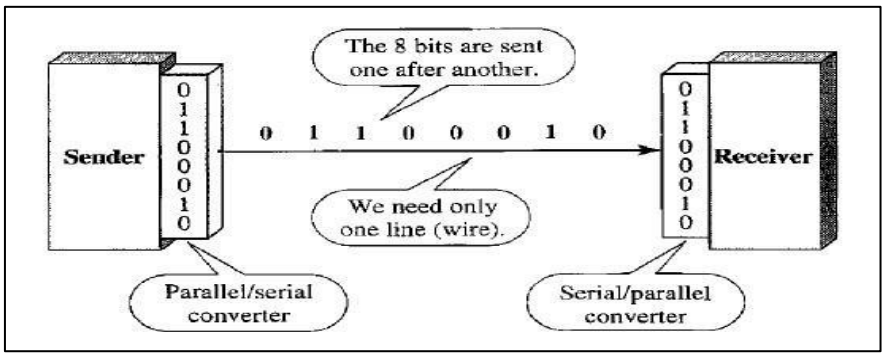
The advantage of parallel transmission is speed. All else being equal, parallel transmission can increase the transfer speed by a factor of *n* over serial transmission.

Disadvantage:

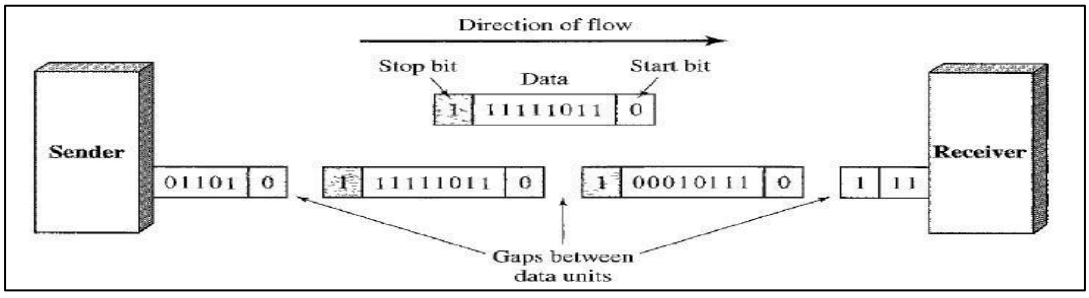
Parallel transmission requires *n* communication lines just to transmit the data stream. Because this is expensive, parallel transmission is usually limited to short distances.

Serial Transmission

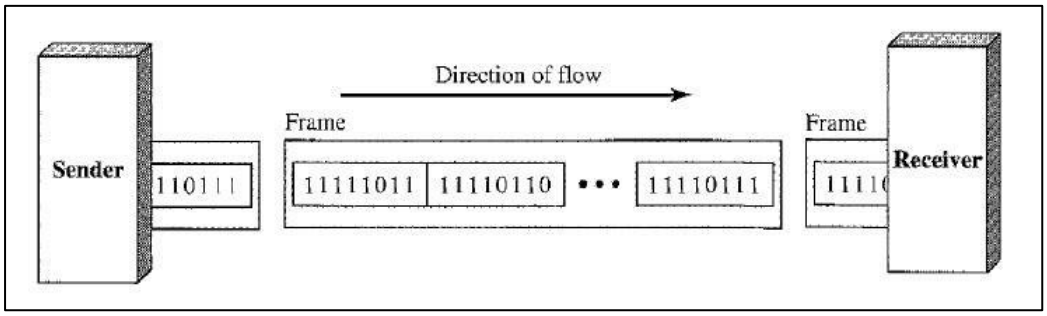
- In serial transmission one bit follows another, so we need only one communication channel rather than n to transmit data between two communicating devices.



- 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.
- Since communication within devices is parallel, conversion devices are required at the interface between the sender and the line (parallel-to-serial) and between the line and the receiver (serial-to-parallel).
- Serial transmission occurs in one of three ways: **asynchronous**, **synchronous**, and **isochronous**.
- **In asynchronous transmission**, we send 1 **start bit (0)** at the beginning and 1 or more **stop bits**
- at the end of each byte. There may be a gap between each byte.



- **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.

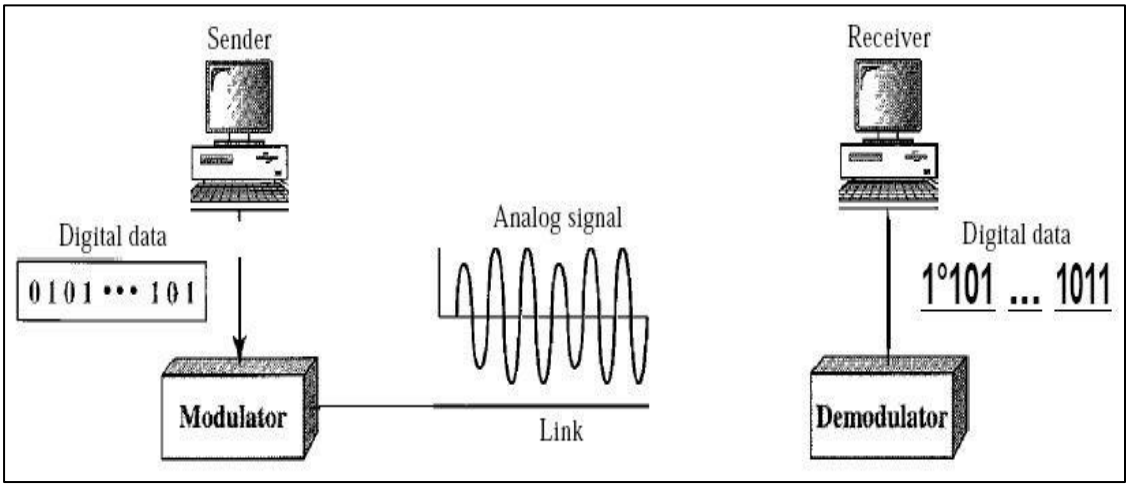


- The **isochronous transmission** guarantees that the data arrive at a fixed rate. In real-time audio and video, in which uneven delays between frames are not acceptable, synchronous transmission fails. For example, TV images are broadcast at the rate of 30 images per second; they must be viewed at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

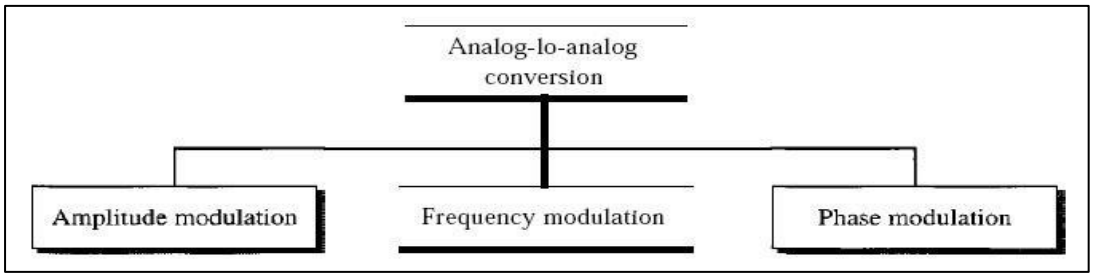
CHAPTER 5

ANALOG TRANSMISSION

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



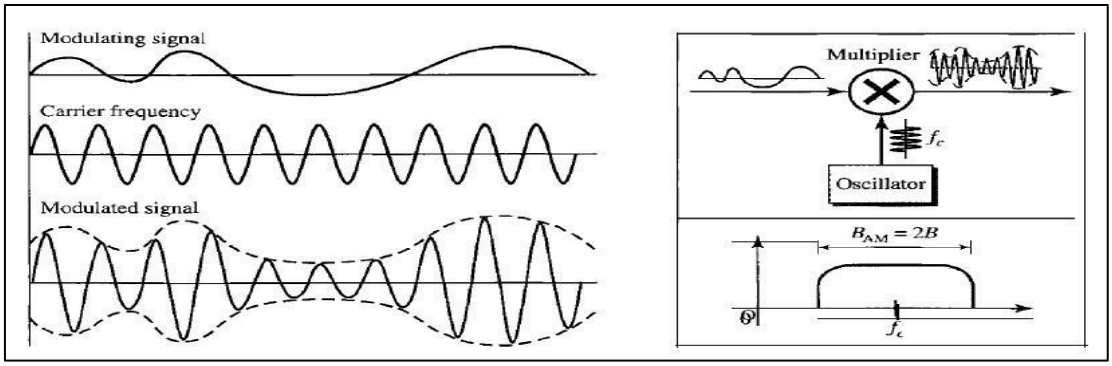
- Analog-to-analog conversion, or analog modulation, is the representation of analog information by an analog signal. One may ask why we need to modulate an analog signal; it is already analog. Modulation is needed if the medium is band pass in nature or if only a band pass channel is available to us. An example is radio. The government assigns a narrow bandwidth to each radio station. The analog signal produced by each station is a low-pass signal, all in the same range. To be able to listen to different stations, the low-pass signals need to be shifted, each to a different range.
- Analog-to-analog conversion can be accomplished in three ways: amplitude modulation (AM), frequency modulation (FM), and phase modulation (PM).



Amplitude Modulation

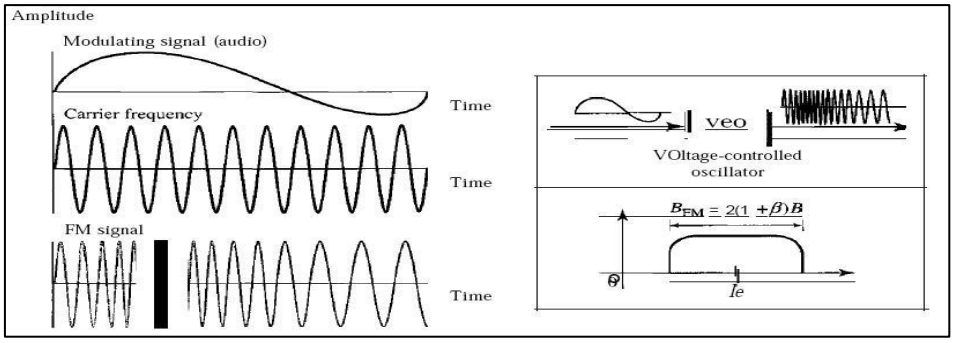
In AM transmission, the carrier signal is modulated so that its amplitude varies with the changing amplitudes of the modulating signal. The frequency and phase of the carrier remain the same; only the amplitude changes to follow variations in the information. Below Figure shows how this concept works.

The modulating signal is the envelope of the carrier.



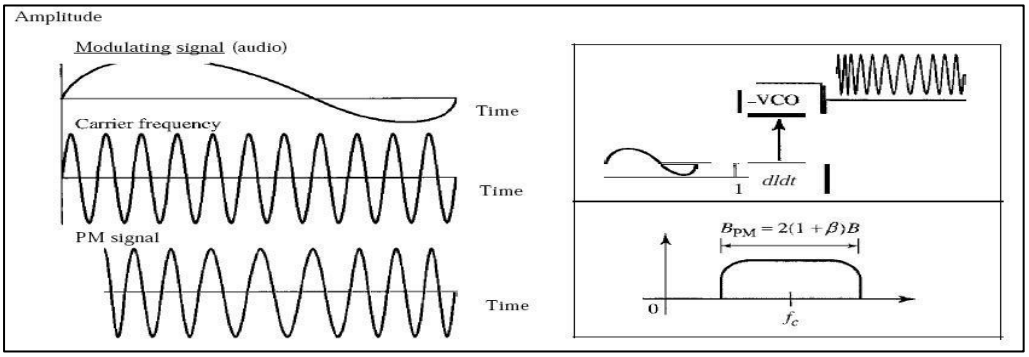
Frequency Modulation

In FM transmission, the frequency of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and phase of the carrier signal remain constant, but as the amplitude of the information signal changes, the frequency of the carrier changes correspondingly.



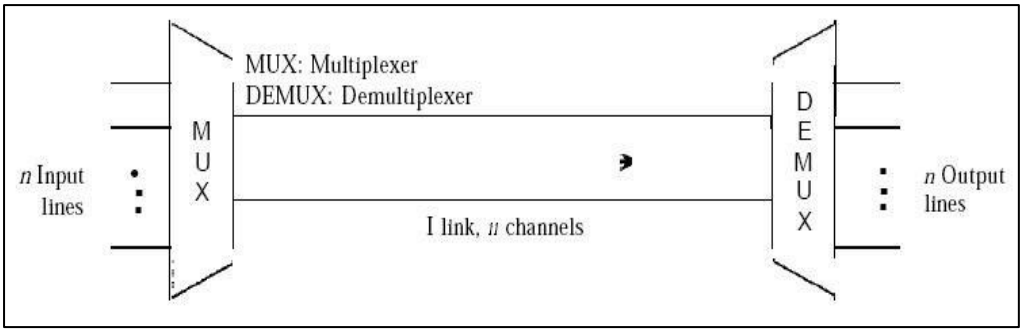
Phase Modulation

In PM transmission, the phase of the carrier signal is modulated to follow the changing voltage level (amplitude) of the modulating signal. The peak amplitude and frequency of the carrier signal remain constant, but as the amplitude of the information signal changes, the phase of the carrier changes correspondingly. In FM, the instantaneous change in the carrier frequency is proportional to the amplitude of the modulating signal; in PM the instantaneous change in the carrier frequency is proportional to the derivative of the amplitude of the modulating signal.

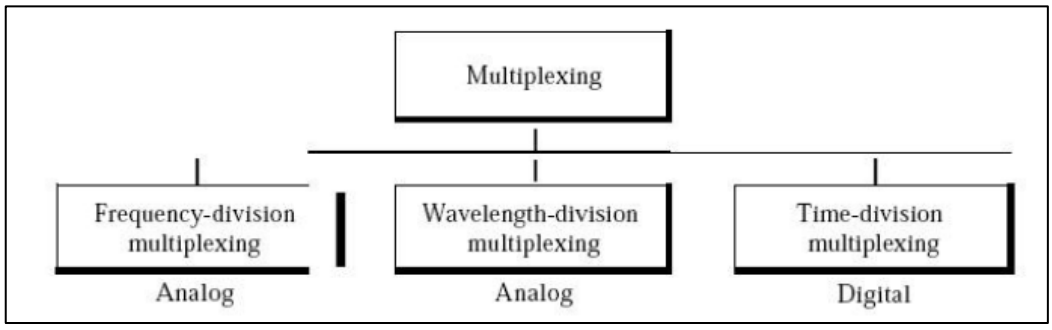


Multiplexing

- 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. We can accommodate this increase by continuing to add individual links each time a new channel is needed; or we can install higher-bandwidth links and use each to carry multiple 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 de-multiplexer (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.



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

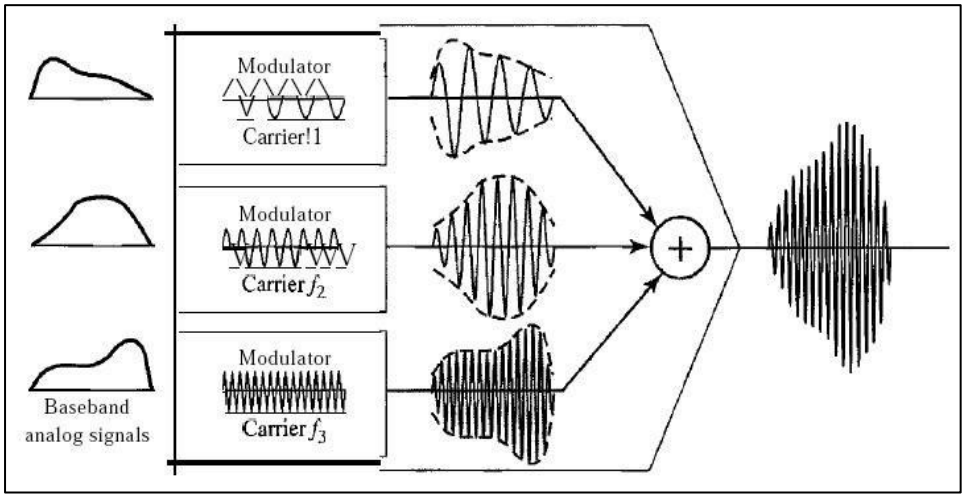


Frequency-Division Multiplexing

- Frequency-division multiplexing (FDM) 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 carrier 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.
- In addition, carrier frequencies must not interfere with the original data frequencies.

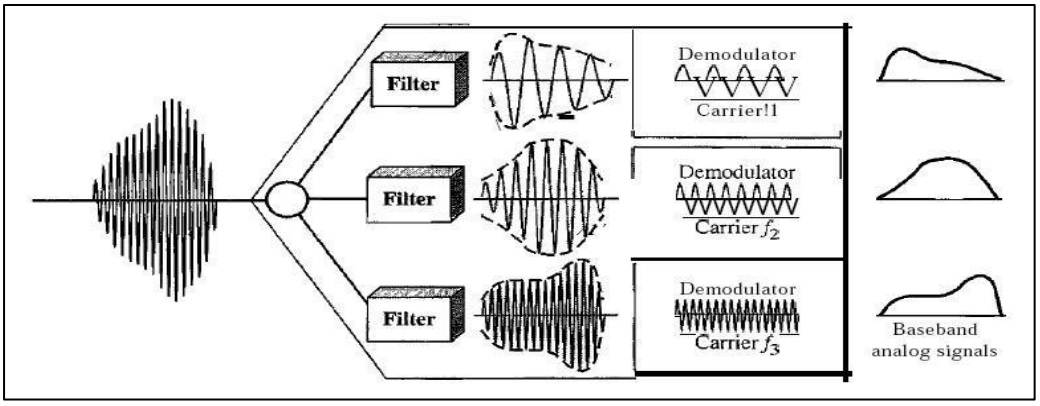
Multiplexing Process

Each source generates a signal of a similar frequency range. Inside the multiplexer, these similar signals modulate different carrier frequencies f1, f2, and f3). The resulting modulated signals are then combined into a single composite signal that is sent out over a media link that has enough bandwidth to accommodate it.



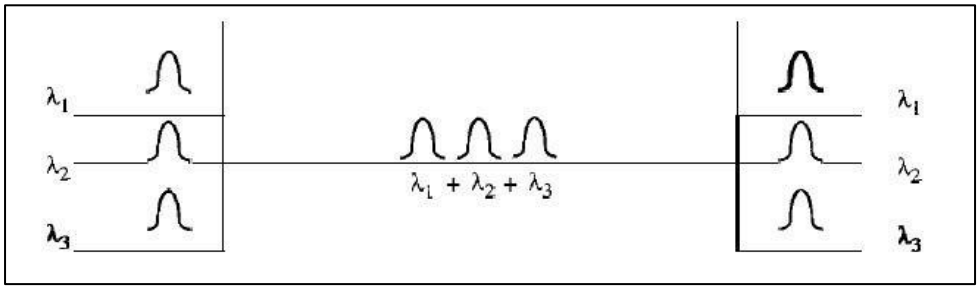
Demultiplexing Process

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.



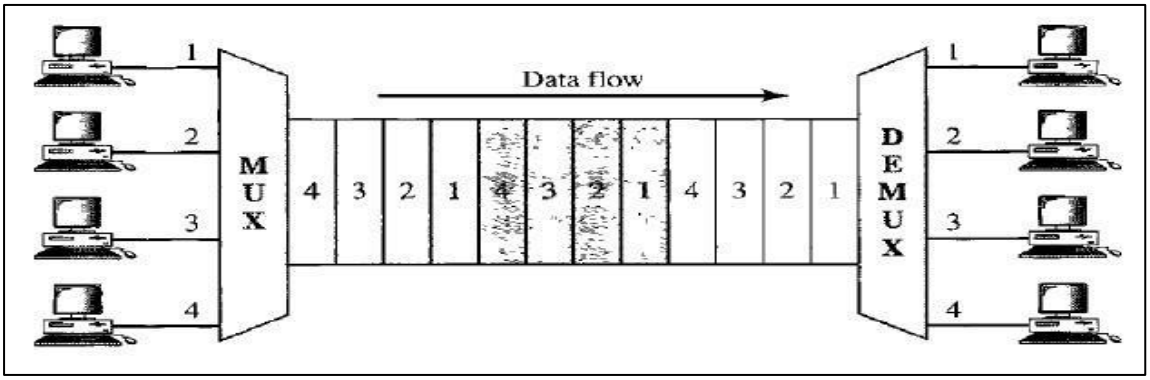
Wavelength-Division Multiplexing

- 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 conceptually the same as FDM, except that the multiplexing and de-multiplexing involve optical signals transmitted through fiber-optic channels. The idea is the same: We are combining different signals of different frequencies. The difference is that the frequencies are very high.



Time-Division Multiplexing

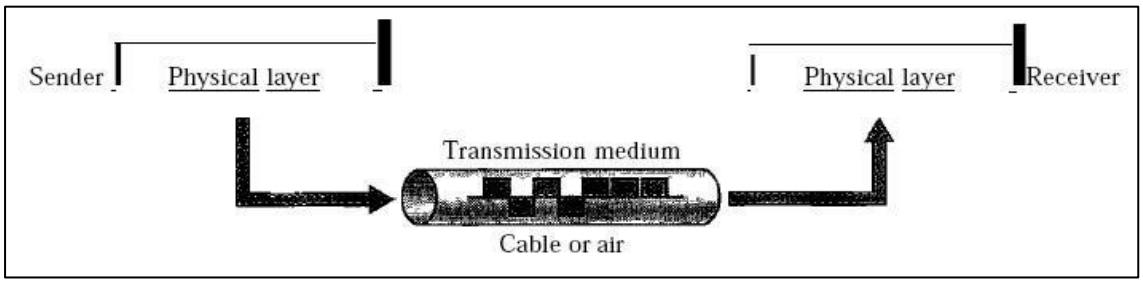
Time-division multiplexing (TDM) is a digital process that allows several connections to share the high bandwidth of a link Instead of sharing a portion of the bandwidth as in FDM, time is shared. Each connection occupies a portion of time in the link.



CHAPTER 6

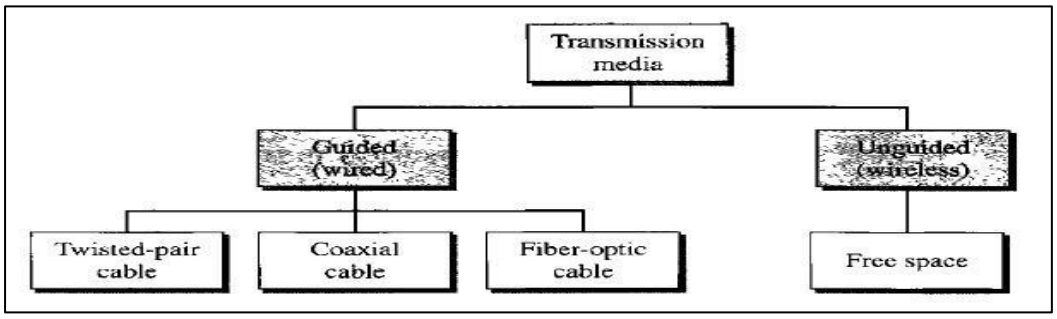
TRANSMISSION MEDIA

- A transmission **medium** can be broadly defined as anything that can carry information from a source to a destination.
- For example, the transmission medium for two people having a dinner conversation is the air. The air can also be used to convey the message in a smoke signal or semaphore. For a written message, the transmission medium might be a mail carrier, a truck, or an airplane.



- In telecommunications, transmission media can be divided into two broad categories: **guided and unguided**. Guided media include **twisted-pair cable**, **coaxial cable**, and **fiber-optic cable**.

Unguided medium is free space. Below Figure shows this taxonomy.

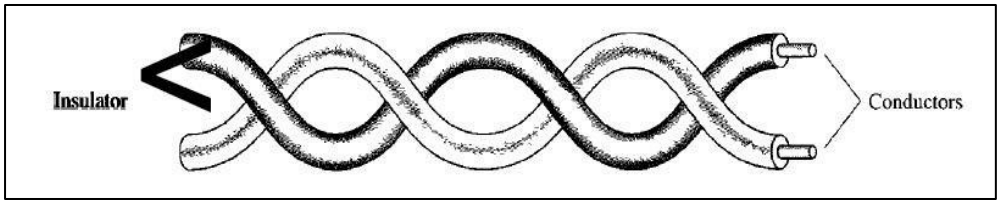


Guided Media

- Guided media, which are those that provide a conduit from one device to another, include twisted-pair cable, coaxial cable, and fiber-optic cable.
- A signal traveling along any of these media is directed and contained by the physical limits of the medium. Twisted-pair and coaxial cable use metallic (copper) conductors that accept and transport signals in the form of electric current.
- Optical fiber is a cable that accepts and transports signals in the form of light.

Twisted-Pair Cable

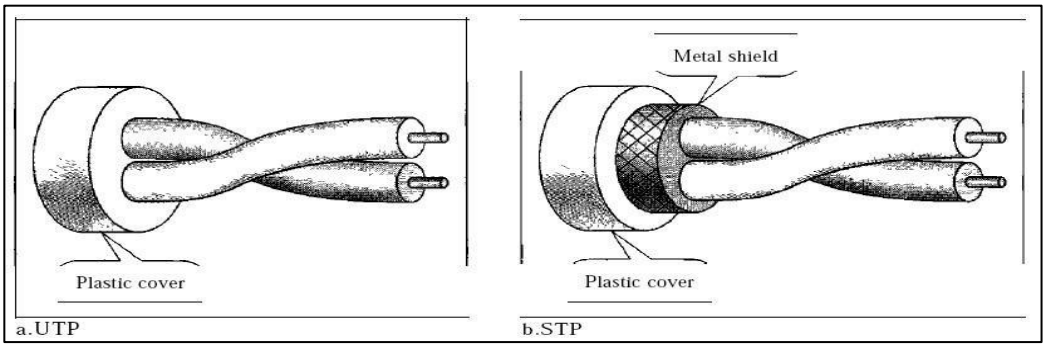
- A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together, as shown in below figure.



- 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.

Unshielded Versus Shielded Twisted-Pair Cable

- 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**.



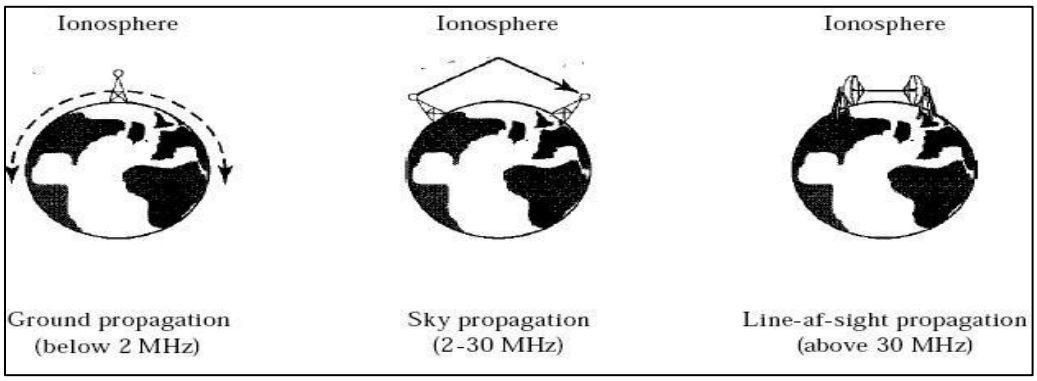
Applications

- Twisted-pair cables are used in telephone lines to provide voice and data channels.
- The local loop-the line that connects subscribers to the central telephone office – commonly consists of unshielded twisted-pair cables.
- The DSL lines that are used by the telephone companies to provide high-data-rate connections also use the high-bandwidth capability of unshielded twisted-pair cables.

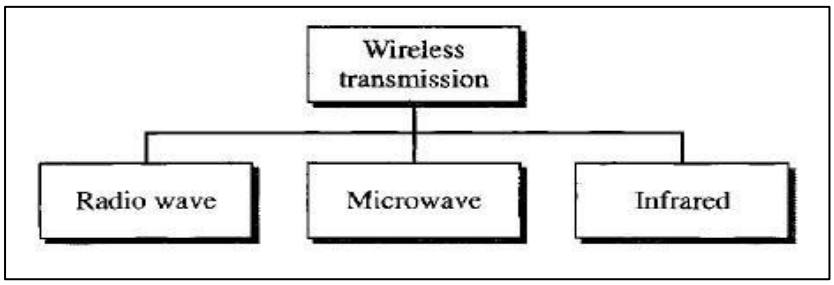
Unguided Media: Wireless

- Unguided media transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as wireless communication. Signals are normally broadcast through free space and thus are available to anyone who has a device capable of receiving them.

- Unguided signals can travel from the source to destination in several ways: **ground propagation**, **sky propagation**, **and line-of-sight propagation,** as shown in Figure.



- In **ground propagation**, radio waves travel through the lowest portion of the atmosphere, hugging the earth.
- In **sky propagation**, higher-frequency radio waves radiate upward into the ionosphere (the layer of atmosphere where particles exist as ions) where they are reflected back to earth.
- In **line-or-sight propagation**, very high-frequency signals are transmitted in straight lines directly from antenna to antenna.
- We can divide wireless transmission into three broad groups: radio waves, microwaves, and infrared waves.



Radio Waves

- Although there is no clear-cut demarcation between radio waves and microwaves, electromagnetic waves ranging in frequencies between 3 kHz and 1 GHz are normally called **radio waves**; waves ranging in frequencies between 1 and 300 GHz are called **microwaves**.
- Radio waves, for the most part, are omni-directional. When an antenna transmits radio waves, they are propagated in all directions. This means that the sending and receiving antennas do not have to be aligned.

- The omni-directional property has a disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band.
- Radio waves, particularly those waves that propagate in the sky mode, can travel long distances.

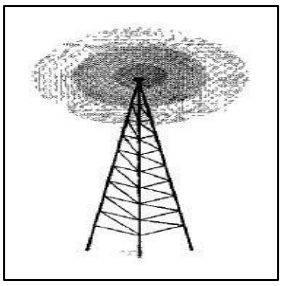
This makes radio waves a good candidate for long-distance broadcasting such as AM radio.

Omni directional Antenna

Radio waves use omni directional antennas that send out signals in all directions. Based on the wavelength, strength, and the purpose of transmission, we can have several types of antennas.

Applications

The omni directional characteristics of radio waves make them useful for multicasting, in which there is one sender but many receivers. AM and FM radio, television, maritime radio, cordless phones, and paging are examples of multicasting.



Microwaves

- Electromagnetic waves having frequencies between I and 300 GHz are called microwaves.
- Microwaves are unidirectional.
- When an antenna transmits microwave waves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas.

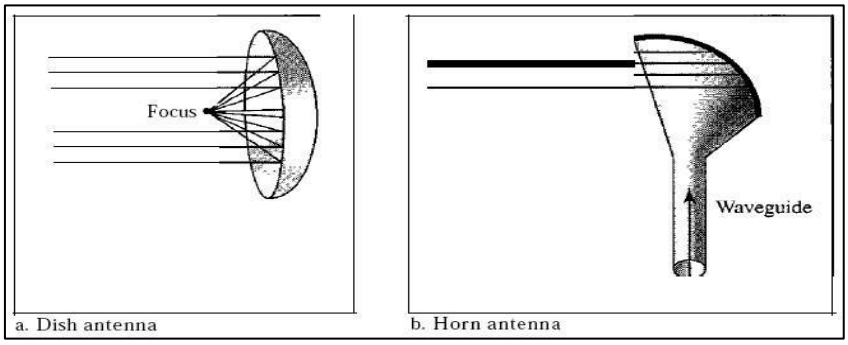
The following describes some characteristics of microwave propagation:

- 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. The curvatures of the earth as well as other blocking obstacles do not allow two short towers to communicate by using microwaves. Repeaters are often needed for long distance communication.

- Very high-frequency microwaves cannot penetrate walls. This characteristic can be a disadvantage if receivers are inside buildings.
- The microwave band is relatively wide, almost 299 GHz. Therefore wider sub bands can be assigned, and a high data rate is possible.
- Use of certain portions of the band requires permission from authorities.

Unidirectional Antenna

Microwaves need unidirectional antennas that send out signals in one direction. Two types of antennas are used for microwave communications: the parabolic dish and the home.



Applications

- Microwaves, due to their unidirectional properties, are very useful when unicast (one-to-one) communication is needed between the sender and the receiver.
- They are used in cellular phones, satellite networks and wireless LANs.

Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nm), can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls.

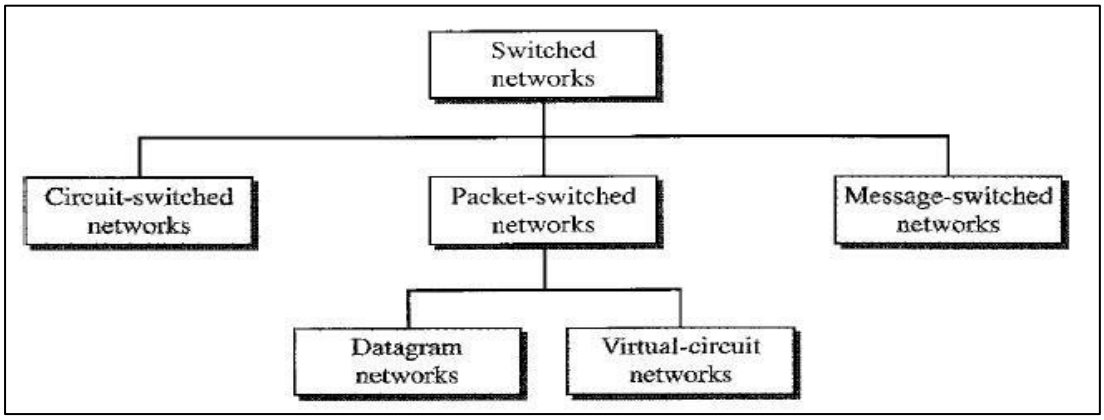
Applications

The infrared band, almost 400 THz, has an excellent potential for data transmission. Such a wide bandwidth can be used to transmit digital data with a very high data rate.

CHAPTER 7

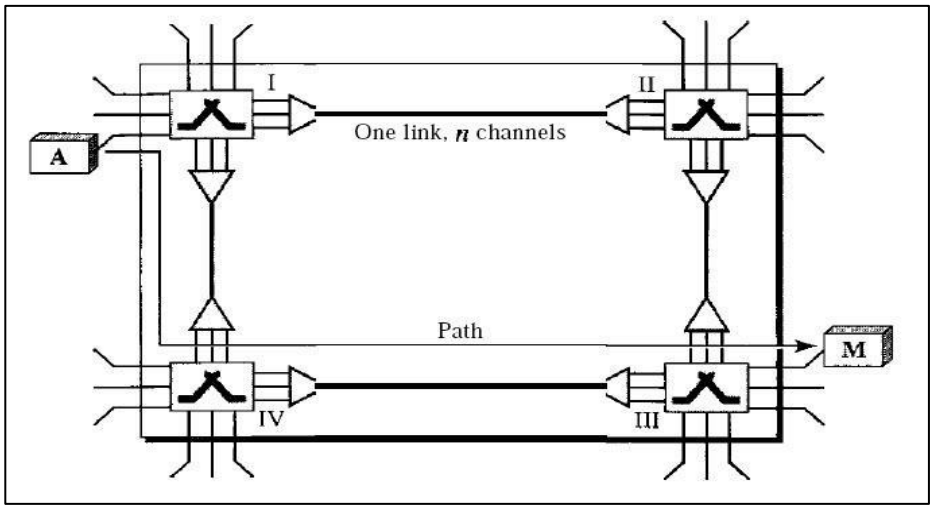
SWITCHING

A switched network consists of a series of interlinked nodes, called switches. Switches are devices capable of creating temporary connections between two or more devices linked to the switch. In a switched network, some of these nodes are connected to the end systems. Traditionally, three methods of switching have been important: **circuit switching**, **packet switching**, and **message switching**. The first two are commonly used today. The third has been phased out in general communications but still has networking applications. We can then divide today's networks into three broad categories: circuit-switched networks, packet-switched networks, and message-switched. Packet switched networks can further be divided into two subcategories-virtual-circuit networks and datagram networks as shown in Figure.



Circuit switching and Telephone Network

- 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.
- However, each connection uses only one dedicated channel on each link. Each link is normally divided into n channels by using FDM or TDM.



- Circuit switching takes place at the physical layer.
- Before starting communication, the stations must make a reservation for the resources to be used during the communication. These resources, such as channels (bandwidth in FDM and time slots in TDM), switch buffers, switch processing time, and switch input/output ports, must remain dedicated during the entire duration of data transfer until the teardown phase.
- Data transferred between the two stations are not packetized (physical layer transfer of the signal). The data are a continuous flow sent by the source station and received by the destination station, although there may be periods of silence.
- There is no addressing involved during data transfer. The switches route the data based on their occupied band (FDM) or time slot (TDM). Of course, there is end-to-end addressing used during the setup phase.

Three Phases

The actual communication in a circuit-switched network requires three phases: connection setup, data transfer, and connection teardown.

Setup Phase

Before the two parties (or multiple parties in a conference call) can communicate, a dedicated circuit (combination of channels in links) needs to be established. The end systems are normally connected through dedicated lines to the switches, so connection setup means creating dedicated channels between the switches.

Data Transfer Phase

After the establishment of the dedicated circuit (channels), the two parties can transfer data.

Teardown Phase

When one of the parties needs to disconnect, a signal is sent to each switch to release the resources.

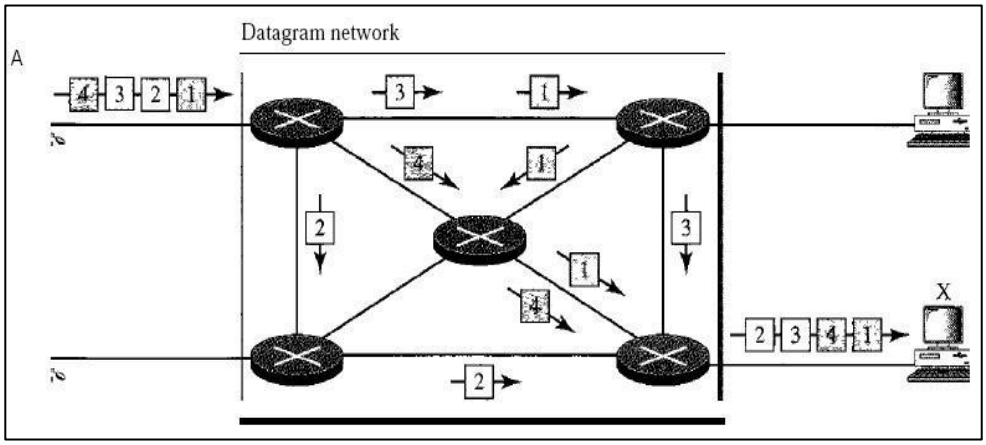
Circuit-Switched Technology in Telephone Networks

The telephone companies have previously chosen the circuit switched approach to switching in the physical layer; today the tendency is moving toward other switching techniques. For example, the telephone number is used as the global address, and a signaling system (called SS7) is used for the setup and teardown phases.

DATAGRAM NETWORKS

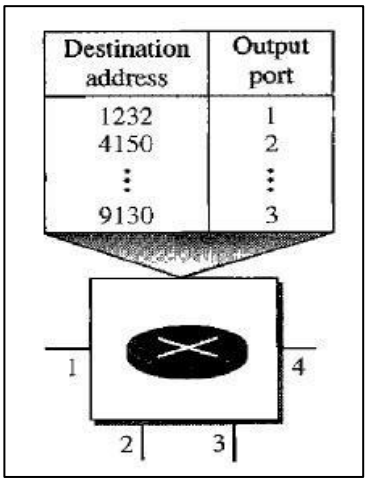
- In a datagram network, each packet is treated independently of all others. Even if a packet is part of a multipacket transmission, the network treats it as though it existed alone. Packets in this approach are referred to as datagrams. Datagram switching is normally done at the network layer.

- The datagram networks are sometimes referred to as connectionless networks. The term connectionless here means that the switch (packet switch) does not keep information about the connection state. There are no setup or teardown phases. Each packet is treated the same by a switch regardless of its source or destination.



Routing Table

If there are no setup or teardown phases, how are the packets routed to their destinations in a datagram network? In this type of network, each switch (or packet switch) has a routing table which is based on the destination address. The routing tables are dynamic and are updated periodically. The destination addresses and the corresponding forwarding output ports are recorded in the tables. This is different from the table of a circuit switched network in which each entry is created when the setup phase is completed and deleted when the teardown phase is over.



[Routing Table]

Destination Address

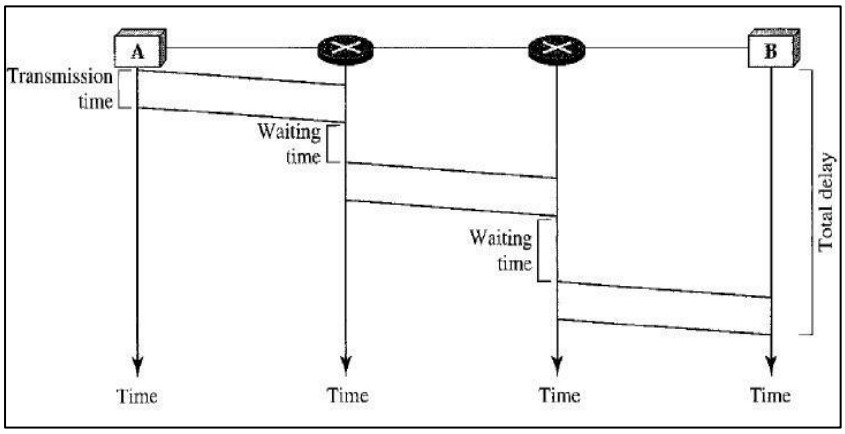
Every packet in a datagram network carries a header that contains, among other information, the destination address of the packet. When the switch receives the packet, this destination address is examined; the routing table is consulted to find the corresponding port through which the packet should be forwarded. This address, unlike the address in a virtual-circuit-switched network, remains the same during the entire journey of the packet.

Efficiency

The efficiency of a datagram network is better than that of a circuit-switched network; resources are allocated only when there are packets to be transferred. If a source sends a packet and there is a delay of a few minutes before another packet can be sent, the resources can be reallocated during these minutes for other packets from other sources.

Delay

There may be greater delay in a datagram network than in a virtual-circuit network. Although there are no setup and teardown phases, each packet may experience a wait at a switch before it is forwarded.



Packet Switching

Packet switching is a digital networking communications method that groups all transmitted data – regardless of content, type, or structure – into suitably sized blocks, called packets. Packet switching features delivery of variable-bit-rate data streams (sequences of packets) over a shared network. When traversing network adapters, switches, routers and other network nodes, packets are buffered and queued, resulting in variable delay and throughput depending on the traffic load in the network.

Packet switching contrasts with another principal networking paradigm, circuit switching, a method which sets up a limited number of dedicated connections of constant bit rate and constant delay between nodes for exclusive use during the communication session. In case

of traffic fees (as opposed to flat rate), for example in cellular communication services, circuit switching is characterized by a fee per time unit of connection time, even when no data is transferred, while packet switching is characterized by a fee per unit of information.

Two major packet switching modes exist:

(1) Connectionless packet switching, also known as datagram switching, and

(2) Connection-oriented packet switching, also known as virtual circuit switching.

In the first case each packet includes complete addressing or routing information. The packets are routed individually, sometimes resulting in different paths and out-of-order delivery.

In the second case a connection is defined and preallocated in each involved node during a connection phase before any packet is transferred.

CHAPTER 8

ERROR DETECTION AND CORRECTION

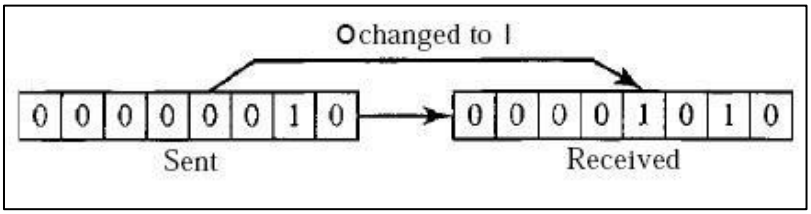
Data can be corrupted during transmission. Some applications require that errors be detected and corrected.

Types of Errors

Whenever bits flow from one point to another, they are subject to unpredictable changes because of interference. This interference can change the shape of the signal. Errors are of two types:

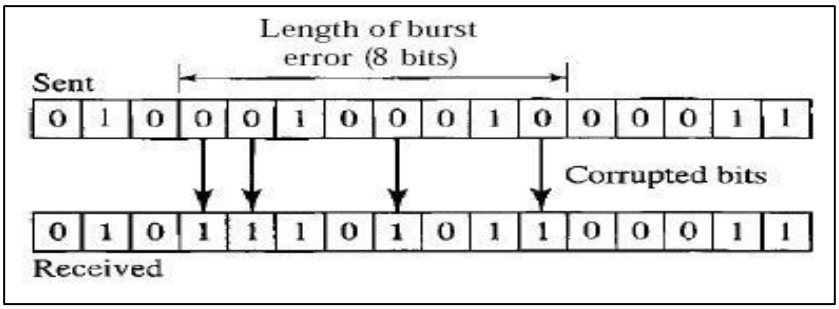
Single-Bit Error

- The term single-bit error means that only 1 bit of a given data unit (such as a byte, character, or packet) is changed from 1 to 0 or from 0 to 1.
- For a single-bit error to occur, the noise must have a duration of only 1) ls, which is very rare; noise normally lasts much longer than this.



Burst Error

- The term burst error means that 2 or more bits in the data unit have changed from 1 to 0 or from 0 to 1.
- A burst error is more likely to occur than a single-bit error. The duration of noise is normally longer than the duration of 1 bit, which means that when noise affects data, it affects a set of bits. The number of bits affected depends on the data rate and duration of noise.



Redundancy

- The central concept in detecting or correcting errors is redundancy.
- To be able to detect or correct errors, we need to send some extra bits with our data.

- These redundant bits are added by the sender and removed by the receiver. Their presence allows the receiver to detect or correct corrupted bits.

Detection versus Correction

- The correction of errors is more difficult than the detection. In error **detection**, we are looking only to see if any error has occurred. The answer is a simple yes or no.
- In error **correction,** we need to know the exact number of bits that are corrupted and more importantly, their location in the message. The number of the errors and the size of the message are important factors.

Forward Error Correction versus Retransmission

There are two main methods of error correction.

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

Coding

Redundancy is achieved through various coding schemes. The sender adds redundant bits through a process that creates a relationship between the redundant bits and the actual data bits. The receiver checks the relationships between the two sets of bits to detect or correct the errors.

- In modular arithmetic, we use only a limited range of integers. We define an upper limit, called a modulus N. We then use only the integers 0 to N -1, inclusive. This is modulo-N arithmetic.
- For example, if the modulus is 12, we use only the integers 0 to 11, inclusive.
- Of particular interest is modulo-2 arithmetic. In this arithmetic, the modulus N is 2. We can use only 0 and 1.

Addition	**Subtraction**
0+0=0	0-0=0
1+0=1	1-0=1
1+1=0	1-1=0
0+1=1	0-1=1

Particularly that addition and subtraction give the same results. In this arithmetic we use the XOR (exclusive OR) operation for both addition and subtraction. The result

of an XOR operation is 0 if two bits are the same; the result is 1 if two bits are different.

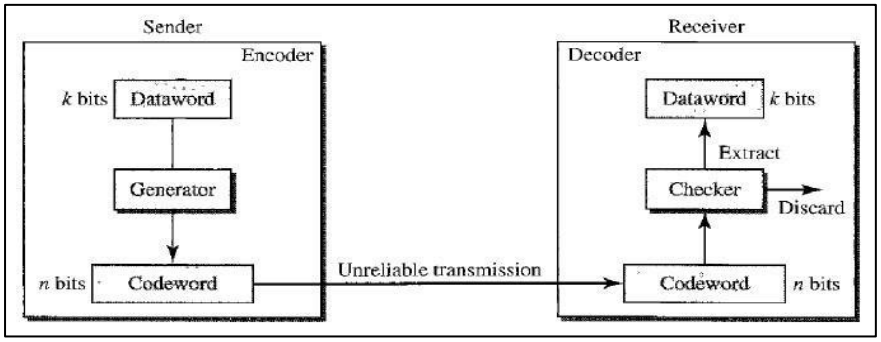
- If the modulus is not 2, addition and subtraction are distinct.

BLOCK CODING

- In block coding, we divide our message into blocks, each of k bits, called data words. We add r redundant bits to each block to make the length n = k + r. The resulting n-bit blocks are called code words.
- With k bits, we can create a combination of 2k data words; with n bits, we can create a combination of 2n code words. Since n > k, the number of possible code words is larger than the number of possible data words. The block coding process is one-to-one; the same data word is always encoded as the same codeword. This means that we have 2n - 2k code words that are not used.

Error Detection

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



[Diagram: Structure of encoder and decoder for Error detection]

Example

Let us assume that k =2 and n =3. Table 1 shows the list of data words and code words. Later, we will see how to derive a codeword from a data word.

TABLE 1

Data words	Code words

00	000
01	011
10	101
11	110

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

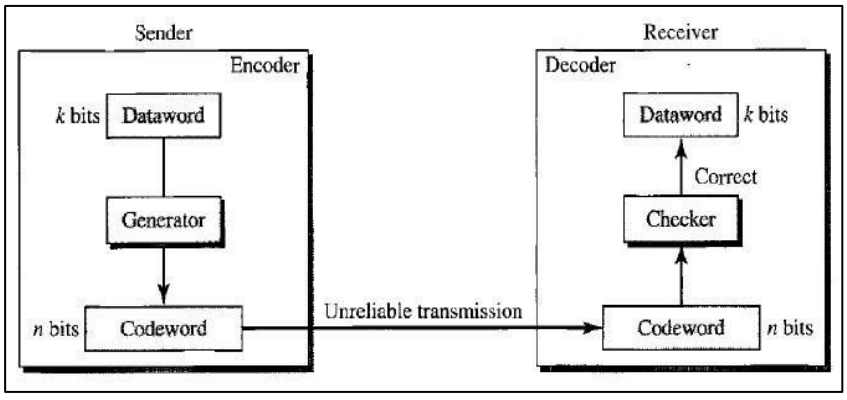
1. The receiver receives 011. It is a valid codeword. The receiver extracts the data word 01 from it.
2. The codeword is corrupted during transmission, and 111 is received (the leftmost bit is corrupted).

This is not a valid codeword and is discarded.

3. The codeword is corrupted during transmission, and 000 is received (the right two bits are corrupted). This is a valid codeword. The receiver incorrectly extracts the data word 00. Two corrupted bits have made the error undetectable.

Error Correction

As we said before, error correction is much more difficult than error detection. In error detection, the receiver needs to know only that the received codeword is invalid; in error correction the receiver needs to find (or guess) the original codeword sent.



[Diagram: Structure of encoder and decoder for Error correction]

Example

Let us assume that k =2 and n =3. Below table shows the list of data words and code words.

Dataword(k)	**Codeword(n=k+r)**
00	00000

01 01011

10

10101 11

11110

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

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

Hamming Distance

- The Hamming distance between two words (of the same size) is the number of differences between the corresponding bits.
- The Hamming distance can easily be found if we apply the *XOR operation on the two words and count the number of 1s in the result*. Note that the Hamming distance is a value greater than zero.
- The minimum Hamming distance is the smallest Hamming distance between all possible pairs in a set of words.

Example:

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

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

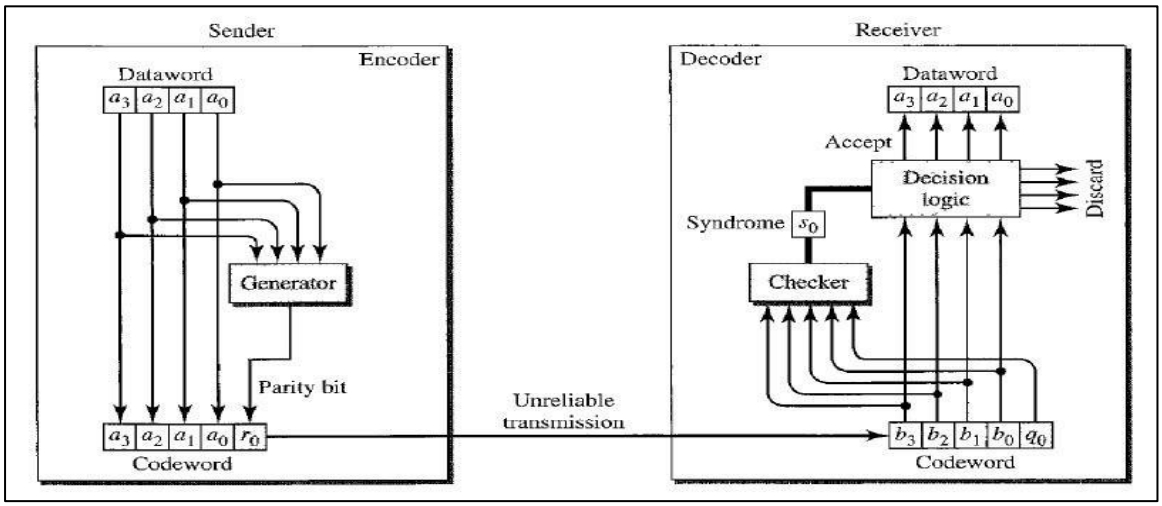
Linear Block Codes

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

- The scheme in Table 1 is a linear block code because the result of XORing any codeword with any other codeword is a valid codeword. For example, the XORing of the second and third code words creates the fourth one.
- Let us now show some linear block codes. These codes are trivial because we can easily find the encoding and decoding algorithms and check their performances.

Simple Parity-Check Code

- A simple parity-check code is a single-bit error-detecting code in which n =k + 1 with d_{min} =2.
- In this code, a k-bit data word is changed to an n-bit codeword where n = k + 1. The extra bit, called the parity bit, is selected to make the total number of 1s in the codeword even.
- The minimum Hamming distance for this category is d_{min} =2, which means that the code is a single-bit error-detecting code; it cannot correct any error.
- **Table 1** is a parity-check code with k = 2 and n =3.



[Encoder and Decoder for Simple parity check]

- The encoder uses a generator that takes a copy of a 4-bit dataword (a0, a1, a2 and a3) and generates a parity bit ro. The dataword bits and the parity bit create the 5-bit codeword. *The parity bit that is added makes the number of 1s in the codeword even*.
- This is normally done by adding the 4 bits of the dataword (modulo-2); the result is the parity bit. In other words,
  - $\mathbf{r_0 = a_3 + a_2 + a_1 + a_0}$ (modulo 2)
- If the number of 1s is even, the result is 0; if the number of 1s is odd, the result is 1. In both cases, the total number of 1s in the codeword is even.
- The sender sends the codeword which may be corrupted during transmission. The receiver receives a 5-bit word. The checker at the receiver does the same thing as the generator in the sender with one exception: The addition is done over all 5 bits. The

result, which is called the syndrome, is just 1 bit. *The syndrome is 0 when the number of 1s in the received codeword is even; otherwise, it is 1*.

- $s_0 = b_3 + b_2 + b_1 + b_0$ (modulo 2)

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

Example:

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

We examine five cases:

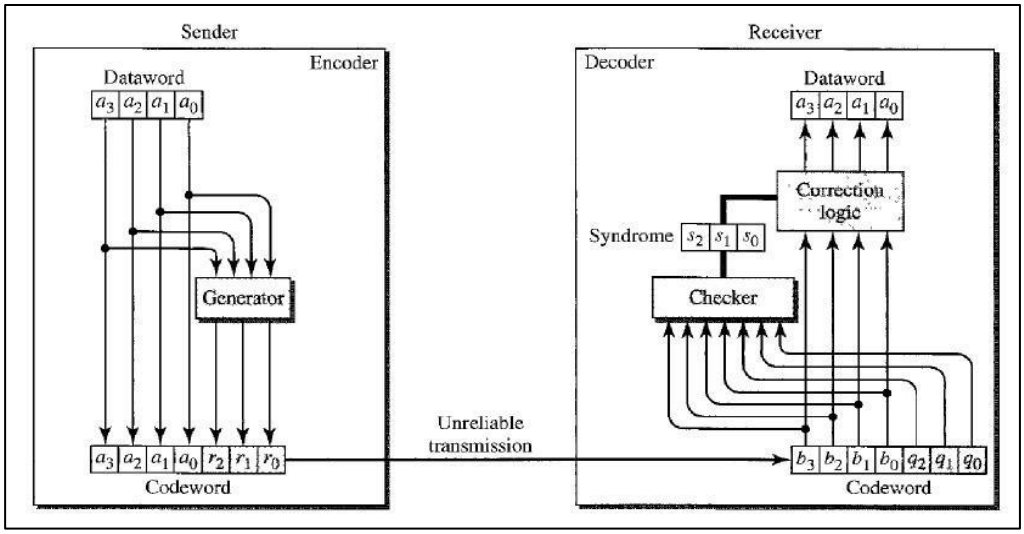
1. No error occurs; the received codeword is 10111. The syndrome is 0. The dataword 1011 is created.
2. One single-bit error changes a_1. The received codeword is 10011. The syndrome is 1. No dataword is created.
3. One single-bit error changes r_o.The received codeword is 10110. The syndrome is 1. No dataword is created. 4. An error changes r_o and a second error changes a_3. The received codeword is 00110. The syndrome is 0. The dataword 0011 is created at the receiver. Note that here the dataword is wrongly created due to the syndrome value. *The simple parity-check decoder cannot detect an even number of errors. The errors cancel each other out and give the syndrome a value of 0.*

5. Three bits-a_3, a_2, and a_1-are changed by errors. The received codeword is 01011. The syndrome is 1. The dataword is not created. *This shows that the simple parity check, guaranteed to detect one single error, can also find any odd number of errors.*

Hamming Code

- These codes were originally designed with $d_{min} = 3$, which means that they can detect up to two errors or correct one single error. Although there are some Hamming codes that can correct more than one error, our discussion focuses on the single-bit error-correcting code.
- Let us find the relationship between n and k in a Hamming code. We need to choose an integer m >= 3.

  The values of n and k are then calculated from m as $n = 2^m - 1$and k=n - m. The number of check bits r =m.
- A Hamming code can only correct a single error or detect a double error.



[Encoder and Decoder for Hamming Code]

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

$r_0=a_2+a_1+a_o$

modulo-2 $r_1 = a_3 + a_2 + a_1$

modulo-2

$r_2=a_1+a_0+a_3$

modulo-2

In other words, each of the parity-check bits handles 3 out of the 4 bits of the dataword. The total number of 1s in each 4-bit combination (3 dataword bits and 1 parity bit) must be even. The checker in the **decoder** creates a 3-bit syndrome (s2s1s0) in which each bit is the parity check for 4 out of the 7 bits in the received codeword:

$s_0=b_2+b_1+b_o+q_0$ modulo-2

$s_1 = b_3 + b_2 + b_1 + q_1$

modulo-2

$s_2=b_1+b_0+b_3+q_2$ modulo-2

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

Syndrome	**000**	**001**	**010**	**011**	**100**	**101**	**110**	**111**
Error	**None**	**q_0**	**q_1**	**b2**	**q2**	**b_0**	**b_3**	**b1**

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

1. The dataword 0100 becomes the codeword 0100011. The codeword 01 00011 is received. The syndrome is 000 (no error), the final dataword is 0100.
2. The dataword 0111 becomes the codeword 0111001. The codeword 0011001 is received. The syndrome is 011. According to Table, b2 is in error. After flipping b2 (changing the 1 to 0), the final dataword is 0111.
3. The dataword 1101 becomes the codeword 1101000. The codeword 0001000 is received (two errors). The syndrome is 101, which means that b_0 is in error. After flipping b_0, we get 0000, the wrong dataword. This shows that our code cannot correct two errors.

Cyclic Codes

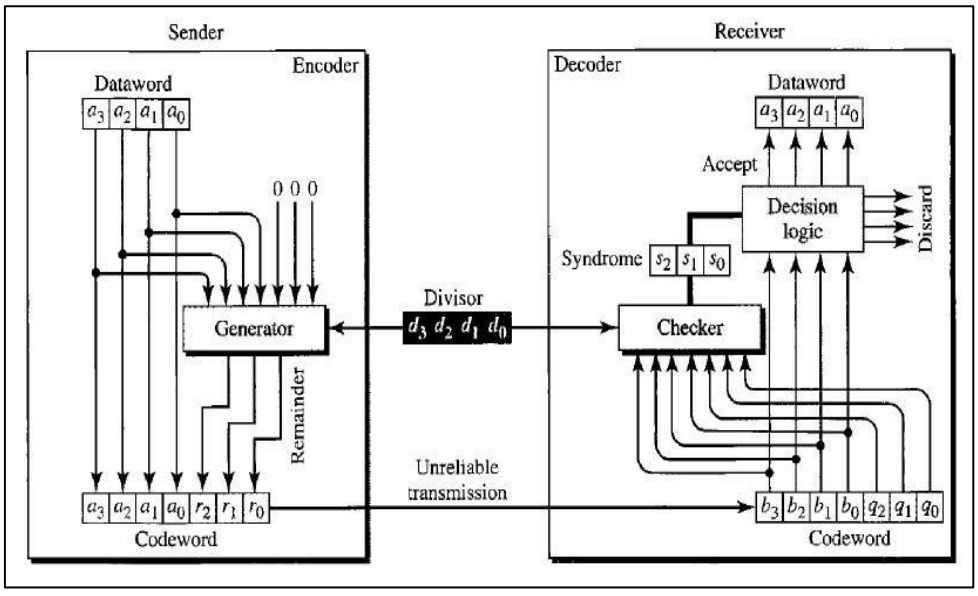
Cyclic codes are special linear block codes with one extra property. In a cyclic code, if a codeword is cyclically shifted (rotated), the result is another codeword. For example, if 1011000 is a codeword and we cyclically left-shift, then 0110001 is also a codeword.

If we call the bits in the first word a_0 to a_6 and the bits in the second word b_0 to b_6, we can shift the bits by using the following:

b1=a0 b2=a1 b3=a2 b4=a3 b5=a4 b6=a5 b0=a6

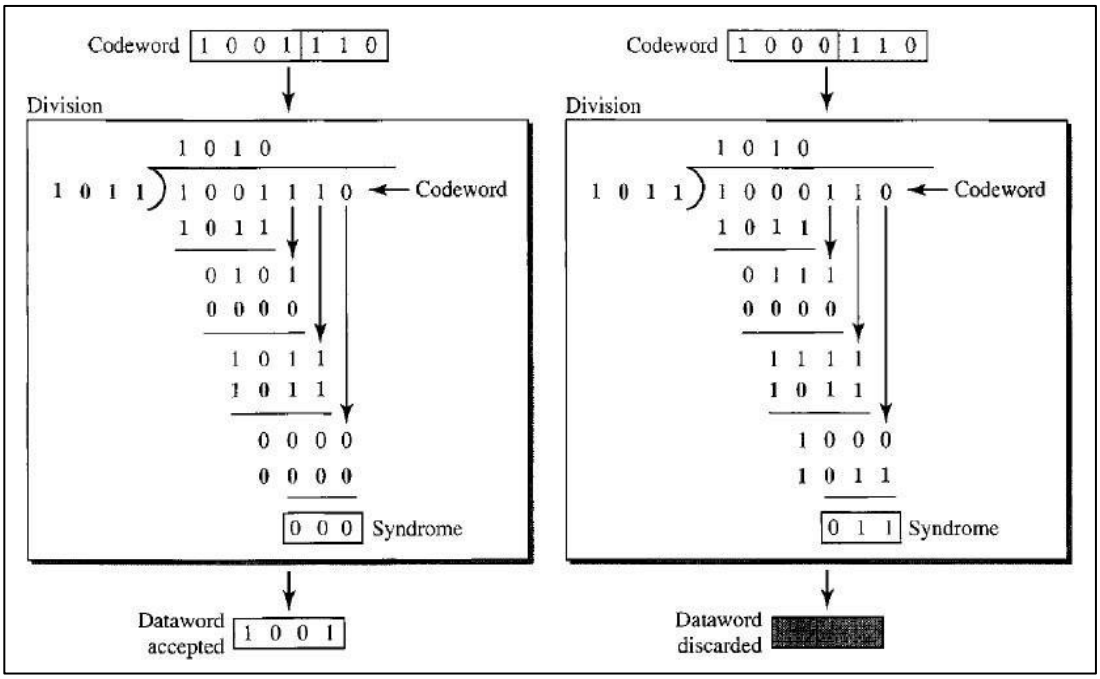
Cyclic Redundancy Check

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



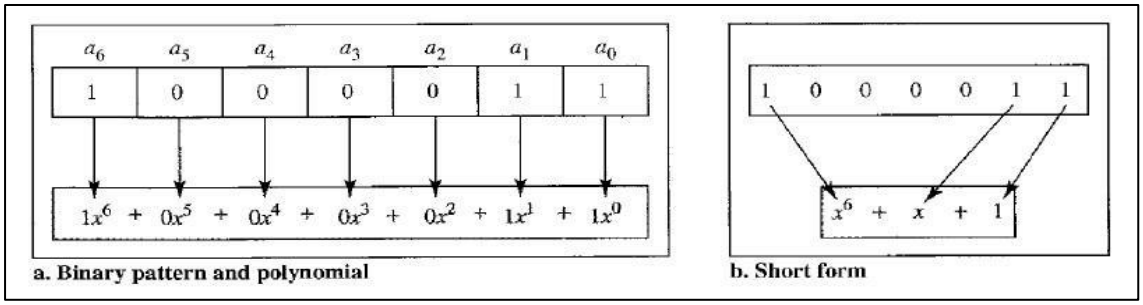
[Encoder and Decoder for CRC]

- In the encoder, the dataword has k bits (4 here); the codeword has n bits (7 here). The size of the dataword is augmented by adding n - k (3 here) 0s to the right-hand side of the word. The nbit result is fed into the generator. The generator uses a divisor of size n - k + 1 (4 here), predefined and agreed upon. The generator divides the augmented dataword by the divisor (modulo-2 division). The quotient of the division is discarded; the remainder ($r_2r_1r_0$) is appended to the dataword to create the codeword.
- The decoder receives the possibly corrupted codeword. A copy of all n bits is fed to the checker which is a replica of the generator. The remainder produced by the checker is a syndrome of n - k (3 here) bits, which is fed to the decision logic analyzer. The analyzer has a simple function. If the syndrome bits are all as, the 4 leftmost bits of the codeword are accepted as the dataword (interpreted as no error); otherwise, the 4 bits are discarded (error).



Polynomial

A pattern of 0s and 1s can be represented as a **polynomial** with coefficients of 0 and 1. The power of each term shows the position of the bit; the coefficient shows the value of the bit. Below figure shows a binary pattern and its polynomial representation.


a_6 a_5 a_4 a_3 a_2 a_1 a_0
1 0 0 0 0 1 1
$1x^6 + 0x^5 + 0x^4 + 0x^3 + 0x^2 + 1x^1 + 1x^0$
a. Binary pattern and polynomial
1 0 0 0 0 1 1
$x^6 + x + 1$
b. Short form

Chapter 9

DATA LINK CONTROL

- The two main functions of the data link layer are **data link control** and **media access control**. The first, data link control, deals with the design and procedures for communication between two adjacent nodes: node-to-node communication.
- The second function of the data link layer is media access control, or how to share the link.
- Data link control functions include **framing**, **flow** and error **control**, and software implemented protocols that provide smooth and reliable transmission of frames between nodes.

Framing

- The data link layer, on the other hand, needs to pack bits into frames, so that each frame is distinguishable from another.
- Framing in the data link layer separates a message from one source to a destination, or from other messages to other destinations, by adding a sender address and a destination address. The destination address defines where the packet is to go; the sender address helps the recipient acknowledge the receipt. Frames can be of fixed or variable size.

1. Fixed-Size Framing

In fixed-size framing, there is no need for defining the boundaries of the frames; the size itself can be used as a delimiter.

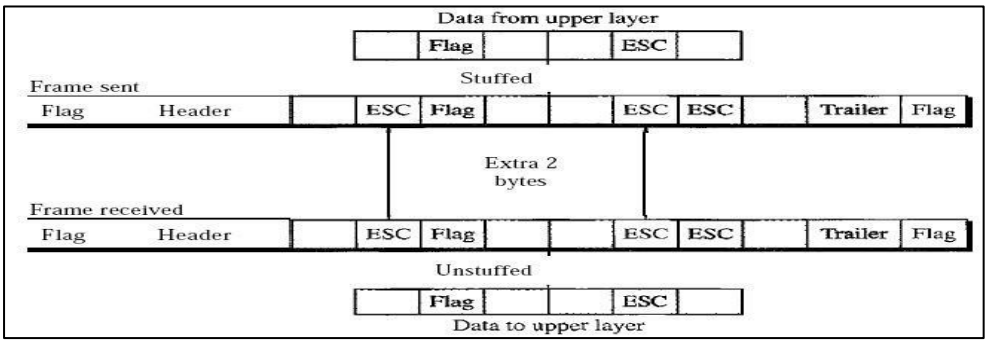
2. Variable-Size Framing

In variable-size framing, we need a way to define the end of the frame and the beginning of the next. Historically, two approaches were used for this purpose: a character-oriented approach and a bit-oriented approach.

Character-Oriented Protocols

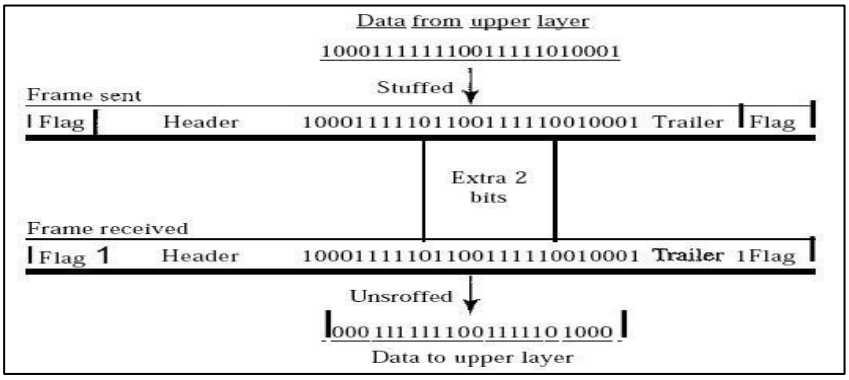
- In a character-oriented protocol, data to be carried are 8-bit characters.
- The header, which normally carries the source and destination addresses and other control information, and the trailer, which carries error detection or error correction redundant bits, are also multiples of 8 bits.
- To separate one frame from the next, an 8-bit (I-byte) flag is added at the beginning and the end of a frame. The flag, composed of protocol-dependent special characters, signals the start or end of a frame.
- Any pattern used for the flag could also be part of the information. If this happens, the receiver, when it encounters this pattern in the middle of the data, thinks it has reached the end of the frame. To fix this problem, a **byte-stuffing** strategy was added to character-oriented framing. In byte stuffing (or character stuffing), a special byte is added to the data section of the frame when there is a character with the same pattern as the flag. The data section is stuffed with an extra byte. This byte is usually

called the escape character (ESC), which has a predefined bit pattern. Whenever the receiver encounters the ESC character, it removes it from the data section and treats the next character as data, not a delimiting flag.



Bit-Oriented Protocols

- In a bit-oriented protocol, the data section of a frame is a sequence of bits to be interpreted by the upper layer as text, graphic, audio, video, and so on.
- Most protocols use a special 8-bit pattern flag 01111110 as the delimiter to define the beginning and the end of the frame.
- **Bit stuffing** is the process of adding one extra 0 whenever five consecutive 1s follow a 0 in the data, so that the receiver does not mistake the pattern 0111110 for a flag.



Flow and Error Control

- Data communication requires at least two devices working together, one to send and the other to receive.
- The most important responsibilities of the data link layer are **flow control** and **error control**.
    - Collectively, these functions are known as **data link control**.

Flow Control

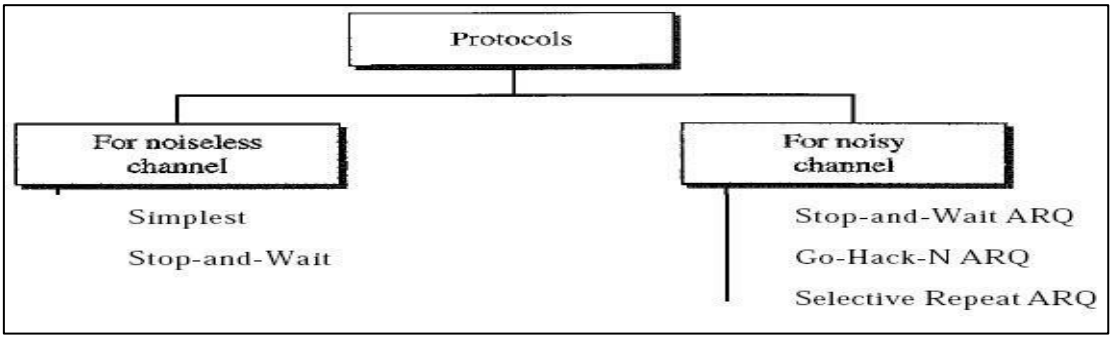
- Flow control refers to a set of procedures used to restrict the amount of data that the sender can send before waiting for acknowledgment.
- Each receiving device has a block of memory, called a buffer, reserved for storing incoming data until they are processed. If the buffer begins to fill up, the receiver must be able to tell the sender to halt transmission until it is once again able to receive.

Error Control

- Error control is both error detection and error correction.
- It allows the receiver to inform the sender of any frames lost or damaged in transmission and coordinates the retransmission of those frames by the sender.
- In the data link layer, the term error control refers primarily to methods of error detection and retransmission. This process is called **Automatic Repeat Request** (ARQ).

Protocols

- Data link layer can combine framing, flow control, and error control to achieve the delivery of data from one node to another.
- We divide the discussion of protocols into those that can be used for noiseless (error-free) channels and those that can be used for noisy (error-creating) channels.
- The protocols in the first category cannot be used in real life, but they serve as a basis for understanding the protocols of noisy channels.



Stop-and- Wait ARQ

- To detect and correct corrupted frames, we need to add redundancy bits to our data frame. When the frame arrives at the receiver site, it is checked and if it is corrupted, it is silently discarded. The detection of errors in this protocol is manifested by the silence of the receiver.
- When the receiver receives a data frame that is out of order, this means that frames were either lost or duplicated. The corrupted and lost frames need to be resent in this protocol. If the receiver does not respond when there is an error, how can the sender know which frame to resend?
- To remedy this problem, the sender keeps a copy of the sent frame. At the same time, it starts a timer. If the timer expires and there is no ACK for the sent frame, the frame is resent, the copy is held, and the timer is restarted.
- Error correction in Stop-and-Wait ARQ is done by keeping a copy of the sent frame and retransmitting of the frame when the timer expires.

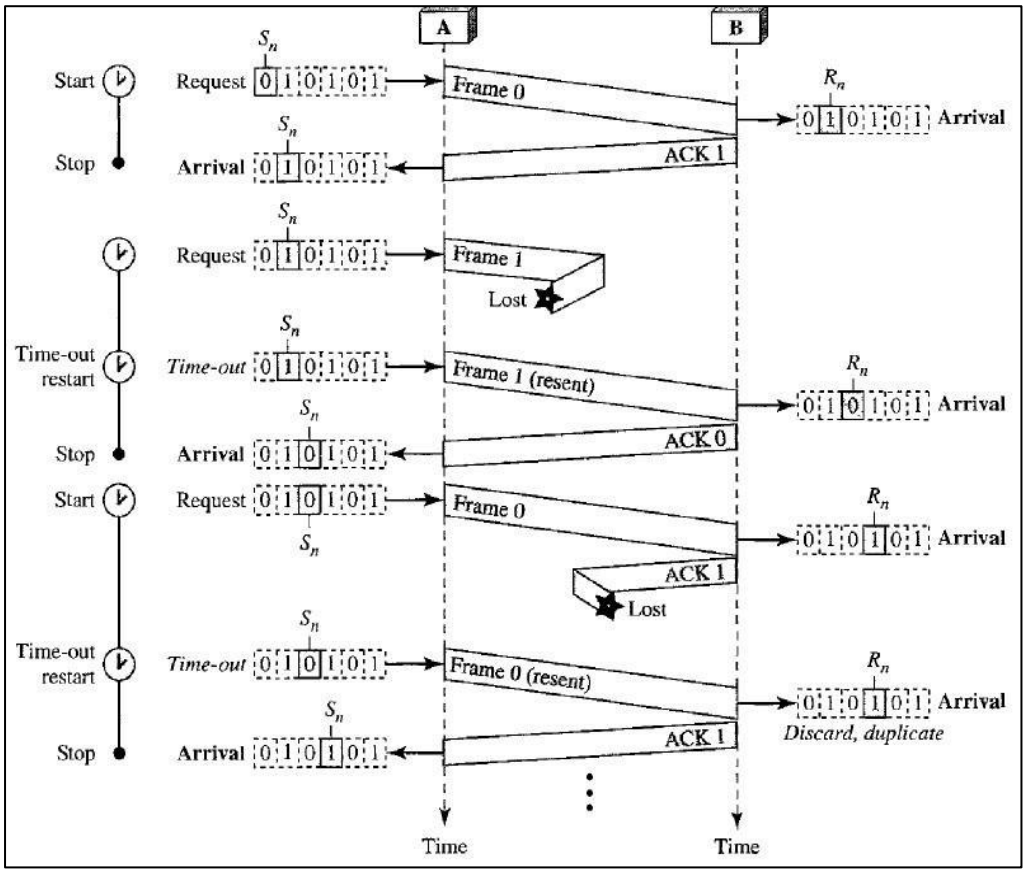
Sequence Numbers

- A frame is numbered by sequence numbers.

- A field is added to the data frame to hold the sequence number of that frame.
- If we decide that the field is m bits long, the sequence numbers start from 0, go to 2^m - 1, and then are repeated.

Let us consider three cases:

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



Go-Back-N ARQ

- To improve the efficiency of transmission (filling the pipe), multiple frames must be in transition while waiting for acknowledgment. In other words, we need to let more

than one frame be outstanding to keep the channel busy while the sender is waiting for acknowledgment.

- The first is called Go-Back-N Automatic Repeat Request. In this protocol we can send several frames before receiving acknowledgments; we keep a copy of these frames until the acknowledgments arrive.
- In Go-Back-N ARQ, the size of the send window must be less than 2^m; the size of the receiver window is always 1.

Sequence Numbers

Frames from a sending station are numbered sequentially. However, because we need to include the sequence number of each frame in the header, we need to set a limit. If the header of the frame allows m bits for the sequence number, the sequence numbers range from 0 to 2^m - 1. For example, if m is 4, the only sequence numbers are 0 through 15 inclusive. However, we can repeat the sequence. So the sequence numbers are
0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12, 13, 14, 15, 0, 1, 2, 3, 4, 5, 6, 7, 8, 9, 10, 11, ...

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

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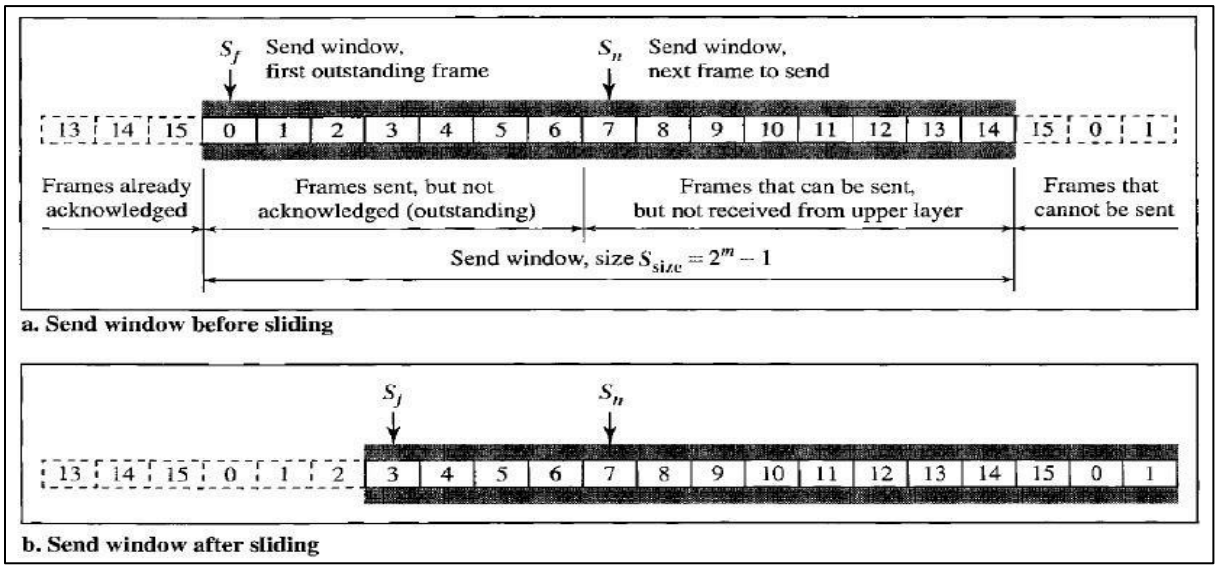
Sliding Window

In Go-Back-N Automatic Repeat Request, *the sliding window is an abstract concept that defines the range of sequence numbers that is the concern of the sender and receiver*. In other words, the sender and receiver need to deal with only part of the possible sequence numbers. The range which is the concern of the sender is called the send sliding window; the range that is the concern of the receiver is called the receive sliding window.
The window at any time divides the possible sequence numbers into four regions.

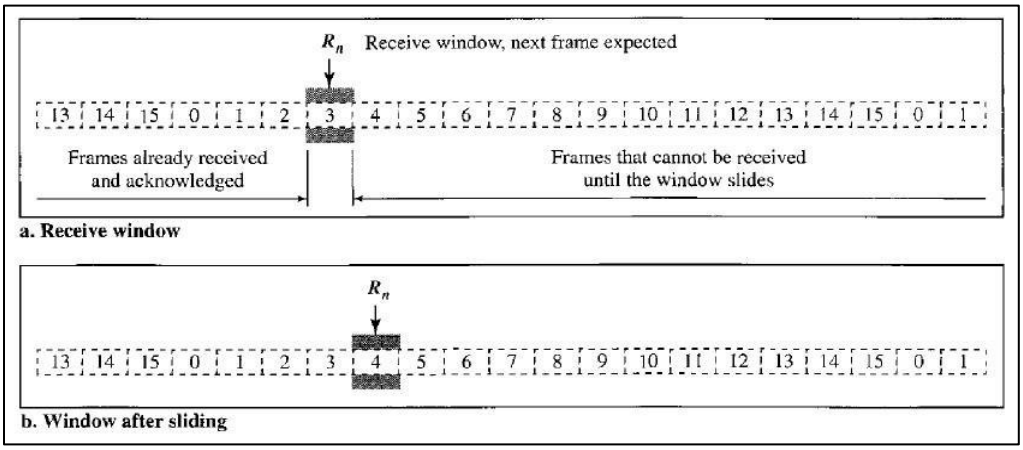
- The first region, from the far left to the left wall of the window, defines the sequence numbers belonging to frames that are already acknowledged. The sender does not worry about these frames and keeps no copies of them.
- The second region defines the range of sequence numbers belonging to the frames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

Send Window for Go Back N ARQ



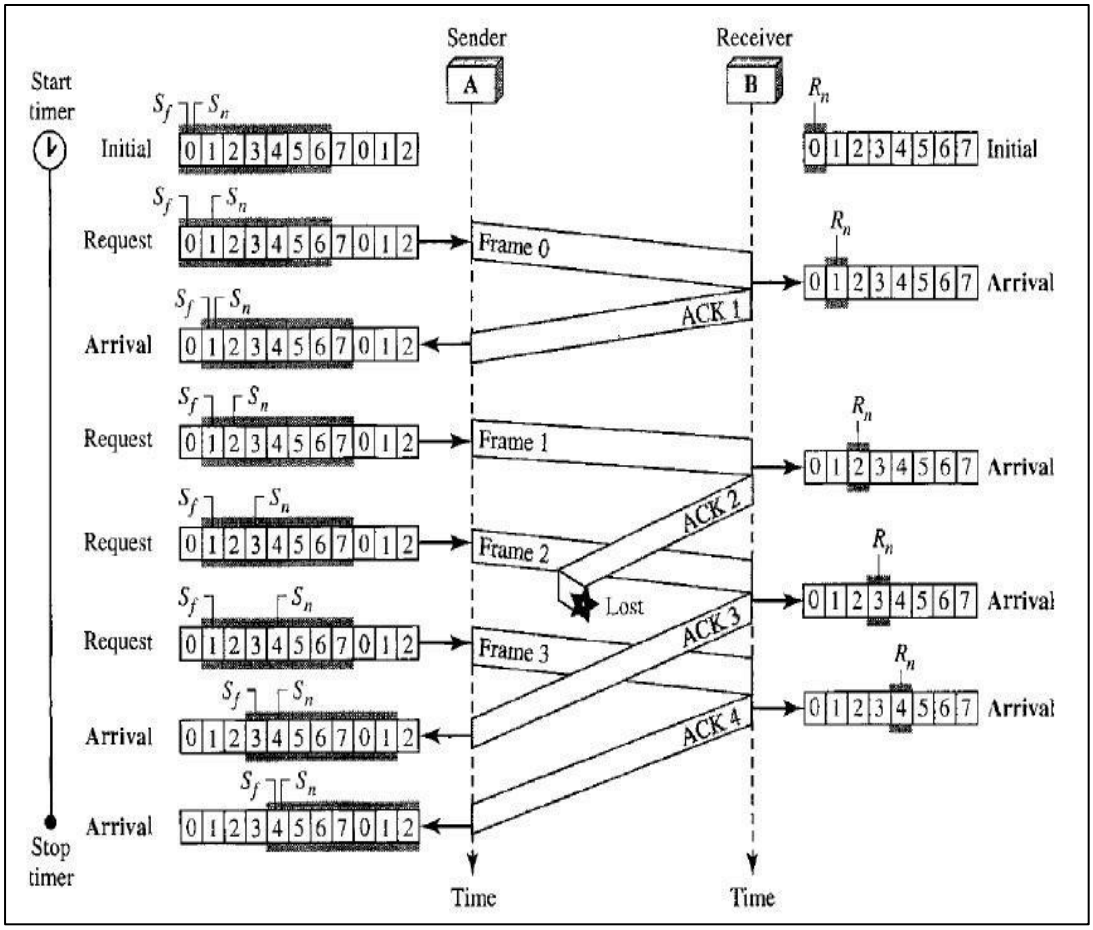
The window itself is an abstraction; three variables define its size and location at any time. We call these variables S_f (send window, the first outstanding frame), S_n (send window, the next frame to be sent), and S_{size} (send window, size).

- The variable S_f defines the sequence number of the first (oldest) outstanding frame.
- The variable S_n holds the sequence number that will be assigned to the next frame to be sent.
- Finally, the variable S_{size} defines the size of the window, which is fixed in our protocol. **Receive Window for Go Back N ARQ**



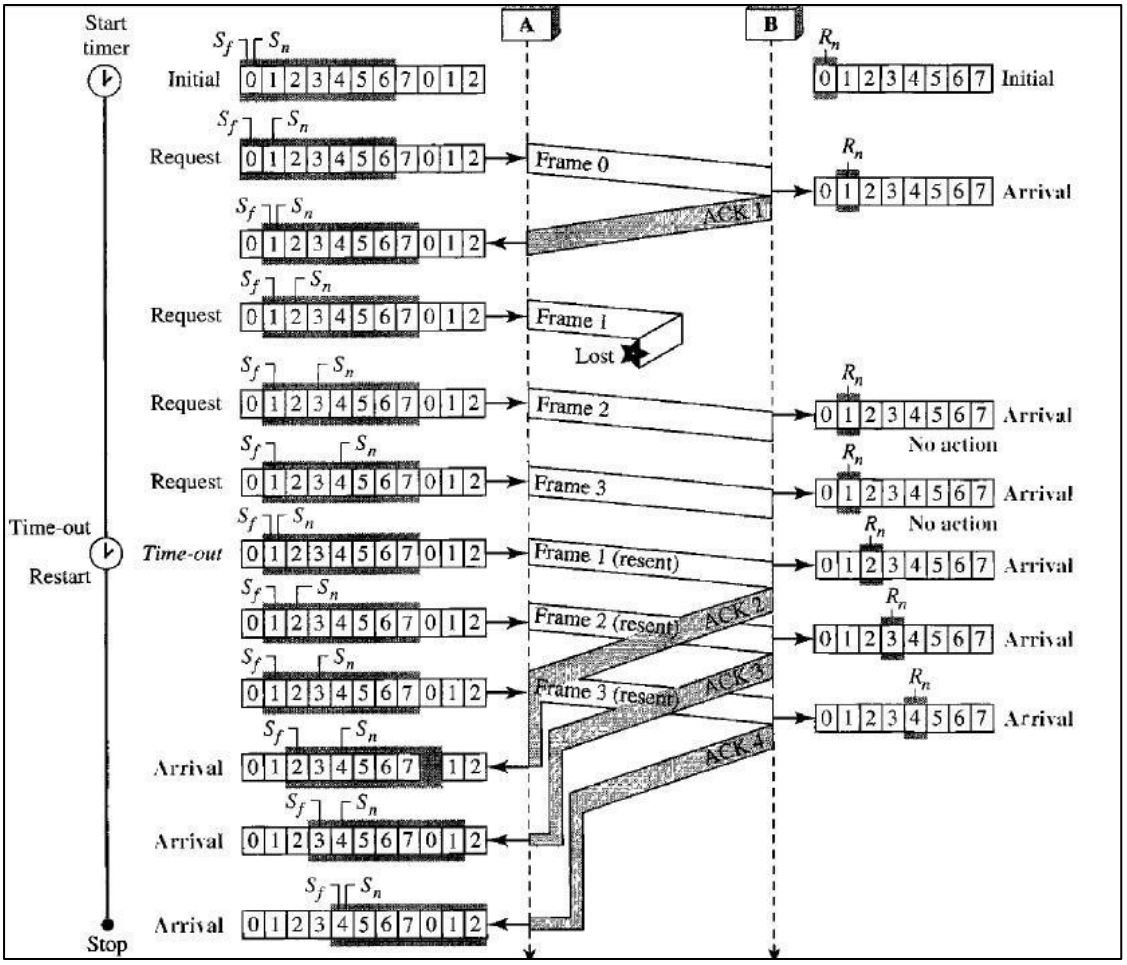
Example:

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



Example:

Below figure shows what happens when a frame is lost. Frames 0, 1, 2, and 3 are sent. However, frame 1 is lost. The receiver receives frames 2 and 3, but they are discarded because they are received out of order (frame 1 is expected). The sender receives no acknowledgment about frames 1, 2, or 3. Its timer finally expires. The sender sends all outstanding frames (1, 2, and 3) because it does not know what is wrong. Note that the resending of frames l, 2, and 3 is the response to one single event. When the sender is responding to this event, it cannot accept the triggering of other events. This means that when ACK 2 arrives, the sender is still busy with sending frame 3. The physica1layer must wait until this event is completed and the data link layer goes back to its sleeping state. We have shown a vertical line to indicate the delay. It is the same story with ACK 3; but when ACK 3 arrives, the sender is busy responding to ACK 2. It happens again when ACK 4 arrives. Note that before the second timer expires, all outstanding frames have been sent and the timer is stopped.



Go-Back-N ARQ versus Stop-and- Wait ARQ

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

Selective Repeat Automatic Repeat Request

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

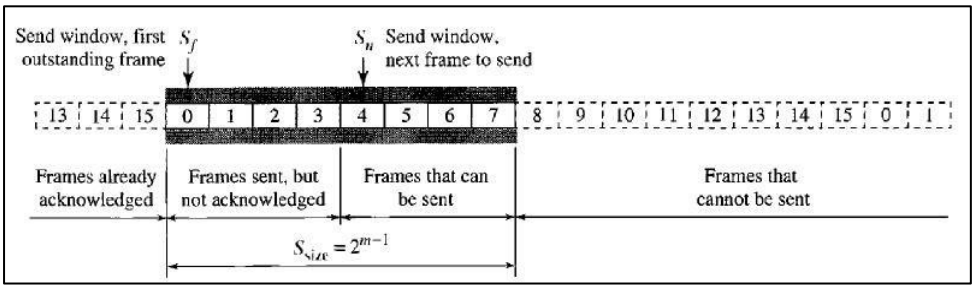
is damaged; only the damaged frame is resent. This mechanism is called Selective Repeat ARQ.

Windows

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

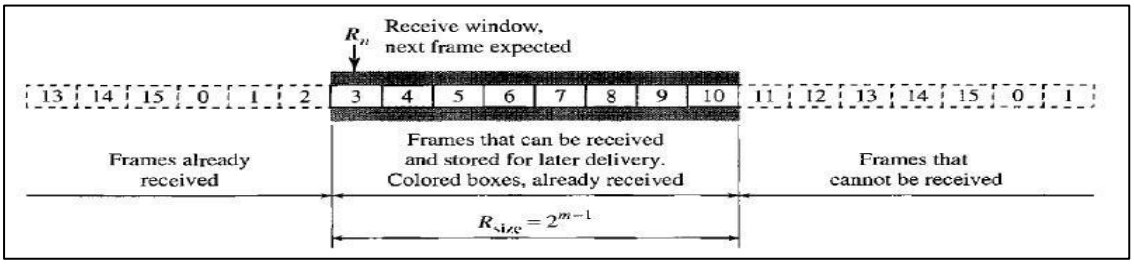
Send window for selective repeat ARQ

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



Receive window for selective repeat ARQ

The receive window in Selective Repeat is totally different from the one in GoBack-N. First, the size of the receive window is the same as the size of the send window (2^m- 1).

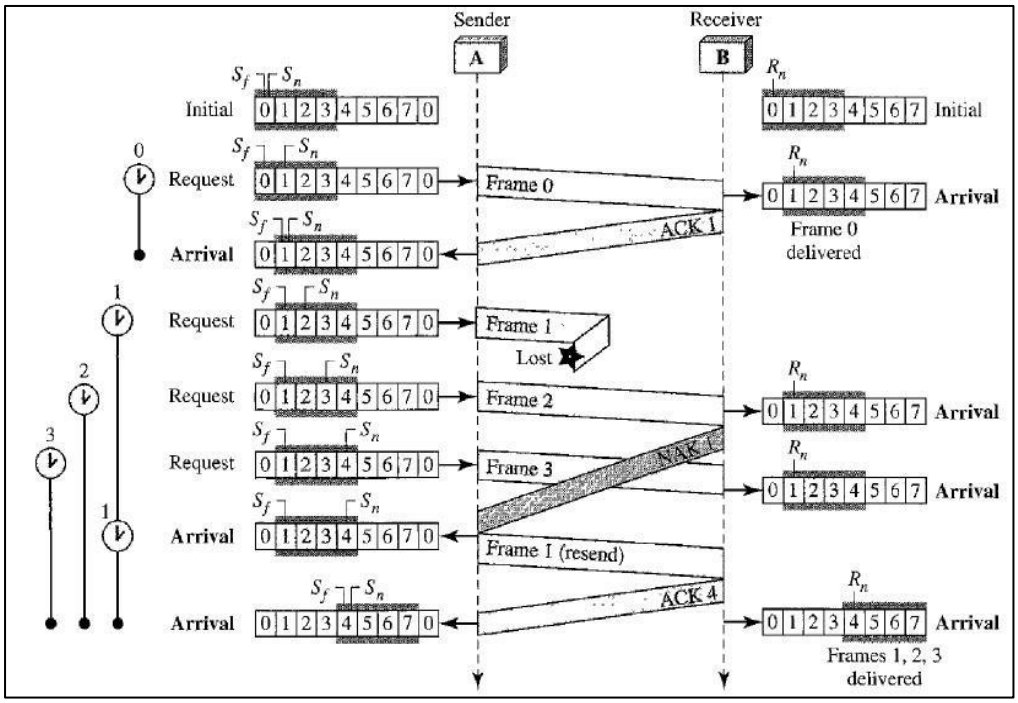


Piggybacking

- Data frames flow in only one direction although control information such as ACK and NAK frames can travel in the other direction. In real life, data frames are normally flowing in both directions: from node A to node B and from node B to node A. This means that the control information also needs to flow in both directions.
- When a frame is carrying data from A to B, it can also carry control information about arrived (or lost) frames from B; when a frame is carrying data from B to A, it can also

carry control information about the arrived (or lost) frames from A. This is called Piggybacking.

- **Piggybacking** is used to improve the efficiency of the bidirectional protocols.



HDLC

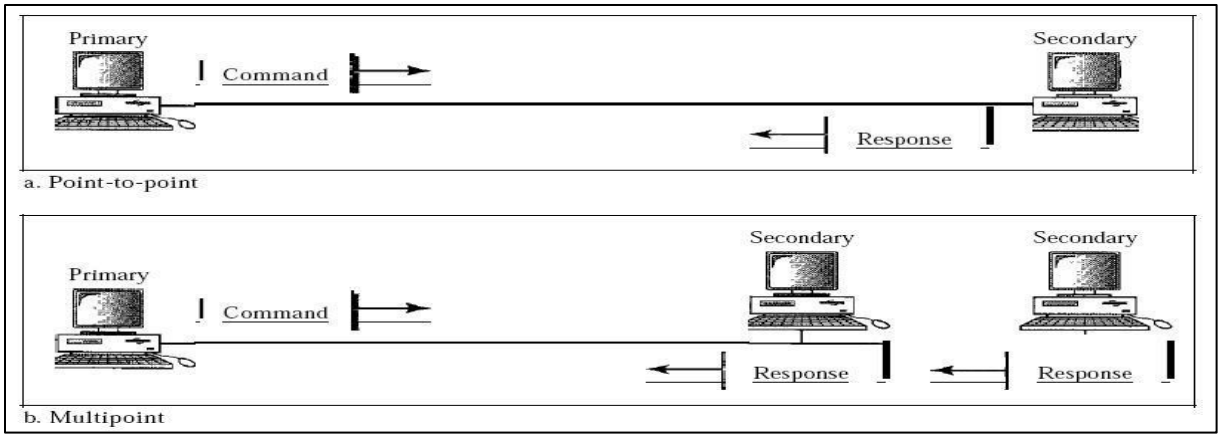
High-level Data Link Control (HDLC) is a bit-oriented protocol for communication over point-to-point and multipoint links.

Configurations and Transfer Modes

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

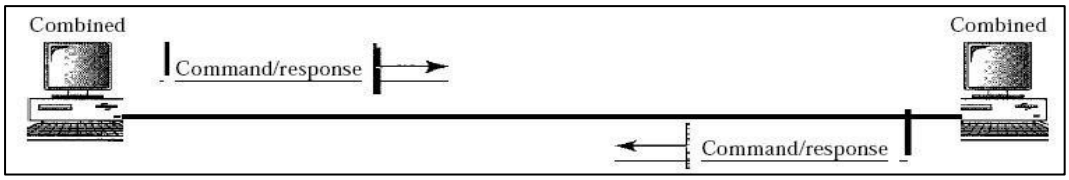
Normal Response Mode

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



Asynchronous Balanced Mode

In asynchronous balanced mode (**ABM**), the configuration is balanced. The link is point-to-point, and each station can function as a primary and a secondary (acting as peers).

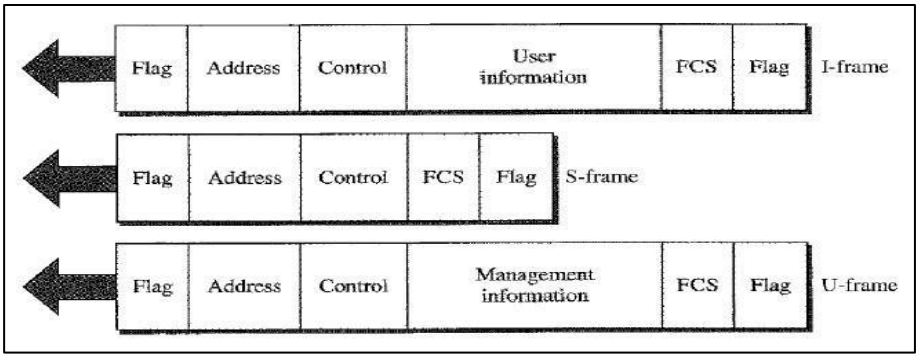


Frames

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

Frame Format

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



Fields

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

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

PPP

Although HDLC is a general protocol that can be used for both point-to-point and multipoint configurations, one of the most common protocols for point-to-point access is the **Point-to-Point Protocol** (PPP). Today, millions of Internet users who need to connect their home computers to the server of an Internet service provider use PPP.

PPP provides several services:

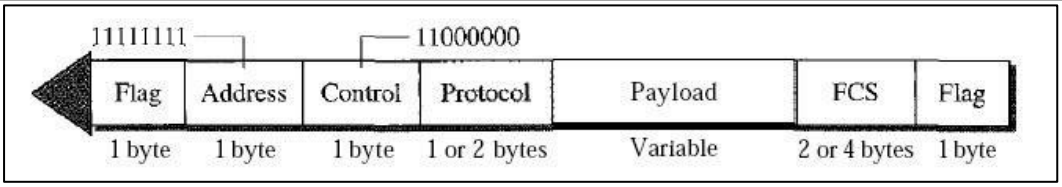
1. PPP defines the format of the frame to be exchanged between devices.
2. PPP defines how two devices can negotiate the establishment of the link and the exchange of data.
3. PPP defines how network layer data are encapsulated in the data link frame.
4. PPP defines how two devices can authenticate each other.

5. PPP provides multiple network layer services supporting a variety of network layer protocols.
6. PPP provides connections over multiple links.
7. PPP provides network address configuration. This is particularly useful when a home user needs a temporary network address to connect to the Internet.

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

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

Frame Format of PPP



Flag: A PPP frame starts and ends with a I-byte flag with the bit pattern 01111110. Although this pattern is the same as that used in HDLC, there is a big difference. PPP is a byte-oriented protocol; HDLC is a bitoriented protocol.

Address: The address field in this protocol is a constant value and set to 11111111 (broadcast address). During negotiation (discussed later), the two parties may agree to omit this byte.

Control: This field is set to the constant value 11000000 (imitating unnumbered frames in HDLC). PPP does not provide any flow control. Error control is also limited to error detection. This means that this field is not needed at all, and again, the two parties can agree, during negotiation, to omit this byte.

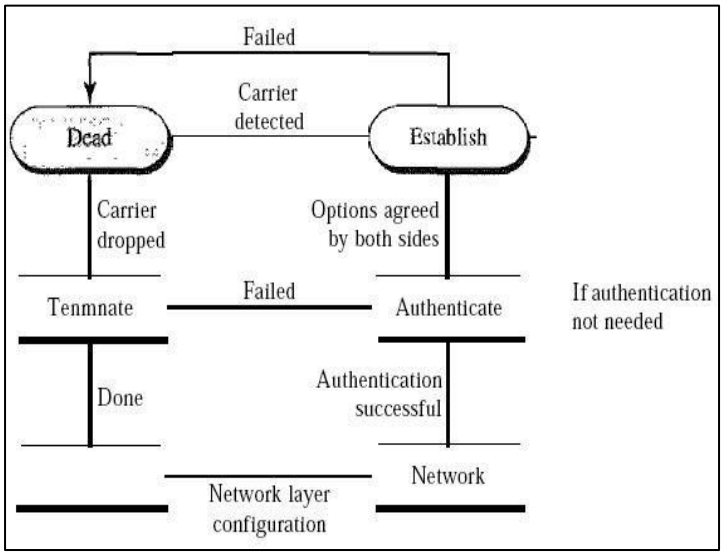
Protocol: The protocol field defines what is being carried in the data field: either user data or other information. We discuss this field in detail shortly. This field is by default 2 bytes long, but the two parties can agree to use only I byte.

Payload field: This field carries either the user data or other information that we will discuss shortly. The data field is a sequence of bytes with the default of a maximum of 1500 bytes; but this can be changed during negotiation. The data field is byte stuffed if the flag byte pattern appears in this field. Because there is no field defining the size of the data field, padding is needed if the size is less than the maximum default value or the maximum negotiated value.

FCS: The frame check sequence (FCS) is simply a 2-byte or 4-byte standard CRC.

Transition Phases

A PPP connection goes through phases which can be shown in a transition phase diagram.



Dead: In the dead phase the link is not being used. There is no active carrier (at the physical layer) and the line is quiet.

Establish: When one of the nodes starts the communication, the connection goes into this phase. In this phase, options are negotiated between the two parties. If the negotiation is successful, the system goes to the authentication phase (if authentication is required) or directly to the networking phase. The link control protocol packets, discussed shortly, are used for this purpose. Several packets may be exchanged here.

Authenticate: The authentication phase is optional; the two nodes may decide, during the establishment phase, not to skip this phase. However, if they decide to proceed with authentication, they send several authentication packets, discussed later. If the result is successful, the connection goes to the networking phase; otherwise, it goes to the termination phase.

Network: In the network phase, negotiation for the network layer protocols takes place. PPP specifies that two nodes establish a network layer agreement before data at the network layer can be exchanged. The reason is that PPP supports multiple protocols at the network layer. If a node is running multiple protocols simultaneously at the network layer, the receiving node needs to know which protocol will receive the data.

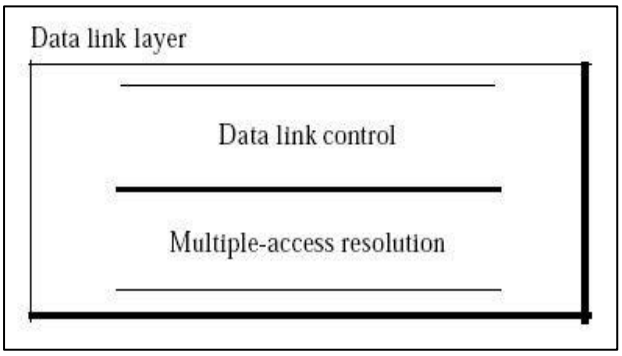
Open: In the open phase, data transfer takes place. When a connection reaches this phase, the exchange of data packets can be started. The connection remains in this phase until one of the endpoints wants to terminate the connection.

Terminate: In the termination phase the connection is terminated. Several packets are exchanged between the two ends for house cleaning and closing the link.

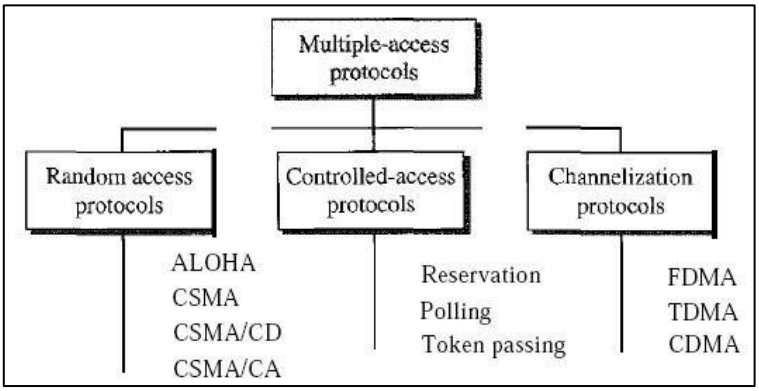
CHAPTER 10

MULTIPLE ACCESS

- If we use our cellular phone to connect to another cellular phone, the channel (the band allocated to the vendor company) is not dedicated. A person a few feet away from us may be using the same channel to talk to her friend.
- We can consider the data link layer as two sublayers. The upper sublayer is responsible for data link control, and the lower sublayer is responsible for resolving access to the shared media. If the channel is dedicated, we do not need the lower sublayer.



Many formal protocols have been devised to handle access to a shared link. We categorize them into three groups. Protocols belonging to each group are shown in below figure:



RANDOM ACCESS

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

- Two features give this method its name. First, there is no scheduled time for a station to transmit. Transmission is random among the stations. That is why these methods are called random access. Second, no rules specify which station should send next. Stations compete with one another to access the medium. That is why these methods are also called contention methods.
- In a random-access method, each station has the right to the medium without being controlled by any other station. However, if more than one station tries to send, there is an access conflict collision-and the frames will be either destroyed or modified. To avoid access conflict or to resolve it when it happens, each station follows a procedure that answers the following questions:
  - When can the station access the medium?
  - What can the station do if the medium is busy?
  - How can the station determine the success or failure of the transmission?
  - What can the station do if there is an access conflict?

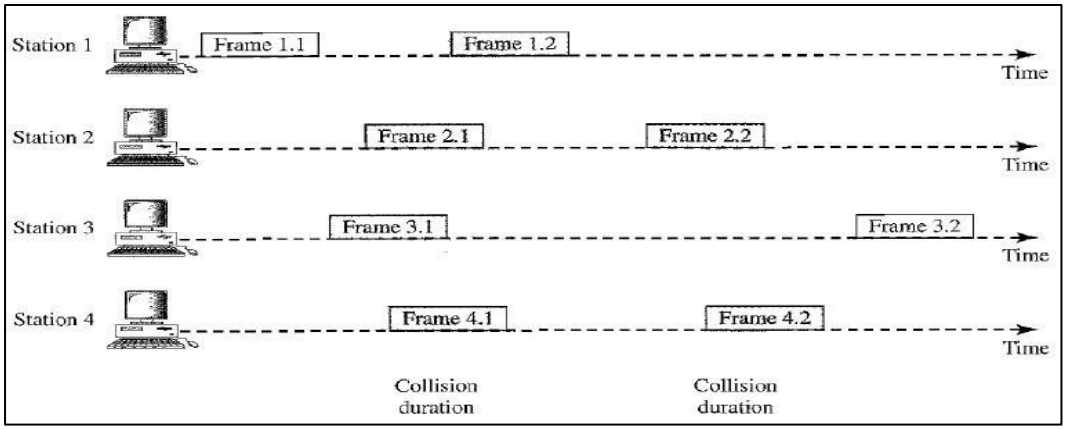
ALOHA

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

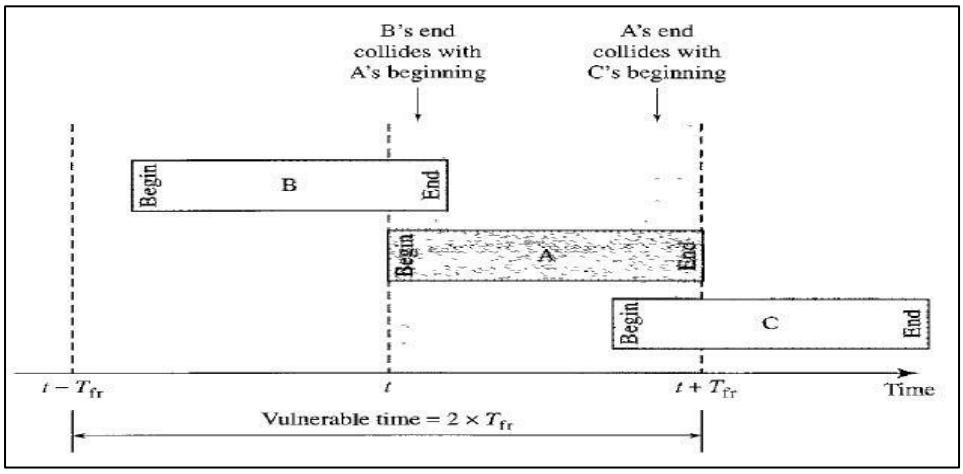
Pure ALOHA

- The original ALOHA protocol is called *pure ALOHA*. This is a simple, but elegant protocol.
- The idea is that each station sends a frame whenever it has a frame to send.
- However, since there is only one channel to share, there is the possibility of collision between frames from different stations.
- It is obvious that we need to resend the frames that have been destroyed during transmission. The pure ALOHA protocol relies on acknowledgments from the receiver.
- When a station sends a frame, it expects the receiver to send an acknowledgment. If the acknowledgment does not arrive after a time-out period, the station assumes that the frame (or the acknowledgment) has been destroyed and resends the frame.

There are four stations (unrealistic assumption) that contend with one another for access to the shared channel. The figure shows that each station sends two frames; there are a total of eight frames on the shared medium. Some of these frames collide because multiple frames are in contention for the shared channel. Below figure shows that only two frames survive: frame 1.1 from station 1 and frame 3.2 from station 3. We need to mention that even if one bit of a frame coexists on the channel with one bit from another frame, there is a collision, and both will be destroyed.



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



Pure ALOHA vulnerable time = 2 x T_{fr}

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

Solution:

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

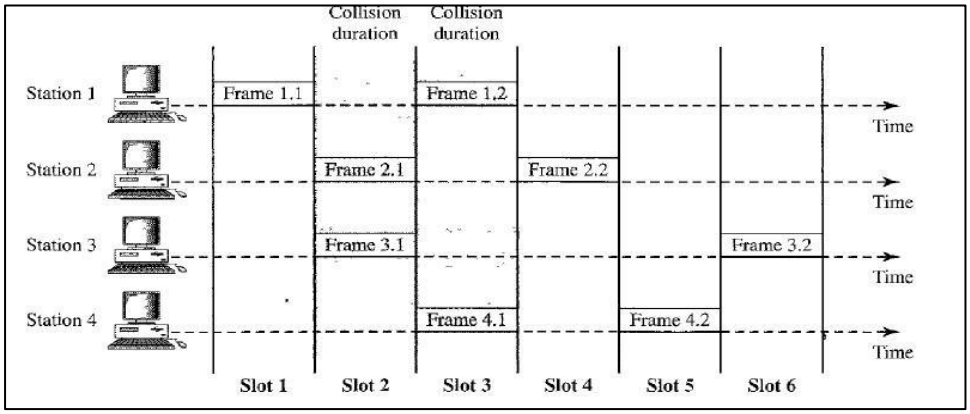
Throughput: Let us call G the average number of frames generated by the system during one frame transmission time. Then it can be proved that the average number of successful transmissions for pure ALOHA is S = G x e^{-2G}. The maximum throughput Smax is 0.184, for G = 1/2. In other words, if one-half a frame is generated during one frame transmission

time (in other words, one frame during two frame transmission times), then 18.4 percent of these frames reach their destination successfully.

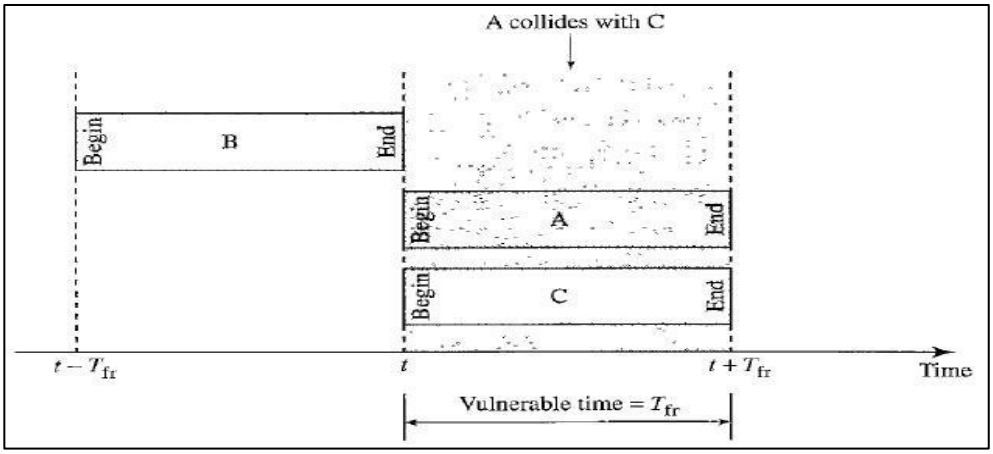
The throughput for pure ALOHA is S =G x e^{-2G}.
The maximum throughput Smax =0.184 when G =(1/2).

Slotted Aloha

Slotted ALOHA was invented to improve the efficiency of pure ALOHA. In slotted ALOHA we divide the time into slots of T_{fr} s and force the station to send only at the beginning of the time slot.



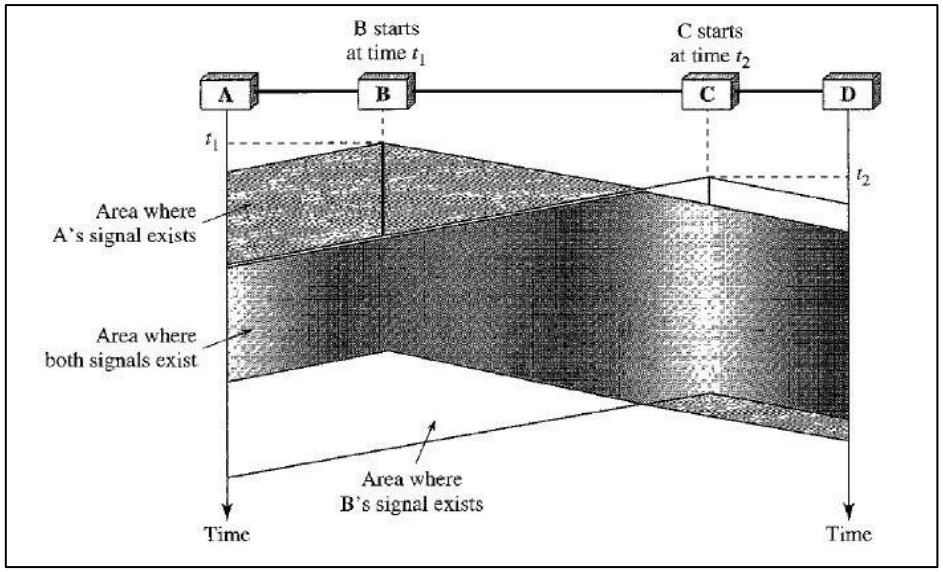
Slotted ALOHA vulnerable time = T_{fr}



The throughput for slotted ALOHA is S =G x e^{-G}.
The maximum throughput Smax =0.368when G =1.

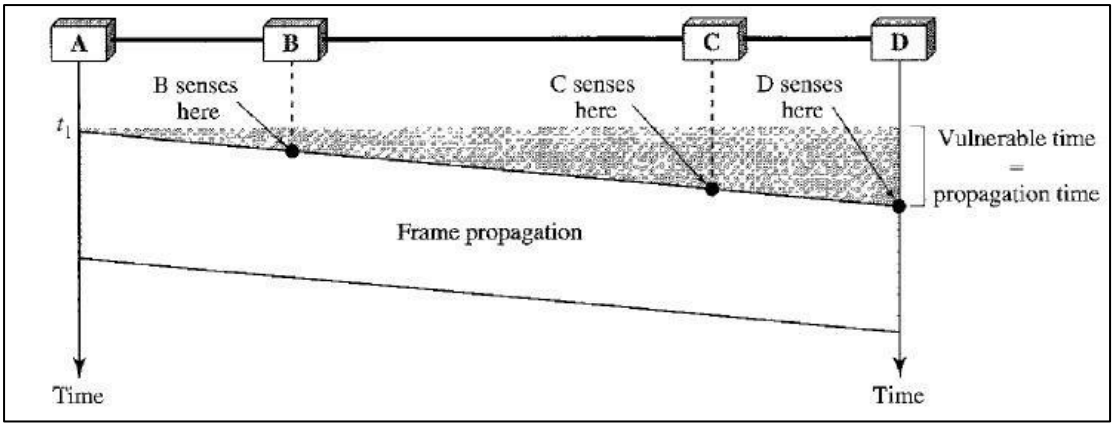
Carrier Sense Multiple Access (CSMA)

- To minimize the chance of collision and, therefore, increase the performance, the CSMA method was developed.
- The chance of collision can be reduced if a station senses the medium before trying to use it.
- Carrier sense multiple access (CSMA) requires that each station first listen to the medium (or check the state of the medium) before sending. In other words, CSMA is based on the principle "sense before transmit" or "listen before talk." CSMA can reduce the possibility of collision, but it cannot eliminate it.



Vulnerable Time

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



Persistence Methods

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

1-Persistent: The **1-persistent method** is simple and straightforward. In this method, after the station finds the line idle, it sends its frame immediately (with probability I). This method has the highest chance of collision because two or more stations may find the line idle and send their frames immediately.

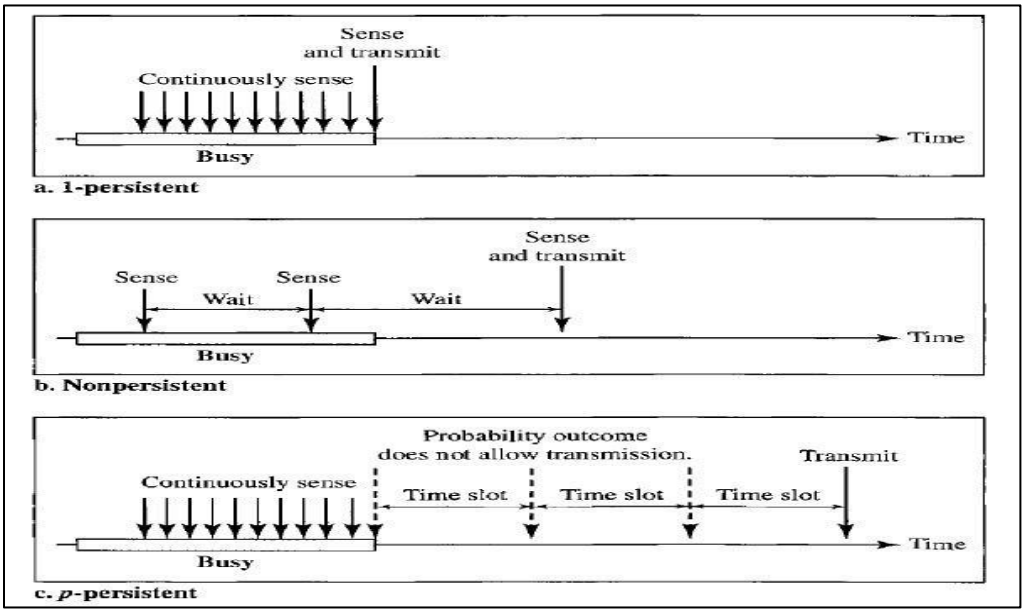
Non-persistent: In the **non-persistent method,** a station that has a frame to send senses the line. If the line is idle, it sends immediately. If the line is not idle, it waits a random amount of time and then senses the line again. The non-persistent approach reduces the chance of collision because it is unlikely that two or more stations will wait the same amount of time and retry to send simultaneously. However, this method reduces the efficiency of the network because the medium remains idle when there may be stations with frames to send.

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

1. With probability p, the station sends its frame.
2. With probability q = 1 - p, the station waits for the beginning of the next time slot and checks the line again.

a. If the line is idle, it goes to step 1.

b. If the line is busy, it acts as though a collision has occurred and uses the backoff procedure.

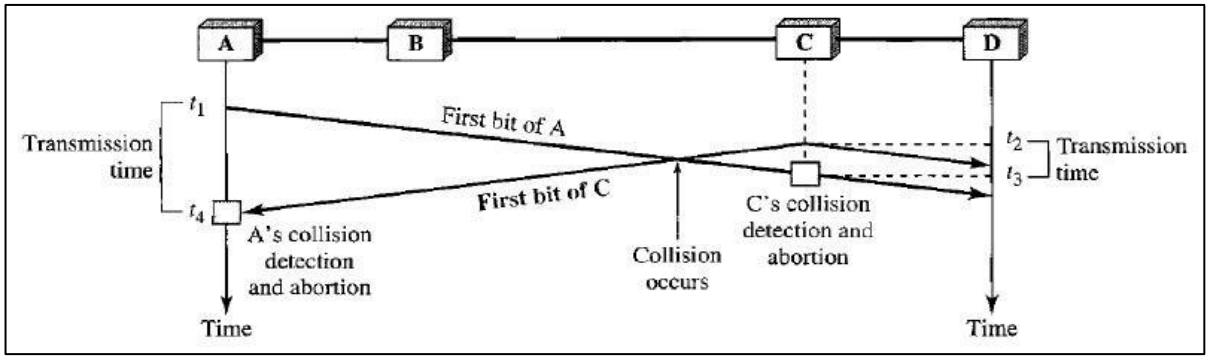


Carrier Sense Multiple Access with Collision Detection (CSMA/CD)

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

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

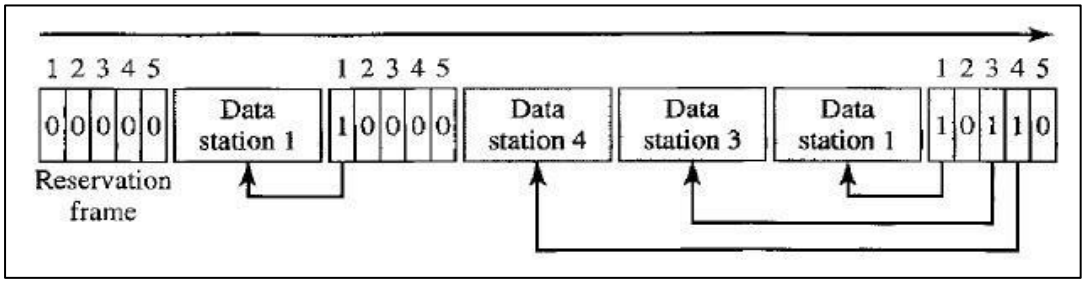
The basic idea behind CSMA/CD is that a station needs to be able to receive while transmitting to detect a collision. When there is no collision, the station receives one signal: its own signal. When there is a collision, the station receives two signals: its own signal and the signal transmitted by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



CONTROLLED ACCESS

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

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



Polling: Polling works with topologies in which one device is designated as a primary station and the other devices are secondary stations. All data exchanges must be made through the primary device even when the ultimate destination is a secondary device. The primary device

controls the link; the secondary devices follow its instructions. It is up to the primary device to determine which device is allowed to use the channel at a given time.

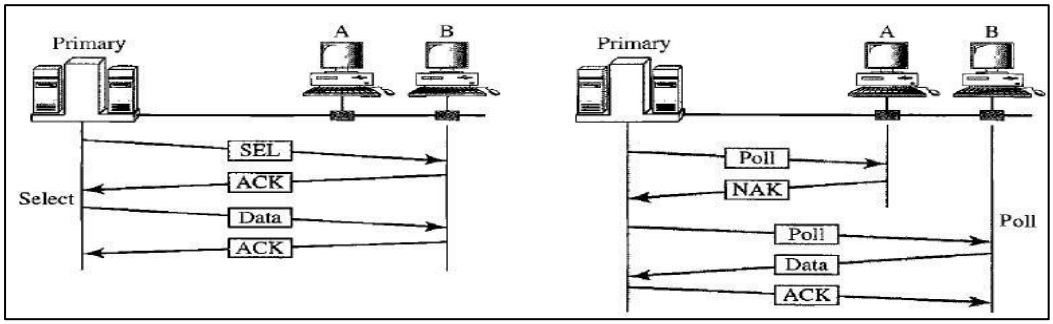
If the primary wants to receive data, it asks the secondary if they have anything to send; this is called poll function. If the primary wants to send data, it tells the secondary to get ready to receive; this is called select function.

1. Select

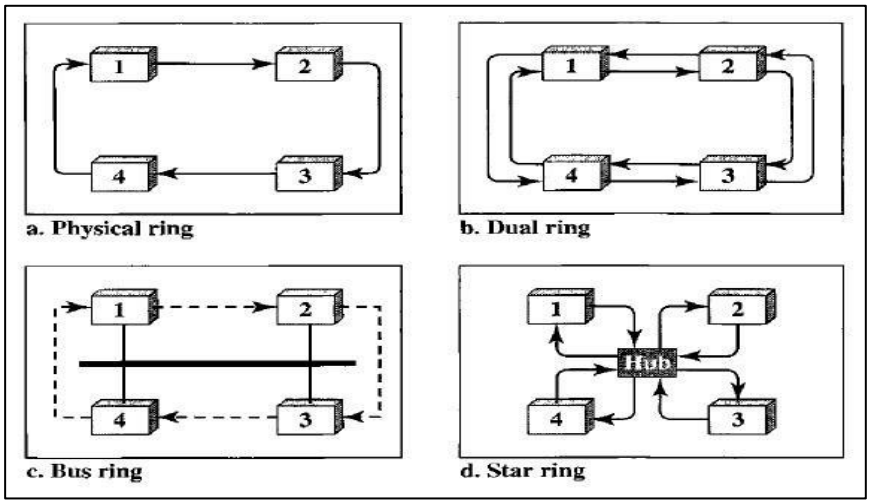
The select function is used whenever the primary device has something to send. Remember that the primary controls the link. If the primary is neither sending nor receiving data, it knows the link is available.

2. Poll

The poll function is used by the primary device to solicit transmissions from the secondary devices. When the primary is ready to receive data, it must ask (poll) each device in turn if it has anything to send.



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

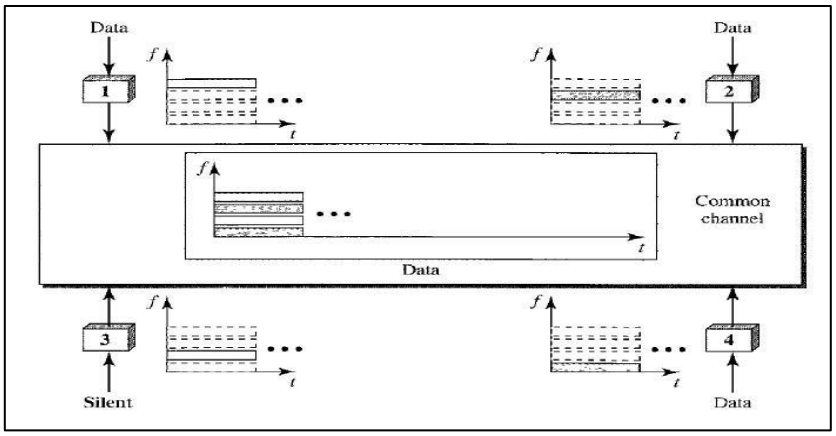


CHANNELIZATION

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.

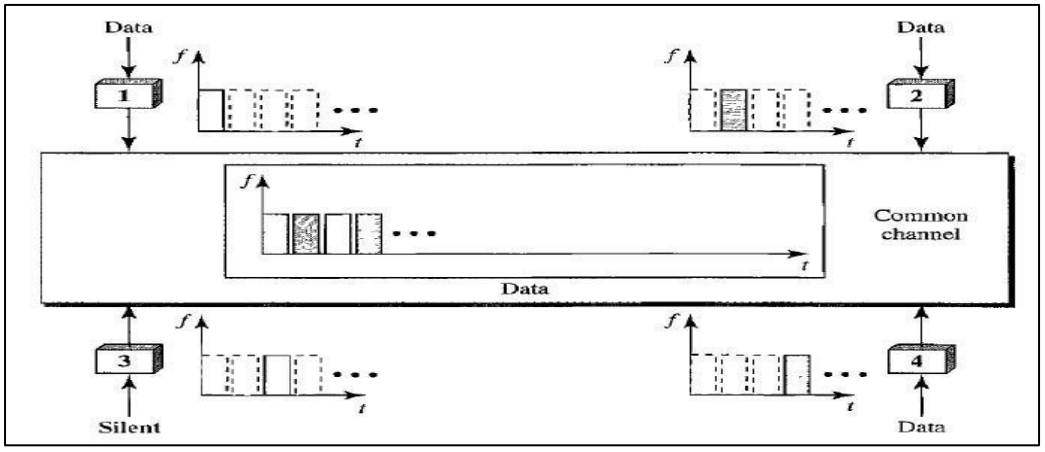
- **Frequency-Division Multiple Access (FDMA)**

  In frequency-division multiple access (FDMA), the available bandwidth is divided into frequency bands. Each station is allocated a band to send its data.



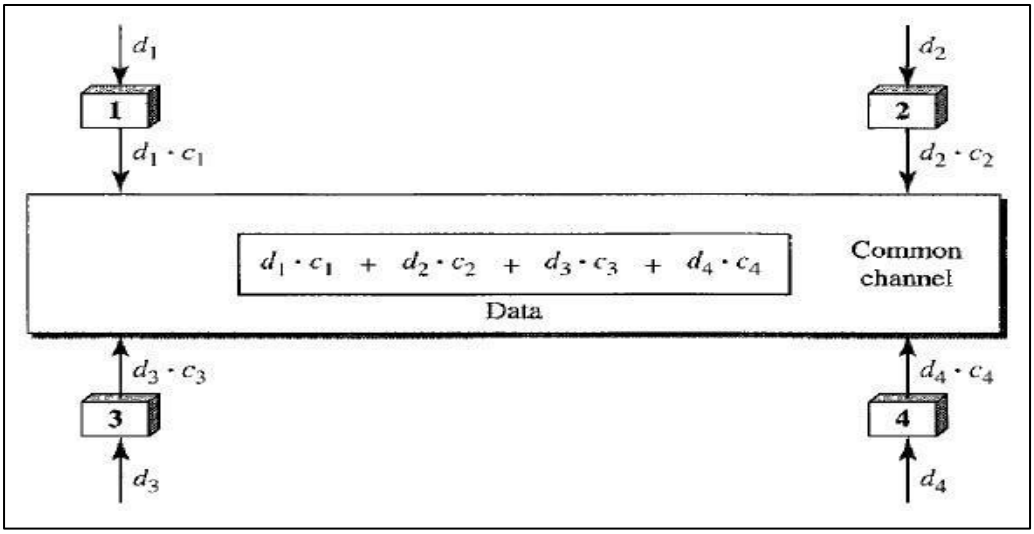
- **Time-Division Multiple Access (TDMA)**

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



- **Code-Division Multiple Access (CDMA)**

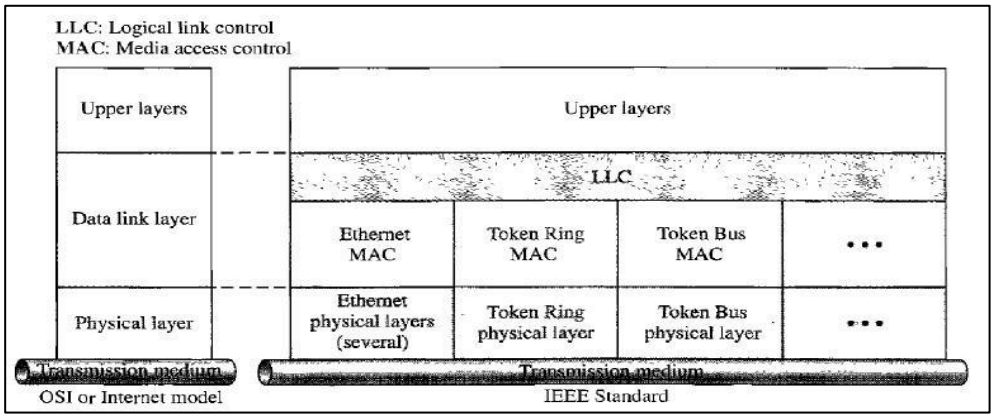
  CDMA differs from FDMA because only one channel occupies the entire bandwidth of the link. It differs from TDMA because all stations can send data simultaneously; there is no timesharing.


d_1
d_2
1
2
$d_1 \cdot c_1$
$d_2 \cdot c_2$
$d_1 \cdot c_1 + d_2 \cdot c_2 + d_3 \cdot c_3 + d_4 \cdot c_4$
Data
Common channel
$d_3 \cdot c_3$
$d_4 \cdot c_4$
3
4
d_3
d_4

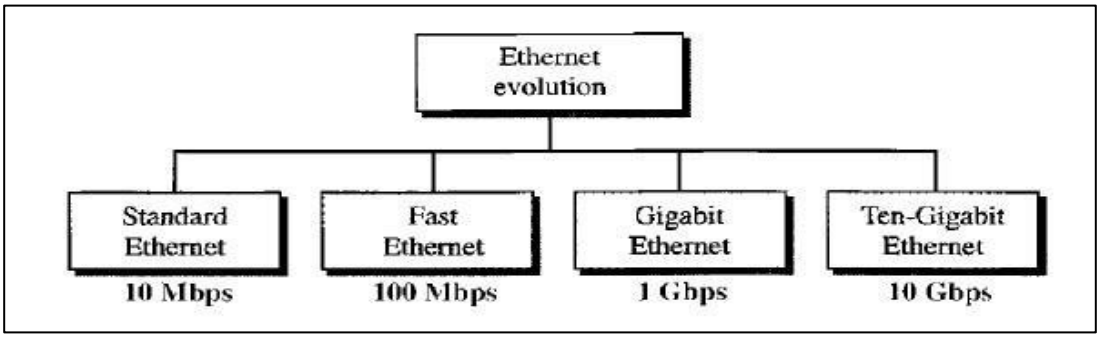
CHAPTER 11

WIRED LANS: Ethernet

The IEEE has subdivided the data link layer into two sub layers: logical link control (LLC) and media access control (MAC). IEEE has also created several physical layer standards for different LAN protocols.



The original Ethernet was created in 1976 at Xerox's Palo Alto Research Center (PARC). Since then, it has gone through four generations: Standard Ethernet (10 Mbps), Fast Ethernet (100 Mbps), Gigabit Ethernet (1 Gbps), and Ten-Gigabit Ethernet (10 Gbps).

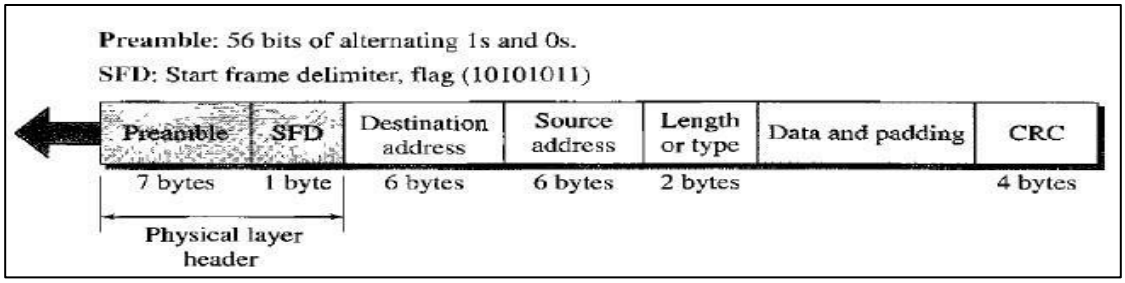


MAC Sublayer

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

Frame Format

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



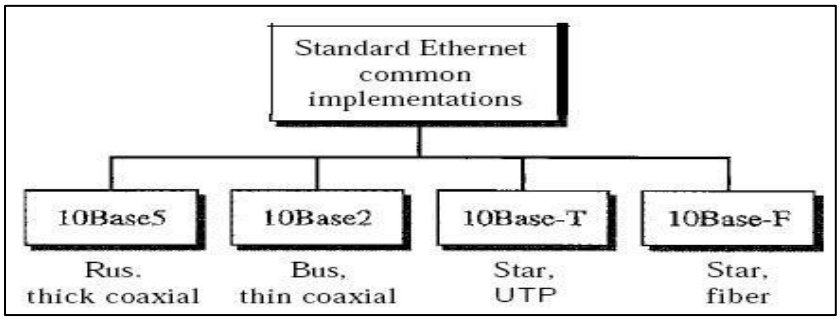
- **Preamble:** The first field of the 802.3 frame contains 7 bytes (56 bits) of alternating 0s and 1s that alerts the receiving system to the coming frame and enables it to synchronize its input timing.
- **Start frame delimiter (SFD):** The second field (1byte: 10101011) signals the beginning of the frame. The SFD warns the station or stations that this is the last chance for synchronization. The last 2 bits is 11 and alerts the receiver that the next field is the destination address.
- **Destination address (DA):** The DA field is 6 bytes and contains the physical address of the destination station or stations to receive the packet. We will discuss addressing shortly.
- **Source address (SA):** The SA field is also 6 bytes and contains the physical address of the sender of the packet. We will discuss addressing shortly.
- **Length or type:** This field is defined as a type field or length field. The original Ethernet used this field as the type field to define the upper-layer protocol using the MAC frame. The IEEE standard used it as the length field to define the number of bytes in the data field. Both uses are common today.
- **Data:** This field carries data encapsulated from the upper-layer protocols. It is a minimum of 46 and a maximum of 1500 byte.
- **CRC:** The last field contains error detection information, in this case a CRC-32.

Physical Layer

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

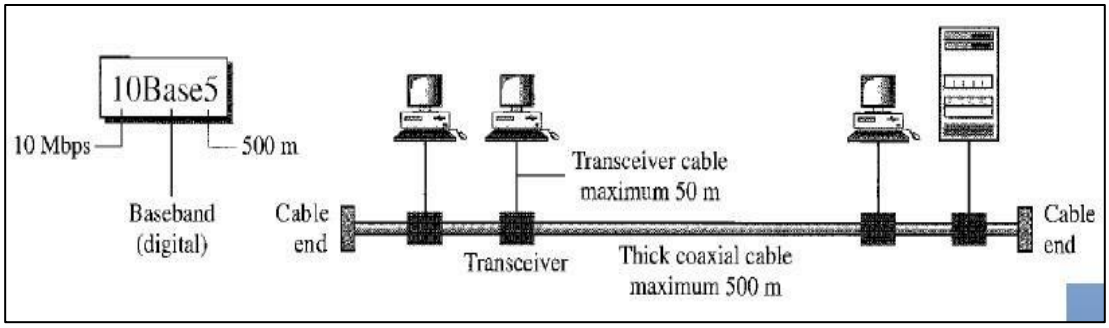
Encoding and Decoding

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



lOBase5: Thick Ethernet

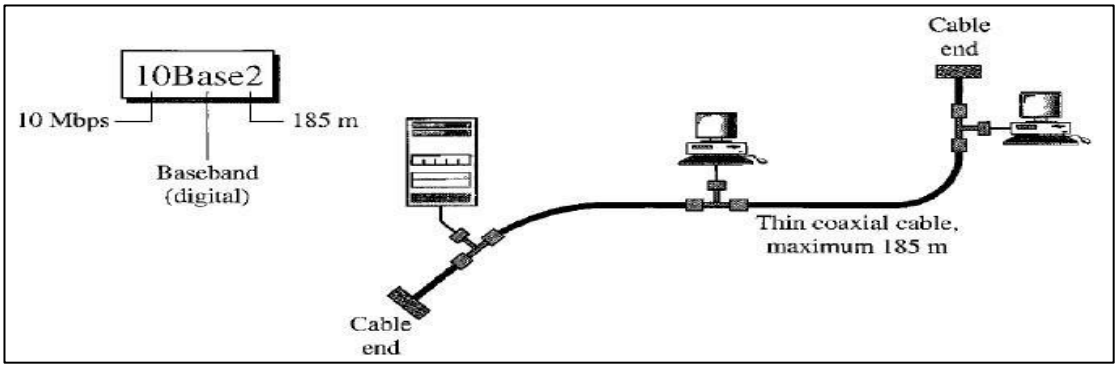
- The first implementation is called **10Base5, thick Ethernet, or Thicknet.**
- 10Base5 was the first Ethernet specification to use a bus topology with an external **transceiver** (transmitter/receiver) connected via a tap to a thick coaxial cable.
- The transceiver is responsible for transmitting, receiving, and detecting collisions.
- The transceiver is connected to the station via a transceiver cable that provides separate paths for sending and receiving.



- The maximum length of the coaxial cable must not exceed *500* m, otherwise, there is excessive degradation of the signal.

10Base2: Thin Ethernet

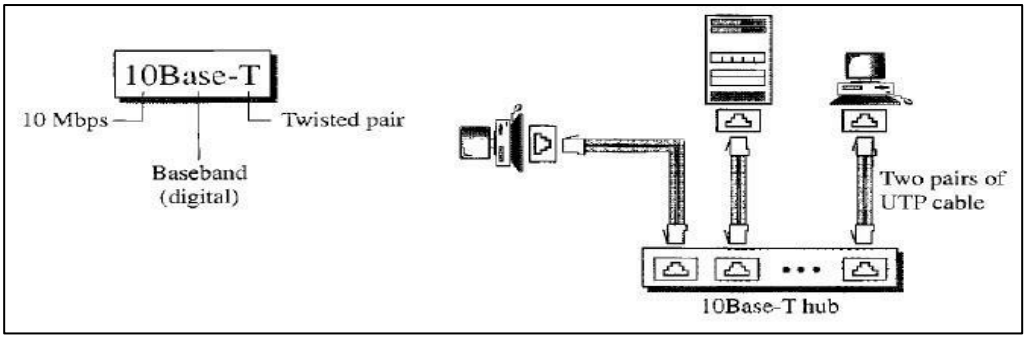
- The second implementation is called lOBase2, **thin** Ethernet, or Cheapernet.
- 10Base2 also uses a bus topology, but the cable is much thinner and more flexible.
- The cable can be bent to pass very close to the stations. In this case, the transceiver is normally part of the network interface card (NIC), which is installed inside the station.



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

10Base-T: Twisted-Pair Ethernet

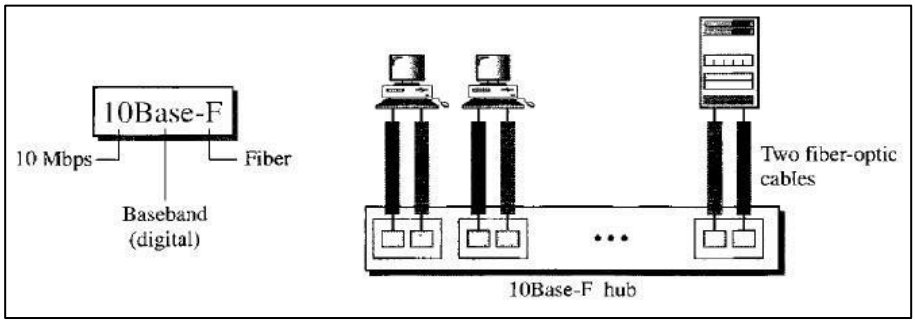
- The third implementation is called 10Base-T or twisted-pair Ethernet. 10Base-T uses a physical star topology.



- Any collision here happens in the hub.
- Compared to l0BaseS or l0Base2, we can see that the hub actually replaces the coaxial cable as far as a collision is concerned.
- The maximum length of the twisted cable here is defined as 100 m, to minimize the effect of attenuation in the twisted cable.

10Base-F: Fiber Ethernet

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



Fast Ethernet

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

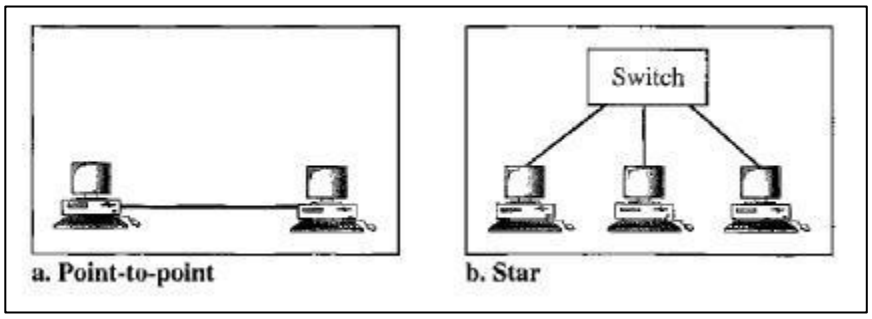
1. Upgrade the data rate to 100 Mbps.
2. Make it compatible with Standard Ethernet.
3. Keep the same 48-bit address.
4. Keep the same frame format.
5. Keep the same minimum and maximum frame lengths.

Physical Layer

The physical layer in Fast Ethernet is more complicated than the one in Standard Ethernet.

We briefly discuss some features of this layer.

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



[Fast Ethernet Topology]

- **Encoding:** Manchester encoding needs a 200-Mbaud bandwidth for a data rate of 100 Mbps, which makes it unsuitable for a medium such as twisted-pair cable. For this reason, the Fast Ethernet designers sought some alternative encoding/decoding scheme.

Gigabit Ethernet

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

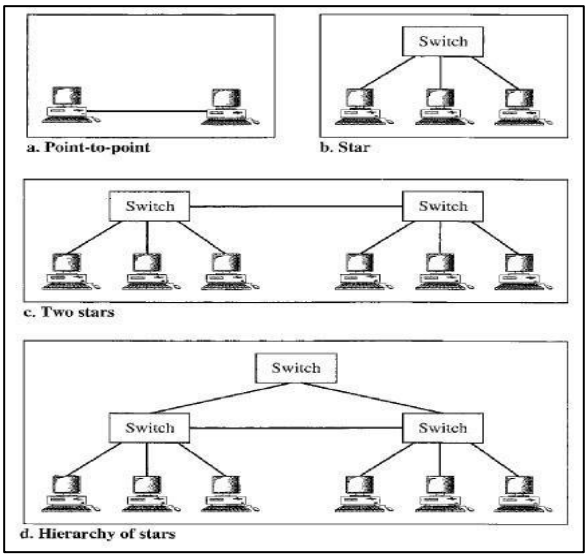
1. Upgrade the data rate to 1 Gbps.
2. Make it compatible with Standard or Fast Ethernet.
3. Use the same 48-bit address.
4. Use the same frame format.
5. Keep the same minimum and maximum frame lengths.
6. To support auto negotiation as defined in Fast Ethernet.

Physical Layer

The physical layer in Gigabit Ethernet is more complicated than that in Standard or Fast Ethernet. We briefly discuss some features of this layer.

- **Topology:** Gigabit Ethernet is designed to connect two or more stations. If there are only two stations, they can be connected point-to-point. Three or more stations need to be connected in a star topology with a hub or a switch at the center. Another possible

configuration is to connect several star topologies or let a star topology be part of another as shown in Figure.



- **Encoding:** Gigabit Ethernet cannot use the Manchester encoding scheme because it involves a very high bandwidth. The two-wire implementations use an NRZ scheme, but NRZ does not selfsynchronize properly.

Token Bus (IEEE 802.4)

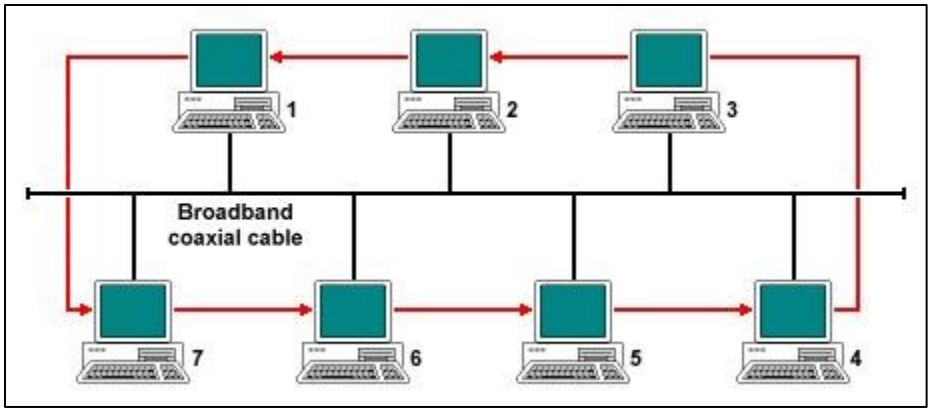
Token bus is a network implementing the token ring protocol over a "virtual ring" on a coaxial cable. A token is passed around the network nodes and only the node possessing the token may transmit. If a node doesn't have anything to send, the token is passed on to the next node on the virtual ring. Each node must know the address of its neighbor in the ring, so a special protocol is needed to notify the other nodes of connections to, and disconnections from, the ring.

Token bus was standardized by IEEE standard 802.4. It is mainly used for industrial applications. Token bus was used by General Motors for their Manufacturing Automation Protocol (MAP) standardization effort. This is an application of the concepts used in token ring networks. The main difference is that the endpoints of the bus do not meet to form a physical ring.

Token Bus suffered from two limitations. Any failure in the bus caused all the devices beyond the failure to be unable to communicate with the rest of the network. Second, adding more stations to the bus was somewhat difficult. Any new station that was improperly attached was unlikely to be able to communicate and all devices beyond it were also affected. Thus, token bus networks were seen as somewhat unreliable and difficult to expand and upgrade.

In order to guarantee the packet delay and transmission in Token bus protocol, a modified Token bus was proposed in Manufacturing Automation Systems and flexible manufacturing system (FMS).

A means for carrying Internet Protocol over token bus was developed.



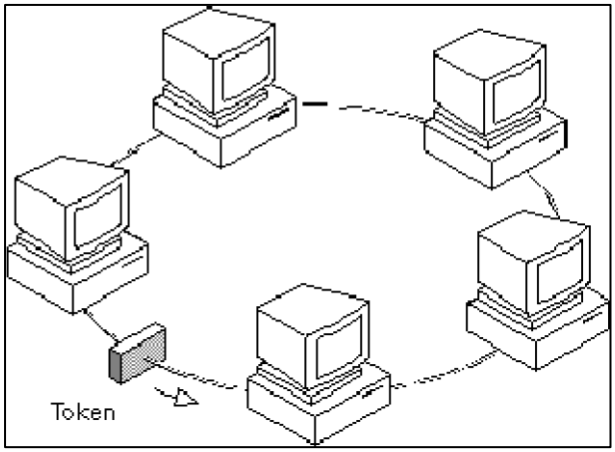
Token ring (IEEE 802.5)

Token ring local area network (LAN) technology is a local area network protocol which resides at the data link layer (DLL) of the OSI model. It uses a special three-byte frame called a token that travels around the ring. Token-possession grants the possessor permission to transmit on the medium. Token ring frames travel completely around the loop.

Stations on a token ring LAN are logically organized in a ring topology with data being transmitted sequentially from one ring station to the next with a control token circulating around the ring controlling access. This token passing mechanism is shared by ARCNET, token bus, and FDDI, and has theoretical advantages over the stochastic CSMA/CD of Ethernet.

Physically, a token ring network is wired as a star, with 'hubs' and arms out to each station and the loop going out-and-back through each.

Each station passes or repeats the special token frame around the ring to its nearest downstream neighbor. This token-passing process is used to arbitrate access to the shared ring media. Stations that have data frames to transmit must first acquire the token before they can transmit them. Token ring LANs normally use differential Manchester encoding of bits on the LAN media.



Token Ring does come with a higher price tag because token ring hardware is more complex and more expensive to manufacture. As a network technology, token ring is passing out of use because it has a maximum speed of 16 Mbps which is slow by today's gigabit Ethernet standards.

Wireless LANs

Wireless communication is one of the fastest-growing technologies. The demand for connecting devices without the use of cables is increasing everywhere. Wireless LANs can be found on college campuses, in office buildings, and in many public areas.

Frame Relay

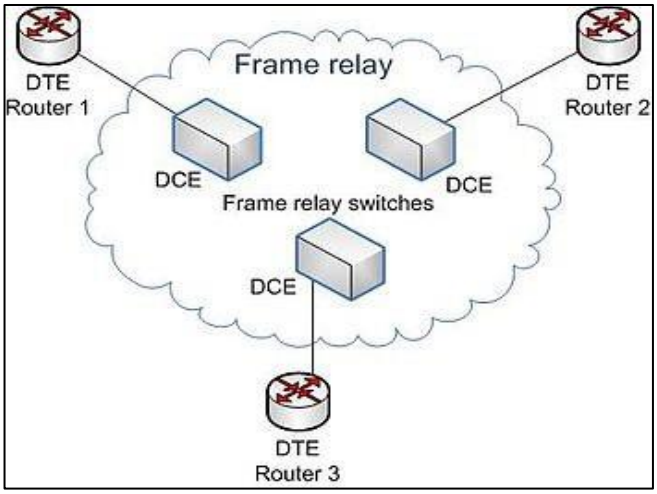
Frame Relay is a standardized wide area network technology that specifies the physical and logical link layers of digital telecommunications channels using a packet switching methodology. Originally designed for transport across Integrated Services Digital Network (ISDN) infrastructure, it may be used today in the context of many other network interfaces.

Frame Relay has its technical base in the older X.25 packet-switching technology, designed for transmitting data on analog voice lines. Unlike X.25, whose designers expected analog signals, Frame Relay offers a fast packet technology, which means that the protocol does not attempt to correct errors. When a Frame Relay network detects an error in a frame, it simply drops that frame. The end points have the responsibility for detecting and retransmitting dropped frames. (However, digital networks offer an incidence of error extraordinarily small relative to that of analog networks.)

Frame Relay often serves to connect local area networks (LANs) with major backbones as well as on public wide-area networks (WANs) and also in private network environments with leased lines over T-1 lines. It requires a dedicated connection during the transmission period. Frame Relay does not provide an ideal path for voice or video transmission, both of which require a steady flow of transmissions. However, under certain circumstances, voice and video transmission do use Frame Relay.

Frame Relay originated as an extension of Integrated Services Digital Network (ISDN). Its designers aimed to enable a packet-switched network to transport the circuit-switched technology. The technology has become a stand-alone and cost-effective means of creating a WAN.

Frame Relay switches create virtual circuits to connect remote LANs to a WAN. The Frame Relay network exists between a LAN border device, usually a router, and the carrier switch. The technology used by the carrier to transport data between the switches is variable and may differ among carriers (i.e. to function, a practical Frame Relay implementation need not rely solely on its own transportation mechanism).



Protocol data unit

Each Frame Relay Protocol data unit (PDU) consists of the following fields:

1. **Flag Field**. The flag is used to perform high-level data link synchronization which indicates the beginning and end of the frame with the unique pattern 01111110. To ensure that the 01111110 pattern does not appear somewhere inside the frame, bit stuffing and destuffing procedures are used.
2. **Address Field**. Each address field may occupy octet 2 to 3, octet 2 to 4, or octet 2 to 5, depending on the range of the address in use. A two-octet address field comprises the EA=ADDRESS FIELD EXTENSION BITS and the C/R=COMMAND/RESPONSE BIT.
    1. **DLCI**-Data Link Connection Identifier Bits. The DLCI serves to identify the virtual connection so that the receiving end knows which information connection a frame belongs to. Note that this DLCI has only local significance. A single physical channel can multiplex several different virtual connections.
    2. **FECN, BECN, DE** bits. These bits report congestion: **FECN**=Forward Explicit Congestion Notification bit

       BECN=Backward Explicit Congestion Notification bit

       DE=Discard Eligibility bit
3. **Information Field**. A system parameter defines the maximum number of data bytes that a host can pack into a frame. Hosts may negotiate the actual maximum frame length at call set-up time. The standard specifies the maximum information field size (supportable by any network)
   as at least 262 octets. Since end-to-end protocols typically operate on the basis of larger information units, Frame Relay recommends that the network support the maximum value

of at least 1600 octets in order to avoid the need for segmentation and reassembling by end-users.

4. **Frame Check Sequence (FCS) Field**. Since one cannot completely ignore the bit error-rate of the medium, each switching node needs to implement error detection to avoid wasting bandwidth due to the transmission of *err*ed frames. The error detection mechanism used in Frame Relay uses the cyclic redundancy check (CRC) as its basis.

Frame Relay versus X.25

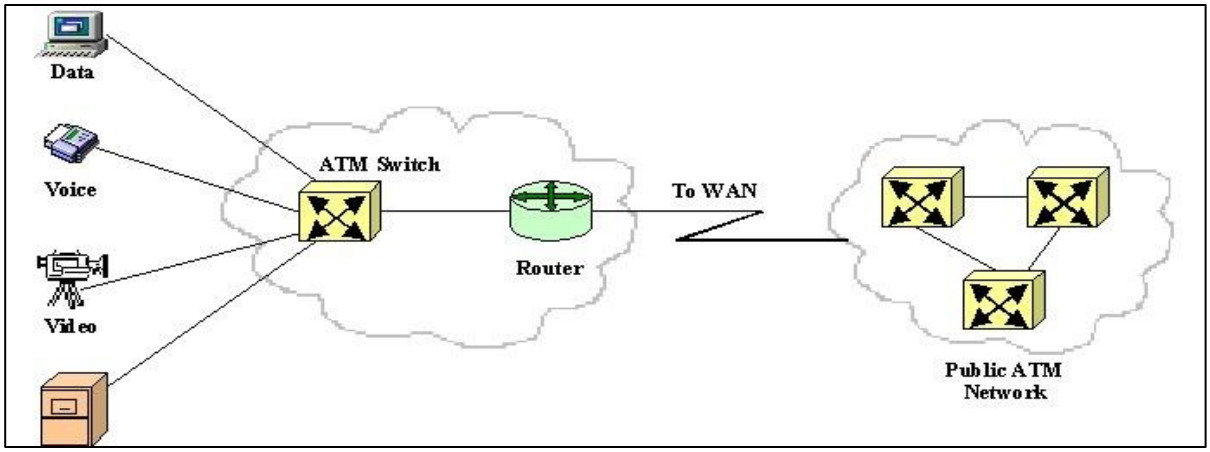
X.25 provides quality of service and error-free delivery, whereas, Frame Relay was designed to relay data as quickly as possible over low error networks. Frame Relay eliminates a number of the higher-level procedures and fields used in X.25. Frame Relay was designed for use on links with error-rates far lower than available when X.25 was designed.

X.25 prepares and sends packets, while Frame Relay prepares and sends frames. X.25 packets contain several fields used for error checking and flow control, most of which are not used by Frame Relay. The frames in Frame Relay contain an expanded link layer address field that enables Frame Relay nodes to direct frames to their destinations with minimal processing. The elimination of functions and fields over X.25 allows Frame Relay to move data more quickly, but leaves more room for errors and larger delays should data need to be retransmitted.

X.25 packet switched networks typically allocated a fixed bandwidth through the network for each X.25 access, regardless of the current load. This resource allocation approach, while apt for applications that require guaranteed quality of service, is inefficient for applications that are highly dynamic in their load characteristics or which would benefit from a more dynamic resource allocation. Frame Relay networks can dynamically allocate bandwidth at both the physical and logical channel level.

ATM

Asynchronous Transfer Mode (ATM) is a standard switching technique, designed to unify telecommunication and computer networks. It uses asynchronous time-division multiplexing, and it encodes data into small, fixed-sized cells. This differs from approaches such as the Internet Protocol or Ethernet that use variable sized packets or frames. ATM provides data link layer services that run over a wide range of OSI physical Layer links. ATM has functional similarity with both circuit switched networking and small packet switched networking. It was designed for a network that must handle both traditional high-throughput data traffic (e.g., file transfers), and real-time, lowlatency content such as voice and video. ATM uses a connection-oriented model in which a virtual circuit must be established between two endpoints before the actual data exchange begins.


Data
Voice
Video
ATM Switch
Router
To WAN
Public ATM
Network

Chapter 12

Network Layer

- Communication at the network layer is host-to-host (computer-to-computer); a computer somewhere in the world needs to communicate with another computer somewhere else in the world.
- Usually, computers communicate through the Internet. The packet transmitted by the sending computer may pass through several LANs or WANs before reaching the destination computer.
- The Internet addresses are 32 bits in length; this gives us a maximum of 2^{32} addresses. These addresses are referred to as IPv4 (IP version 4) addresses or simply IP addresses if there is no confusion.
- The need for more addresses, in addition to other concerns about the IP layer, motivated a new design of the IP layer called the new generation of IP or IPv6 (lP version 6). In this version, the Internet uses 128-bit addresses that give much greater flexibility in address allocation. These addresses are referred to as IPv6 (IP version 6) addresses.

IPv4 ADDRESSES

- An **IPv4** address is a 32-bit address that uniquely and universally defines the connection of a device (for example, a computer or a router) to the Internet.
- IPv4 addresses are unique. They are unique in the sense that each address defines one, and only one, connection to the Internet. Two devices on the Internet can never have the same address at the same time.

Address Space

- A protocol such as IPv4 that defines addresses has an address space.
- An address space is the total number of addresses used by the protocol. If a protocol uses N bits to define an address, the address space is 2^N because each bit can have two different values (0 or 1) and N bits can have 2^N values.
- IPv4 uses 32-bit addresses, which means that the address space is 232 or 4,294,967,296 (more than 4 billion). This means that, theoretically, if there were no restrictions, more than 4 billion devices could be connected to the Internet.

Notations

Each TCP/IP host is identified by a logical IP address. This address is unique for each host that communicates by using TCP/IP. Each 32-bit IP address identifies a location of a host system on the network in the same way that a street address identifies a house on a city street.

Just as a street address has a standard two-part format (a street name and a house number), each IP address is separated internally into two parts--a network ID and a host ID:

- The network ID, also known as a network address, identifies a single network segment within a larger TCP/IP internetwork (a network of networks). All the systems that attach and

share access to the same network have a common network ID within their full IP address. This ID is also used to uniquely identify each network within the larger internetwork.

- The host ID, also known as a host address, identifies a TCP/IP node (a workstation, server, router, or other TCP/IP device) within each network. The host ID for each device identifies a single system uniquely within its own network.

There are two prevalent notations to show an IPv4 address: binary notation and dotted decimal notation.

- **Binary Notation**

  In binary notation, the IPv4 address is displayed as 32 bits. Each octet is often referred to as a byte. So it is common to hear an IPv4 address referred to as a 32-bit address or a 4-byte address. The following is an example of an IPv4 address in binary notation:
  01110101 10010101 00011101 00000010

- **Dotted-Decimal Notation**

  To make the IPv4 address more compact and easier to read, Internet addresses are usually written in decimal form with a decimal point (dot) separating the bytes. The following is the dotted decimal notation of the above address:
  117.149.29.2

Example 1

Change the following IPv4 addresses from binary notation to dotted-decimal notation.

a. 10000001 00001011 00001011 11101111

b. 11000001 10000011 00011011 11111111 **Solution:**

We replace each group of 8 bits with its equivalent decimal number (see Appendix B) and add dots for separation.

a. 129.11.11.239

b. 193.131.27.255

Example 2

Change the following IPv4 addresses from dotted-decimal notation to binary notation.

a. 111.56.45.78

b. 221.34.7.82

Solution :

We replace each decimal number with its binary equivalent (see Appendix B).

a. 01101111 00111000 00101101 01001110

b. 11011101 00100010 00000111 01010010

Example 3

Find the error, if any, in the following IPv4 addresses.

a. 111.56.045.78

b. 221.34.7.8.20

c. 75.45.301.14

d. 11100010.23.14.67

Solution :

a. There must be no leading zero (045).

b. There can be no more than four numbers in an IPv4 address.

c. Each number needs to be less than or equal to 255 (301 is outside this range).

d. A mixture of binary notation and dotted-decimal notation is not allowed.

Classful Addressing

- IPv4 addressing, at its inception, used the concept of classes. This architecture is called classful addressing.
- In classful addressing, the address space is divided into five classes: A, B, C, D, and E. Each class occupies some part of the address space.

[w.x.y.z]

Class	Value of *w*	Network ID	Host ID	Number of networks	Number of hosts per network
A	0-127	*w*	*x.y.z*	126	16,777,214
B	128-191	*w.x*	*y.z*	16,384	65,534
C	192-223	*w.x.y*	*z*	2,097,152	254
D	224-239	Reserved for multicast addressing	N/A	N/A	N/A
E	240-255	Reserved for experimental use	N/A	N/A	N/A

Example 4

Find the class of each address.

a. 00000001 00001011 00001011 11101111

b. 11000001 10000011 00011011 11111111

c. 14.23.120.8

d. 252.5.15.111 **Solution:**

a. The first bit is O. This is a class A address.

b. The first 2 bits are 1; the third bit is O. This is a class C address.

c. The first byte is 14 (between 0 and 127); the class is A.

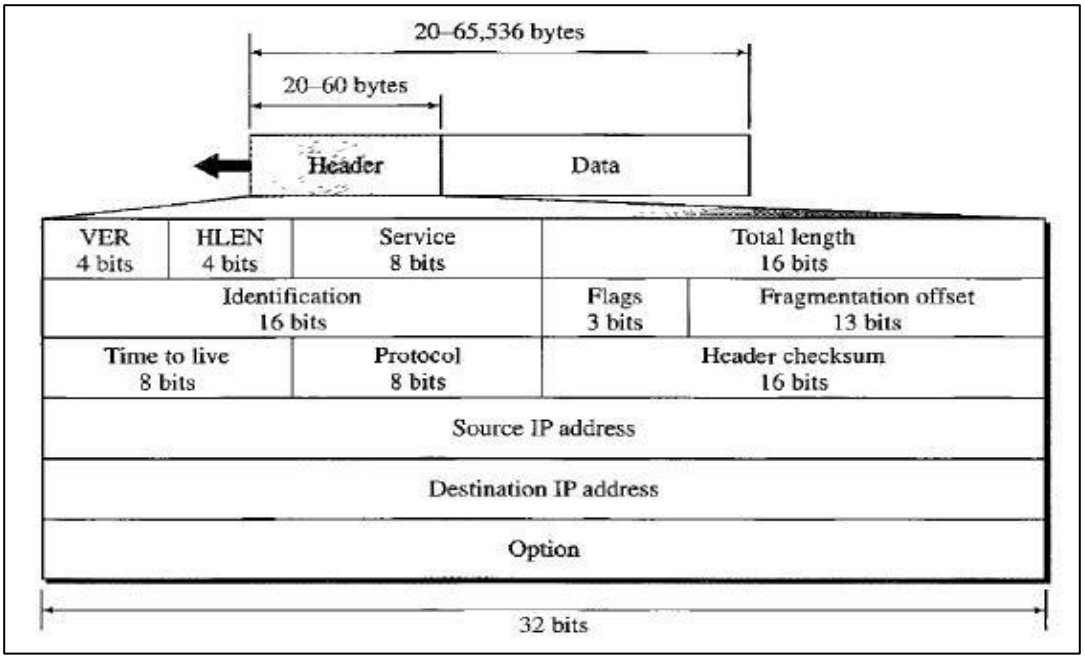
d. The first byte is 252 (between 240 and 255); the class is E.

IPv4

IPv4 is an unreliable and connectionless datagram protocol-a best-effort delivery service. The term besteffort means that IPv4 provides no error control or flow control (except for error detection on the header). IPv4 assumes the unreliability of the underlying layers and does its best to get a transmission through to its destination, but with no guarantees.

Datagram:

Packets in the IPv4 layer are called datagrams. Below figure shows the IPv4 datagram format.



Version (VER)

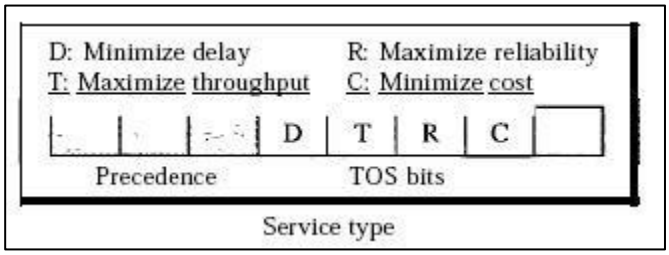
This 4-bit field defines the version of the IPv4 protocol. Currently the version is 4. However, version 6 (or IPng) may totally replace version 4 in the future.

Header length (HLEN)

This 4-bit field defines the total length of the datagram header in 4-byte words. This field is needed because the length of the header is variable (between 20 and 60 bytes). When there are no options, the header length is 20 bytes, and the value of this field is 5 (5 x 4 = 20). When the option field is at its maximum size, the value of this field is 15 (15 x 4 = 60).

Services

This field, previously called service type, is now called differentiated services.



Service Type

In this interpretation, the first 3 bits are called precedence bits. The next 4 bits are called type of service (TOS) bits, and the last bit is not used.

- Precedence is a 3-bit subfield ranging from 0 (000 in binary) to 7 (111 in binary). The precedence defines the priority of the datagram in issues such as congestion. If a router is congested and needs to discard some datagrams, those datagrams with lowest precedence are discarded first.
- TOS bits is a 4-bit subfield with each bit having a special meaning. Although a bit can be either 0 or 1, one and only one of the bits can have the value of 1 in each datagram. The bit patterns and their interpretations are given in below table.

TOS	Bits Description
0000	Normal (default)
0001	Minimize Cost
0010	Maximize Reliability
0100	Maximize Throughput
1000	Minimize Delay

Total length

This is a 16-bit field that defines the total length (header plus data) of the IPv4 datagram in bytes.

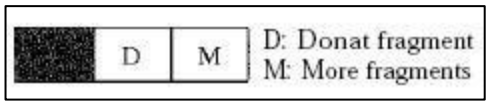
Length of data =total length - header length

Identification

This 16-bit field identifies a datagram originating from the source host. The combination of the identification and source IPv4 address must uniquely define a datagram as it leaves the source host.

Flags

This is a 3-bit field. The first bit is reserved. The second bit is called the do not fragment bit. If its value is 1, the machine must not fragment the datagram. The third bit is called the more fragment bit. If its value is 1, it means the datagram is not the last fragment; there are more fragments after this one. If its value is 0, it means this is the last or only fragment.



Fragmentation offset

This 13-bit field shows the relative position of this fragment with respect to the whole datagram.

Time to live

A datagram has a limited lifetime in its travel through an internet. This field was originally designed to hold a timestamp, which was decremented by each visited router. The datagram was discarded when the value became zero.

Protocol

This 8-bit field defines the higher-level protocol that uses the services of the IPv4 layer. An IPv4 datagram can encapsulate data from several higher-level protocols such as TCP, UDP, ICMP, and IGMP.

This field specifies the final destination protocol to which the IPv4 datagram is delivered.

Checksum

The checksum concept and its calculation are discussed later in this chapter.

Source address

This 32-bit field defines the IPv4 address of the source. This field must remain unchanged during the time the IPv4 datagram travels from the source host to the destination host.

Destination address

This 32-bit field defines the IPv4 address of the destination. This field must remain unchanged during the time the IPv4 datagram travels from the source host to the destination host.

Options

The header of the IPv4 datagram is made of two parts: a fixed part and a variable part. The fixed part is 20 bytes long and was discussed in the previous section. The variable part comprises the options that can be a maximum of 40 bytes.

IPv6

IPv6 (Internetworking Protocol, version 6), also known as **IPng** (Internetworking Protocol, next generation), was proposed and is now a standard. In IPv6, the Internet protocol was extensively modified to accommodate the unforeseen growth of the Internet. The format and the length of the IP address were changed along with the packet format.

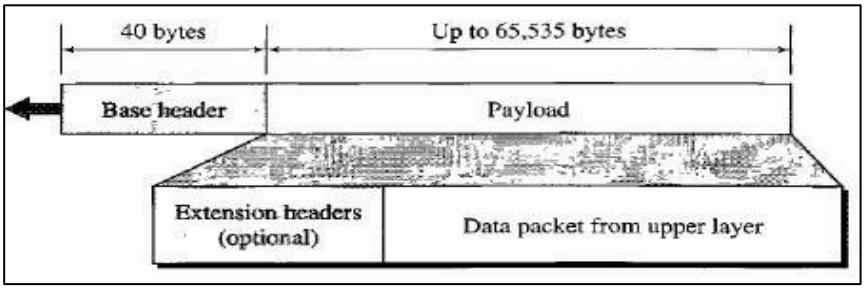
Advantages

The next-generation IP, or IPv6, has some advantages over IPv4 that can be summarized as follows:

- **Larger address space:** An IPv6 address is 128 bits long. Compared with the 32-bit address of IPv4, this is a huge (296) increase in the address space.
- **Better header format:** IPv6 uses a new header format in which options are separated from the base header and inserted, when needed, between the base header and the upper-layer data. This simplifies and speeds up the routing process because most of the options do not need to be checked by routers.
- **New options:** IPv6 has new options to allow for additional functionalities.
- **Allowance for extension:** IPv6 is designed to allow the extension of the protocol if required by new technologies or applications.
- **Support for resource allocation:** In IPv6, the type-of-service field has been removed, but a mechanism has been added to enable the source to request special handling of the packet. This mechanism can be used to support traffic such as real-time audio and video.
- **Support for more security:** The encryption and authentication options in IPv6 provide confidentiality and integrity of the packet.

Packet Format

The IPv6 packet is shown in below Figure. Each packet is composed of a mandatory base header followed by the payload. The payload consists of two parts: optional extension headers and data from an upper layer. The base header occupies 40 bytes, whereas the extension headers and data from the upper layer contain up to 65,535 bytes of information.



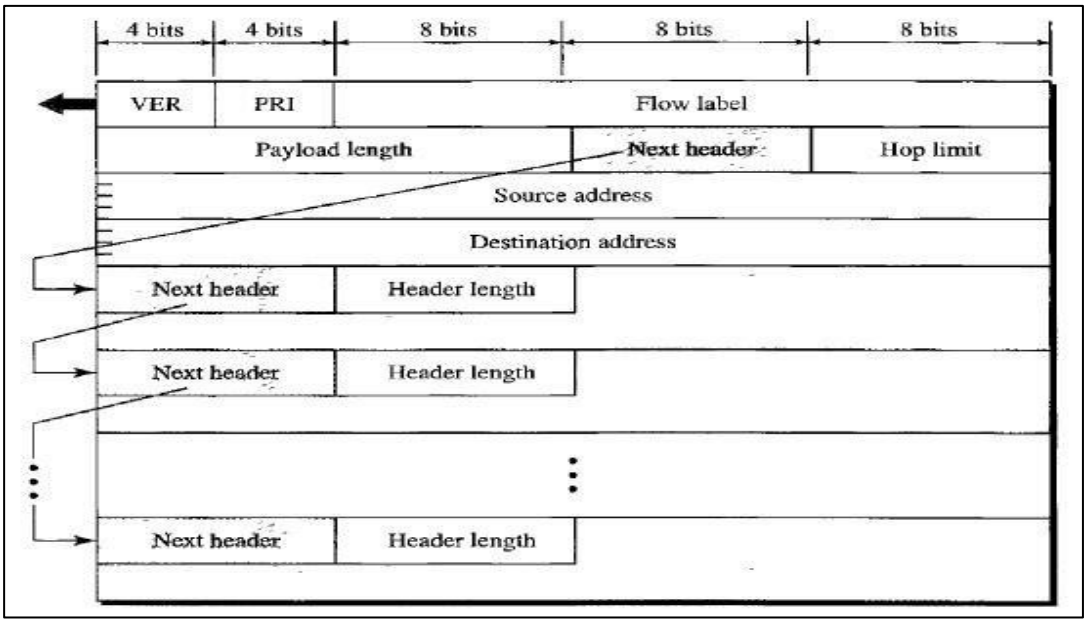
[IPv6 datagram header and payload]

Base Header

Below figure shows the base header with its eight fields.

These fields are as follows:

- **Version:** This 4-bit field defines the version number of the IP. For IPv6, the value is 6.
- **Priority:** The 4-bit priority field defines the priority of the packet with respect to traffic congestion.
- **Flow label.** The flow label is a 3-byte (24-bit) field that is designed to provide special handling for a particular flow of data.
- **Payload length.** The 2-byte payload length field defines the length of the IP datagram excluding the base header.
- **Next header.** The next header is an 8-bit field defining the header that follows the base header in the datagram. The next header is either one of the optional extension headers used by IP or the header of an encapsulated packet such as UDP or TCP. Each extension header also contains this field. Table 20.6 shows the values of next headers. Note that this field in version 4 is called the *protocol*.
- **Hop limit**. This 8-bit hop limit field serves the same purpose as the TTL field in IPv4.
- **Source address.** The source address field is a 16-byte (128-bit) Internet address that identifies the original source of the datagram.
- **Destination address:** The destination address field is a 16-byte (128-bit) Internet address that usually identifies the final destination of the datagram. However, if source routing is used, this field contains the address of the next router.



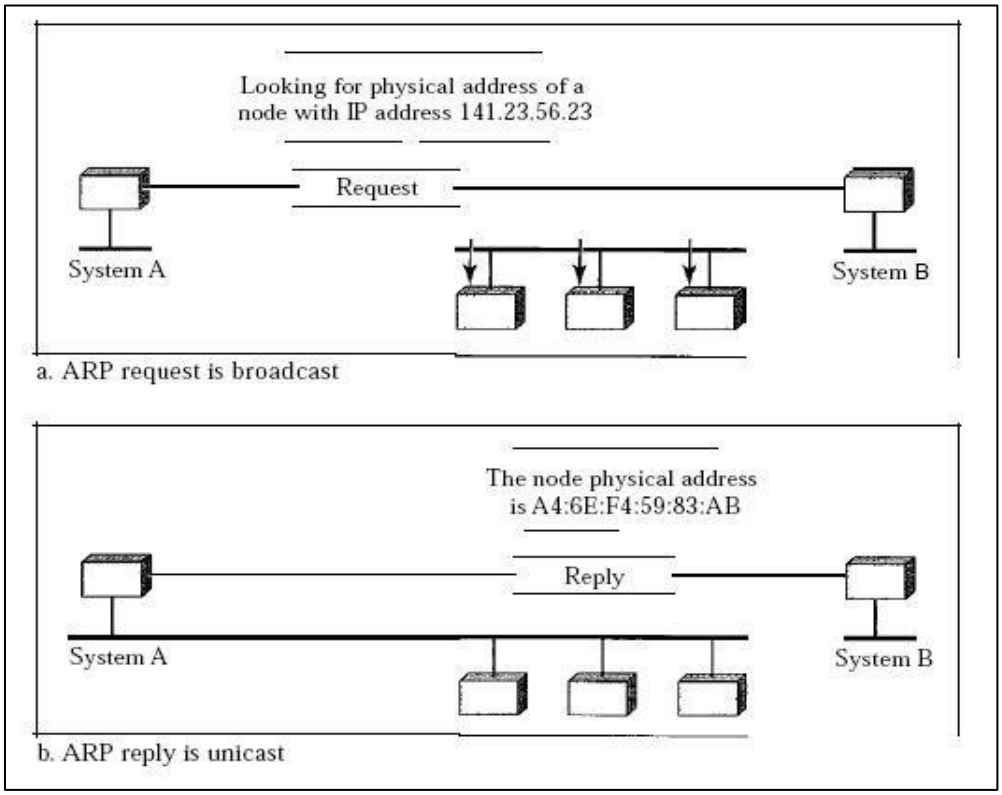
[Format of an IPv6 datagram]

Addressing

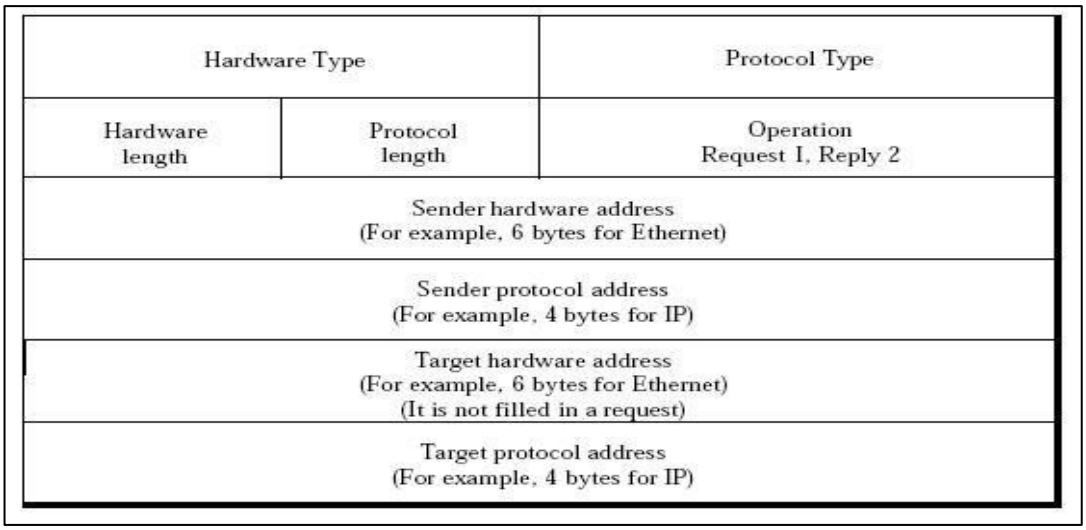
- Delivery of a packet to a host or a router requires two levels of addressing: logical and physical.
- We need to be able to map a logical address to its corresponding physical address and vice versa. These can be done by using either static or dynamic mapping.
- Static mapping involves a table that is stored in each machine on the network. Each machine that knows, for example, the IP address of another machine but not its physical address can look it up in the table.
- In dynamic mapping each time a machine knows one of the two addresses (logical or physical), it can use a protocol to find the other one.

Mapping Logical to Physical Address: ARP

Anytime a host or a router has an IP datagram to send to another host or router, it has the logical (IP) address of the receiver. The logical (IP) address is obtained from the DNS if the sender is the host or it is found in a routing table if the sender is a router. But the IP datagram must be encapsulated in a frame to be able to pass through the physical network. This means that the sender needs the physical address of the receiver. The host or the router sends an ARP query packet. The packet includes the physical and IP addresses of the sender and the IP address of the receiver. Because the sender does not know the physical address of the receiver, the query is broadcast over the network.



Every host or router on the network receives and processes the ARP query packet, but only the intended recipient recognizes its IP address and sends back an ARP response packet. The response packet contains the recipient's IP and physical addresses. The packet is unicast directly to the inquirer by using the physical address received in the query packet.



The fields are as follows:

- **Hardware type.** This is a 16-bit field defining the type of the network on which ARP is running. Each LAN has been assigned an integer based on its type. For example, Ethernet is given type 1. ARP can be used on any physical network.
- **Protocol type.** This is a 16-bit field defining the protocol. For example, the value of this field for the IPv4 protocol is 080016, ARP can be used with any higher-level protocol.
- **Hardware length.** This is an 8-bit field defining the length of the physical address in bytes. For example, for Ethernet the value is 6. o **Protocol length**. This is an 8-bit field defining the length of the logical address in bytes. For example, for the IPv4 protocol the value is 4. o **Operation.** This is a 16-bit field defining the type of packet. Two packet types are defined: ARP request (1) and ARP reply (2).
- **Sender hardware address.** This is a variable-length field defining the physical address of the sender. For example, for Ethernet this field is 6 bytes long.
- **Sender protocol address.** This is a variable-length field defining the logical (for example, IP) address of the sender. For the IP protocol, this field is 4 bytes long.
- **Target hardware address.** This is a variable-length field defining the physical address of the target. For example, for Ethernet this field is 6 bytes long. For an ARP request message, this field is alI 0s because the sender does not know the physical address of the target.

- **Target protocol address.** This is a variable-length field defining the logical (for example, IP) address of the target. For the IPv4 protocol, this field is 4 bytes long.

RARP

Reverse Address Resolution Protocol (RARP) finds the logical address for a machine that knows only its physical address. Each host or router is assigned one or more logical (IP) addresses, which are unique and independent of the physical (hardware) address of the machine. To create an IP datagram, a host or a router needs to know its own IP address or addresses. The IP address of a machine is usually read from its configuration file stored on a disk file.

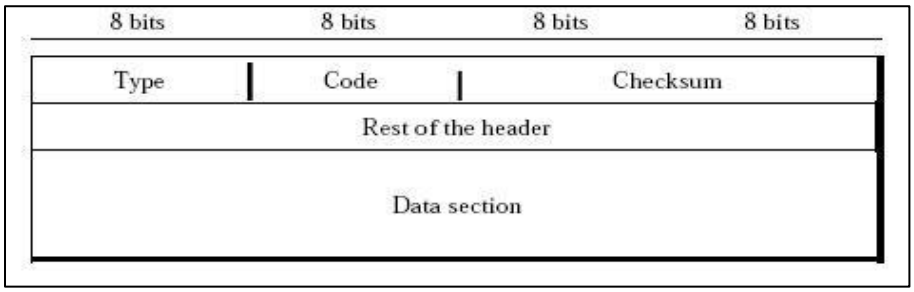
ICMP

The IP protocol is a best-effort delivery service that delivers a datagram from its original source to its final destination. However, it has two deficiencies: lack of error control and lack of assistance mechanisms.

The Internet Control Message Protocol (ICMP) has been designed to compensate for the above two deficiencies. It is a companion to the IP protocol.

Types of Messages

- ICMP messages are divided into two broad categories: error-reporting messages and query messages.
- The error-reporting messages report problems that a router or a host (destination) may encounter when it processes an IP packet.
- The query messages, which occur in pairs, help a host or a network manager get specific information from a router or another host.



An ICMP message has an 8-byte header and a variable-size data section. Although the general format of the header is different for each message type, the first 4 bytes are common to all.

One of the main responsibilities of ICMP is to report errors. Although technology has produced increasingly reliable transmission media, errors still exist and must be handled. Five types of errors are handled: destination unreachable, source quench, time exceeded, parameter problems, and redirection.

- **Destination Unreachable:** When a router cannot route a datagram or a host cannot deliver a datagram, the datagram is discarded and the router or the host sends a destination-unreachable message back to the source host that initiated the datagram.
- **Source-quench:** The source-quench message in ICMP was designed to add a kind of flow control to the IP. When a router or host discards a datagram due to congestion, it sends a sourcequench message to the sender of the datagram. This message has two purposes. First, it informs the source that the datagram has been discarded.
- Second, it warns the source that there is congestion somewhere in the path and that the source should slow down (quench) the sending process.
- **Time Exceeded:** When the datagram is discarded, a time-exceeded message must be sent by the router to the original source. Second, a time-exceeded message is also generated when not all fragments that make up a message arrive at the destination host within a certain time limit.
- **Parameter Problem:** Any ambiguity in the header part of a datagram can create serious problems as the datagram travels through the Internet. If a router or the destination host discovers an ambiguous or missing value in any field of the datagram, it discards the datagram and sends a parameter-problem message back to the source.
- **Redirection:** When a router needs to send a packet destined for another network, it must know the IP address of the next appropriate router. The same is true if the sender is a host. Both routers and hosts, then, must have a routing table to find the address of the router or the next router.

ICMPV6

The ARP and IGMP protocols in version 4 are combined in ICMPv6. Just as in ICMPv4, we divide the ICMP messages into two categories. However, each category has more types of messages than before.

Error Reporting: As we saw in our discussion of version 4, one of the main responsibilities of ICMP is to report errors. Five types of errors are handled: destination unreachable, packet too big, time exceeded parameter problems, and redirection.

- **Destination Unreachable:** The concept of the destination-unreachable message is exactly the same as described for ICMP version 4.
- **Packet Too Big:** This is a new type of message added to version 6. If a router receives a datagram that is larger than the maximum transmission unit (MTU) size of the network through which the datagram should pass, two things happen. First, the router discards the datagram and then an ICMP error packet-a packet-too-big message-is sent to the source.
- **Time Exceeded:** This message is similar to the one in version 4.
- **Parameter Problem:** This message is similar to its version 4 counterpart.
- **Redirection:** The purpose of the redirection message is the same as described for version 4.

Chapter 13

Transport Layer

Process-to-process Delivery

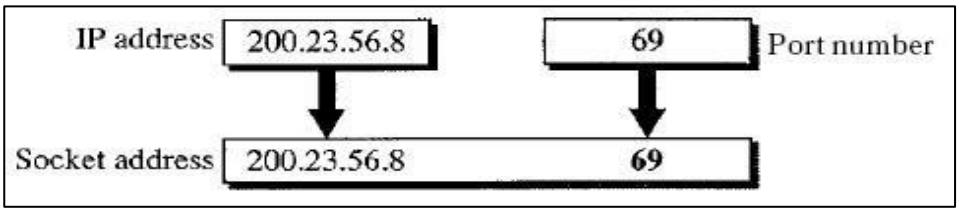
- The **data link layer** is responsible for delivery of frames between two neighboring nodes over a link. This is called **node-to-node delivery**.
- The **network layer** is responsible for delivery of datagram between two hosts. This is called **host-to-host delivery**.
- Real communication takes place between two processes (application programs) in a network. This is called process-to process delivery.
- The **transport layer** is responsible for **process-to-process delivery**-the delivery of a packet, part of a message, from one process to another.
- At the transport layer, we need a transport layer address, called a port number, to choose among multiple processes running on the destination host.
- In the Internet model, the port numbers are 16-bit integers between 0 and 65,535.
- The client program defines itself with a port number, chosen randomly by the transport layer software running on the client host. This is the ephemeral port number.
- The server process must also define itself with a port number. This port number, however, cannot be chosen randomly.

Socket Addresses

Process-to-process delivery needs two identifiers, IP address and the port number, at each end to make a connection.

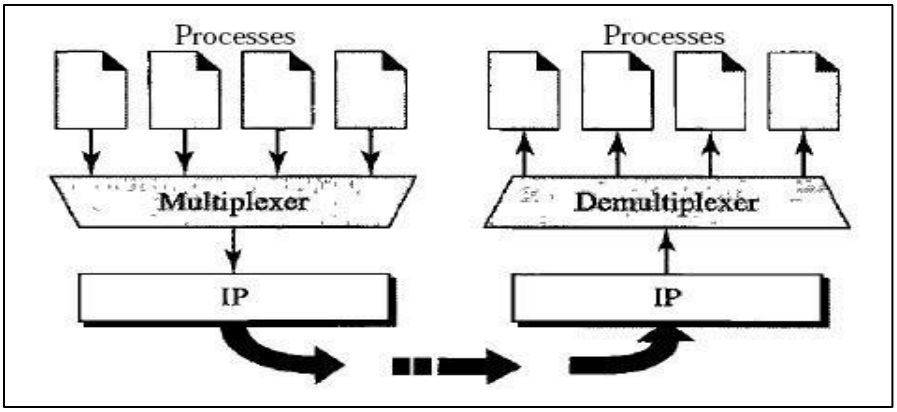
The combination of an IP address and a port number is called a socket address. The client socket address defines the client process uniquely just as the server socket address defines the server process uniquely.

A transport layer protocol needs a pair of socket addresses: the client socket address and the server socket address.



Multiplexing and Demultiplexing

The addressing mechanism allows multiplexing and demultiplexing by the transport layer.



Multiplexing

At the sender site, there may be several processes that need to send packets. However, there is only one transport layer protocol at any time. This is a many-to-one relationship and requires multiplexing. The protocol accepts messages from different processes, differentiated by their assigned port numbers.

After adding the header, the transport layer passes the packet to the network layer.

Demultiplexing

At the receiver site, the relationship is one-to-many and requires demultiplexing. The transport layer receives datagrams from the network layer. After error checking and dropping of the header, the transport layer delivers each message to the appropriate process based on the port number.

Connectionless Versus Connection-Oriented Service

A transport layer protocol can either be connectionless or connection-oriented.

- **Connectionless Service**

  In a connectionless service, the packets are sent from one party to another with no need for connection establishment or connection release. The packets are not numbered; they may be delayed or lost or may arrive out of sequence. There is no acknowledgment either. We will see shortly that one of the transport layer protocols in the Internet model, UDP, is connectionless.

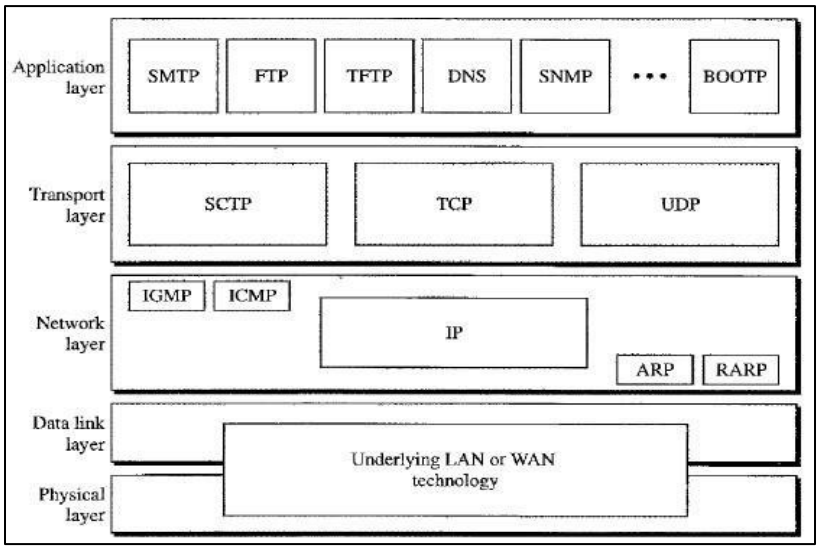
- **Connection Oriented Service**

  In a connection-oriented service, a connection is first established between the sender and the receiver. Data are transferred. At the end, the connection is released. We will see shortly that TCP and SCTP are connection-oriented protocols.

Reliable Versus Unreliable

- The transport layer service can be reliable or unreliable.
- If the application layer program needs reliability, we use a reliable transport layer protocol by implementing flow and error control at the transport layer. This means a slower and more complex service.

- On the other hand, if the application program does not need reliability because it uses its own flow and error control mechanism or it needs fast service or the nature of the service does not demand flow and error control (real-time applications), then an unreliable protocol can be used.



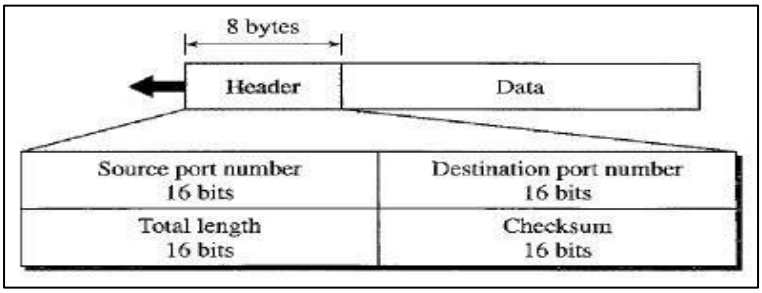
[Position of TCP and UDP]

User Datagram Protocol (UDP)

- The User Datagram Protocol (UDP) is called a connectionless, unreliable transport protocol. It does not add anything to the services of IP except to provide process-to-process communication instead of host-to-host communication. Also, it performs very limited error checking.
- If UDP is so powerless, why would a process want to use it? With the disadvantages come some advantages. UDP is a very simple protocol using a minimum of overhead. If a process wants to send a small message and does not care much about reliability, it can use UDP. Sending a small message by using UDP takes much less interaction between the sender and receiver than using TCP.

User Datagram

UDP packets, called user datagrams, have a fixed-size header of 8 bytes. Below figure shows the format of a user datagram.



[User Datagram Format]

The fields are as follows:

Source port number

- This is the port number used by the process running on the source host. It is 16 bits long, which means that the port number can range from 0 to 65,535.
- If the source host is the client (a client sending a request), the port number, in most cases, is an ephemeral port number requested by the process and chosen by the UDP software running on the source host.
- If the source host is the server (a server sending a response), the port number, in most cases, is a well-known port number.

Destination port number

- This is the port number used by the process running on the destination host. It is also 16 bits long.
- If the destination host is the server (a client sending a request), the port number, in most cases, is a well-known port number.
- If the destination host is the client (a server sending a response), the port number, in most cases, is an ephemeral port number. In this case, the server copies the ephemeral port number it has received in the request packet.

Length

This is a 16-bit field that defines the total length of the user datagram, header plus data. The 16 bits can define a total length of 0 to 65,535 bytes.

Checksum

This field is used to detect errors over the entire user datagram (header plus data).

UDP Operation

UDP uses concepts common to the transport layer.

- **Connectionless Services:** As mentioned previously, UDP provides a connectionless service. This means that each user datagram sent by UDP is an independent datagram. There is no relationship between the different user datagrams even if they are coming from the same source process and going to the same destination program. The user datagrams are not numbered. Also, there is no connection establishment and no connection termination, as is the case for TCP. This means that each user datagram can travel on a different path.
- **Flow and Error Control:** UDP is a very simple, unreliable transport protocol. There is no flow control and hence no window mechanism. The receiver may overflow with incoming messages. There is no error control mechanism in UDP except for the checksum. This means that the sender does not know if a message has been lost or duplicated. When the receiver detects an error through the checksum, the user datagram is silently discarded.

- **Encapsulation and Decapsulation:** To send a message from one process to another, the UDP protocol encapsulates and decapsulates messages in an IP datagram.
- **Queuing:**In UDP, queues are associated with ports. At the client site, when a process starts, it requests a port number from the operating system. Some implementations create both an incoming and an outgoing queue associated with each process. Other implementations create only an incoming queue associated with each process.

Use of UDP

The following lists some uses of the UDP protocol:

- UDP is suitable for a process that requires simple request-response communication with little concern for flow and error control.
- UDP is suitable for a process with internal flow and error control mechanisms. For example, the Trivial File Transfer Protocol (TFTP) process includes flow and error control
- UDP is a suitable transport protocol for multicasting. Multicasting capability is embedded in the UDP software but not in the TCP software.
- UDP is used for management processes such as SNMP.
- UDP is used for some route updating protocols such as Routing Information Protocol (RIP).

TCP

- TCP is called a connection-oriented, reliable transport protocol.
- It adds connection-oriented and reliability features to the services of IP.

TCP Services

Before we discuss TCP in detail, let us explain the services offered by TCP to the processes at the application layer.

- **Process-to-Process Communication**

Like UDP, TCP provides process-to-process communication using port numbers.

- **Stream Delivery Service**

TCP, unlike UDP, is a stream-oriented protocol. In UDP, a process (an application program) sends messages, with predefined boundaries, to UDP for delivery. UDP adds its own header to each of these messages and delivers them to IP for transmission. Each message from the process is calIed a user datagram and becomes, eventually, one IP datagram. Neither IP nor UDP recognizes any relationship between the datagrams.

TCP, on the other hand, allows the sending process to deliver data as a stream of bytes and allows the receiving process to obtain data as a stream of bytes. TCP creates an environment in which the two processes seem to be connected by an imaginary "tube" that carries their data across the Internet.

- **Full-Duplex Communication**

TCP offers full-duplex service, in which data can flow in both directions at the same time. Each TCP then has a sending and receiving buffer, and segments move in both directions.

- **Connection-Oriented Service**

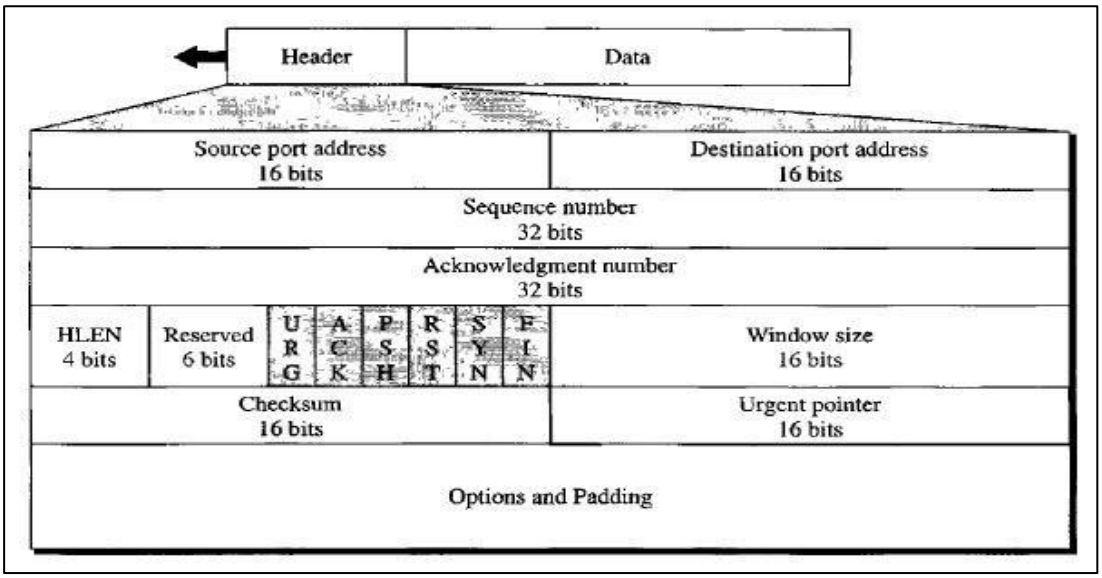
TCP, unlike UDP, is a connection-oriented protocol. When a process at site A wants to send and receive data from another process at site B, the following occurs:

1. The two TCPs establish a connection between them.
2. Data are exchanged in both directions.
3. The connection is terminated.

TCP Segment Format

The segment consists of a 20- to 60-byte header, followed by data from the application program. The header is 20 bytes if there are no options and up to 60 bytes if it contains options.

- **Source port address:** This is a 16-bit field that defines the port number of the application program in the host that is sending the segment. This serves the same purpose as the source port address in the UDP header.
- **Destination port address:** This is a 16-bit field that defines the port number of the application program in the host that is receiving the segment. This serves the same purpose as the destination port address in the UDP header.
- **Sequence number:** This 32-bit field defines the number assigned to the first byte of data contained in this segment. As we said before, TCP is a stream transport protocol. To ensure connectivity, each byte to be transmitted is numbered. The sequence number tells the destination which byte in this sequence comprises the first byte in the segment. During connection establishment, each party uses a random number generator to create an initial sequence number (ISN), which is usually different in each direction.



- **Acknowledgment number:** This 32-bit field defines the byte number that the receiver of the segment is expecting to receive from the other party. If the receiver of the segment has successfully received byte number *x* from the other party, it defines *x* + I as the acknowledgment number. Acknowledgment and data can be piggybacked together.

- **Header length:** This 4-bit field indicates the number of 4-byte words in the TCP header. The length of the header can be between 20 and 60 bytes. Therefore, the value of this field can be between 5 (5 x 4 =20) and 15 (15 x 4 =60).
- **Reserved:**This is a 6-bit field reserved for future use.
- **Control:** This field defines 6 different control bits or flags. One or more of these bits can be set at a time.

URG	The value of the urgent pointer field is valid.
ACK	The value of the acknowledgment field is valid.
PSH	Push the data.
RST	Reset the connection.
SYN	Synchronize sequence numbers during connection.
FIN	Terminate the connection

- **Window size:** This field defines the size of the window, in bytes, that the other party must maintain.
- **Checksum:**
- This 16-bit field contains the checksum. The calculation of the checksum for TCP follows the same procedure as the one described for UDP. However, the inclusion of the checksum in the UDP datagram is optional, whereas the inclusion of the checksum for TCP is mandatory.
- **Urgent pointer:** This l6-bit field, which is valid, only if the urgent flag is set, is used when the segment contains urgent data. It defines the number that must be added to the sequence number to obtain the number of the last urgent byte in the data section of the segment.
- **Options:** There can be up to 40 bytes of optional information in the TCP header.

Congestion Control and Quality of Service

Congestion control and quality of service are two issues so closely bound together that improving one means improving the other and ignoring one usually means ignoring the other. Most techniques to prevent or eliminate congestion also improve the quality of service in a network.

Congestion

- An important issue in a packet-switched network is **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.

QUALITY OF SERVICE 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. However, the sensitivity of application programs to reliability is not the same. For example, it is more important that electronic mail, file transfer, and Internet access have reliable transmissions than telephony or audio conferencing.
- **Delay:** Source-to-destination delay is another flow characteristic. Again applications can tolerate delay in different degrees. In this case, telephony, audio conferencing, video conferencing, and remote log-in need minimum delay, while delay in file transfer or e-mail is less important.
- **Jitter:** Jitter is the variation in delay for packets belonging to the same flow. For example, if four packets depart at times 0, 1, 2, 3 and arrive at 20, 21, 22, 23, all have the same delay, 20 units of time. On the other hand, if the above four packets arrive at 21, 23, 21, and 28, they will have different delays: 21, 22, 19, and 24. Jitter is defined as the variation in the packet delay. High jitter means the difference between delays is large; low jitter means the variation is small.
- **Bandwidth:** Different applications need different bandwidths. In video conferencing we need to send millions of bits per second to refresh a color screen while the total number of bits in an email may not reach even a million.

Chapter 14

Application Layer: DNS

- The application layer consists of various applications. Out of those one is DNS, which stands for **Domain Name System**. The very first question arise: 'what is the need of this application?'.
- To begin with let's start with a real world example. There are many identifiers to be a unique person in the world, such as SSN, name, and Passport number along with the county who issued it, etc. In the similar fashion, every computer or host and router in the world has a unique identifying 32-bit 'IP' address. Say if we need some information that is on other part of the world. We need to know the IP address of that machine.
- Remembering IP addresses is difficult, as it contains all numbers. To remember IP addresses of more than one host becomes cumbersome. Therefore a name has been assigned to almost every IP address which makes it easier for humans to remember.
- DNS provides mapping of **IP address and Domain name.**

DNS Services

1. Host name to IP address translation

The primary purpose of DNS is to provide translation of host name to IP address and vice versa. The backward facility (translating IP address to domain name) is known as Reverse DNS.

2. Host aliasing

Host aliasing is referred to another name given to the same machine on the network. It is used because a hostname may have a complicated name instead of that a simple term may be used.

3. Mail server aliasing

It is highly desirable that an email address should contain simple letters, or should be something that can be easy to remember. E.g. richard@gmail.com can be remembered easily but if the original mail server address, say la4.mail1.google.com, were to be used it would be difficult to remember.

4. Load distribution

A set of IP address is provided to one canonical name which prevents the load to be present only on one server. "When the request comes to the DNS server to resolve the domain name, it gives out one of the several canonical names in a rotated order. This redirects the request to one of the several servers in a server group. Once the BIND feature of DNS resolves the domain to one of the servers, subsequent requests from the same client is sent to the same server."

Problems that arise when we try to centralize DNS:

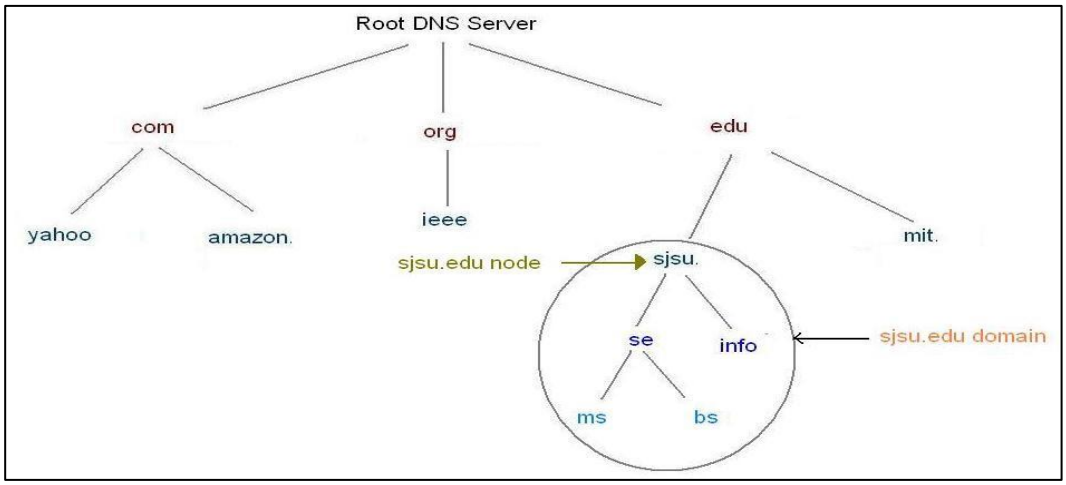
1. Single point of failure

2. Increase in traffic volume

3. Distant centralized database

4. Maintenance

As centralized DNS does not scale because of the reasons mentioned above, a need arose to implement DNS in a distributed manner. The DNS is a distributed system, implemented in a hierarchy of many name servers. The decentralized administration is achieved through delegation.



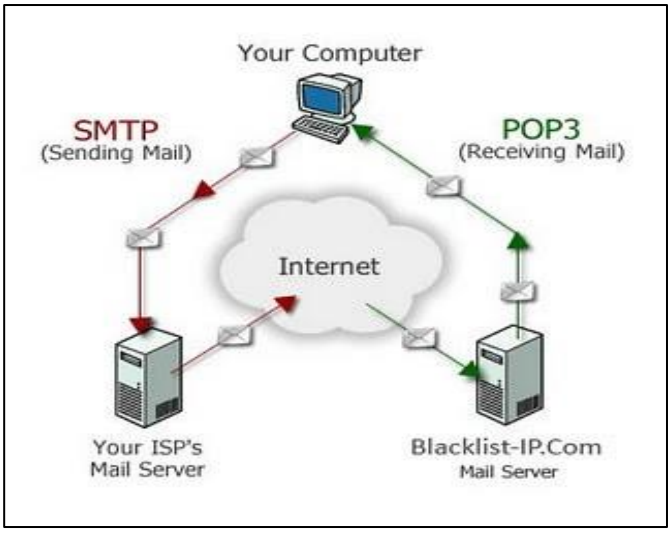
A domain may contain many sub-domains inside it. To identify if domain is a sub-domain of another domain, you need to compare the domain name with its parent domain name. E.g. se.sjsu.edu is a subdomain of the sjsu.edu domain. The other way to determine the sub-domains is through looking at the levels of the tree.

Simple Mail Transfer Protocol (SMTP)

Simple Mail Transfer Protocol (**SMTP**) is an Internet standard for electronic mail (e-mail) transmission across Internet Protocol (IP) networks.

SMTP is a connection-oriented, text-based protocol in which a mail sender communicates with a mail receiver by issuing command strings and supplying necessary data over a reliable ordered data stream channel, typically a Transmission Control Protocol (TCP) connection. An SMTP session consists of commands originated by an SMTP client (the initiating agent, sender, or transmitter) and corresponding responses from the SMTP server (the listening agent, or receiver) so that the session is opened, and session parameters are exchanged. A session may include zero or more SMTP transactions. An SMTP transaction consists of three command/reply sequences (see example below.) They are:

1. **MAIL** command, to establish the return address, a.k.a. Return-Path, 5321.From, mfrom, or envelope sender. This is the address for bounce messages.
2. **RCPT** command, to establish a recipient of this message. This command can be issued multiple times, one for each recipient. These addresses are also part of the envelope.
3. **DATA** to send the message text. This is the content of the message, as opposed to its envelope. It consists of a message header and a message body separated by an empty line. DATA is actually a group of commands, and the server replies twice: once to the DATA command proper, to acknowledge that it is ready to receive the text, and the second time after the end-of-data sequence, to either accept or reject the entire message.

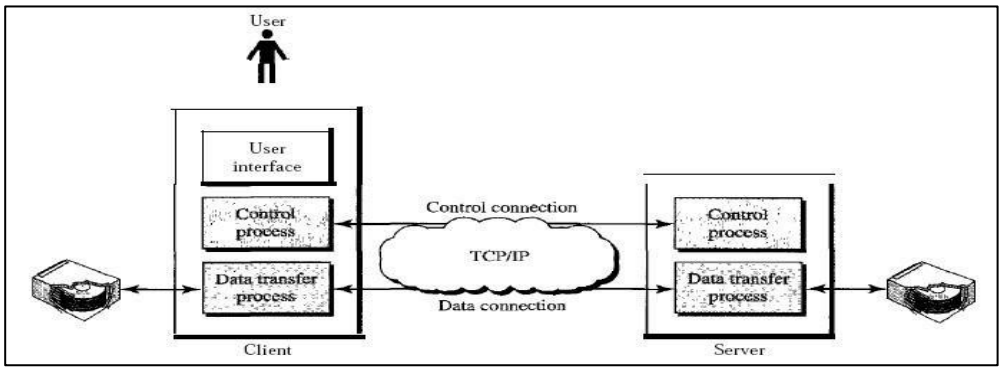


FTP

File Transfer Protocol (FTP)

- File Transfer Protocol (FTP) is the standard mechanism provided by TCP/IP for copying a file from one host to another. Although transferring files from one system to another seems simple and straightforward, some problems must be dealt with first. For example, two systems may use different file name conventions. Two systems may have different ways to represent text and data. Two systems may have different directory structures. All these problems have been solved by FTP in a very simple and elegant approach.
- FTP differs from other client/server applications in that it establishes two connections between the hosts. One connection is used for data transfer, the other for control information (commands and responses). Separation of commands and data transfer makes FTP more efficient. The control connection uses very simple rules of communication.
- We need to transfer only a line of command or a line of response at a time. The data connection, on the other hand, needs more complex rules due to the variety of data types transferred. However, the difference in complexity is at the FTP level, not TCP.

- For TCP, both connections are treated the same.FTP uses two well-known TCP ports: Port 21 is used for the control connection, and port 20 is used for the data connection.



WWW

- The **World Wide Web** (WWW) is a repository of information linked together from points all over the world. The WWW has a unique combination of flexibility, portability, and user-friendly features that distinguish it from other services provided by the Internet.
- Each site holds one or more documents, referred to as *Web pages.* Each Web page can contain a link to other pages in the same site or at other sites. The pages can be retrieved and viewed by using browsers.

Client (Browser)

A variety of vendors offer commercial browsers that interpret and display a Web document, and all use nearly the same architecture. Each browser usually consists of three parts: a controller, client protocol, and interpreters. The controller receives input from the keyboard or the mouse and uses the client programs to access the document. After the document has been accessed, the controller uses one of the interpreters to display the document on the screen. The client protocol can be one of the protocols described previously such as FTP or HTTP (described later in the chapter). The interpreter can be HTML, Java, or JavaScript, depending on the type of document.

Server

The Web page is stored at the server. Each time a client request arrives, the corresponding document is sent to the client. To improve efficiency, servers normally store requested files in a cache in memory; memory is faster to access than disk. A server can also become more efficient through multithreading or multiprocessing. In this case, a server can answer more than one request at a time.

Uniform Resource Locator

- A client that wants to access a Web page needs the address. To facilitate the access of documents distributed throughout the world, HTTP uses locators. The uniform resource locator (URL) is a standard for specifying any kind of information on the Internet. The URL defines four things: protocol, host computer, port, and path.

- The protocol is the client/server program used to retrieve the document. Many different protocols can retrieve a document; among them are FTP or HTTP. The most common today is HTTP.

Protocol://host:port/path

HTTP

- The Hypertext Transfer Protocol (HTTP) is a protocol used mainly to access data on the World Wide Web. HTTP functions as a combination of FTP and SMTP. It is similar to FTP because it transfers files and uses the services of TCP. However, it is much simpler than FTP because it uses only one TCP connection. There is no separate control connection; only data are transferred between the client and the server.
- HTTP is like SMTP because the data transferred between the client and the server look like SMTP messages. In addition, the format of the messages is controlled by MIME-like headers.
- Unlike SMTP, the HTTP messages are not destined to be read by humans; they are read and interpreted by the HTTP server and HTTP client (browser). SMTP messages are stored and forwarded, but HTTP messages are delivered immediately. The commands from the client to the server are embedded in a request message. The contents of the requested file or other information are embedded in a response message. HTTP uses the services of TCP on well-known port 80.