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M.Sc. CYBER SECURITY
Second Semester

Computer Networks



COMPUTER NETWORKS

COURSE BLOCK -1

Preface

With clear, straightforward text and engaging examples, this book brings an active style of learning to the study of computer networking. It can be used as a primary textbook to accompany lecture material or as a companion text to provide hands-on assignments. It is also an ideal guide to self-study for the computing professional. In fact, this book can help any interested readers look under the hood of the network they use every day and become a more informed network consumer.

The book consists of a set of exercises, each of which involves analyzing traces of actual network activity. Important concepts are presented in the context of actual traces of real-world scenarios. Readers learn the details of networking protocols in the best ways possible—by seeing them in action!

Well-chosen examples make it clear how the details of networking protocols are relevant to everyday life. For example, security issues are highlighted throughout the book. Readers will see what is sent over the network when they browse the web or shop online. They will see what a home wireless network looks like to someone driving by if WEP is not enabled. These and many more concrete illustrations help readers understand how to secure their networks against attack.

Anywhere people are learning about computer networking, this book will help them learn by doing rather than simply hearing information. The material contained in this book has been used Master's level courses

UNIT–I: INTRODUCTION TO COMPUTER NETWORKS

Structure

- 1.0 Objectives
- 1.1 Introduction
- 1.2 Basics of computer networks
- 1.3 Components of computer network
- 1.4 Direction of data flow
- 1.5 Networks
- 1.6 Network criteria
- 1.7 Summary
- 1.8 Keywords
- 1.9 Questions
- 1.10 Reference

1.0 OBJECTIVES

After going through this lesson you will be able to

- discuss basics of computer networks D
- describe components of computer networks D
- elucidate direction of data flow E
- discuss networks D
- explain network criteria E

1.1 INTRODUCTION

A computer network is a collection of two or more computer systems that are linked together. A network connection can be established using either cable or wireless media. Hardware and software are used to connect computers and tools in any network.

A computer network consists of various kinds of nodes. Servers, networking hardware, personal computers, and other specialized or general-purpose hosts can all be nodes in a computer network. Hostnames and network addresses are used to identify them.

1.2 BASICS OF COMPUTER NETWORKS

The term 'Network' means a group, chain or a collection of something that come together for the purpose of communication. In computer's reference – the same definition holds value but there is a slight change which is – A network is a group of related computers that exchange resources, files and possess some kind of communications within themselves.

The communication word in this reference means 'DATA'. Since these activities are going in between the computers they are called as Computer Networks.

Definition: A computer network is a group of devices connected with each other through a transmission medium such as wires, cables etc. These devices can be computers, printers, scanners, Fax machines etc. The purpose of having computer network is to send and receive data stored in other devices over the network. These devices are often referred as nodes.

ADVANTAGES

Many organizations already have a substantial number of computers, often located far apart. For example, a company with many offices may have a computer at each location to keep track of customer orders, monitor sales, and do the local payroll. Previously, each of these computers may have worked in isolation from others but at some point, management decided to connect them together in a network. In general, we can refer to it as:

Resource Sharing:

The aim is to make all programs, data and peripherals available to anyone on the network irrespective of the physical location of the resources and the user.

Reliability: A file can have copies on two or three different machines, so if one of them is unavailable (hardware crash), the other copies could be used. For military, banking, air reservation and many other applications it is of great importance.

Cost Factor: Personal computers have better price/performance

ratio than microcomputers. So it is better to have PC's, one per user, with data stored on one shared file server machine.

Communication Medium: Using a network, it is possible for managers, working far apart, to prepare financial reports of the company. The changes at one end can be immediately noticed at another and hence it speeds up co-operation among them.

ELEMENTARY TERMINOLOGY OF NETWORKS

It is sometimes to learn about the components/terms mostly used in networking. Whenever we talk about a network it includes the hardware and the software that make up the network. Now let us have a look at some typical hardware components of network.

Nodes (Workstations)

The term nodes refer to the computers that are attached to a network and are seeking to share the resource of the network. Of course, if there were no nodes (also called workstations), there would be no network at all.

A computer becomes a workstation of a network as soon as it is attached to a network.

Server

Def: A computer that facilitates "the sharing of data" software" and hardware resources (e.g. "printer's" modem etc.) on the network" is termed as a SERVER.

On small networks, sometimes, all the shareable stuff (like files, data, software etc.) is stored on the server. A network can have more than one server also. Each server has a unique name on the network and all users of network identify the server by its unique name. Servers can be of two types: non-dedicated and dedicated servers.

Non-

dedicated Servers: On small networks, a workstation that can double up as a server, is known as non-dedicated servers since it is not completely dedicated to the cause of serving. Such servers can facilitate the resource-sharing among workstations on a proportionately smaller scale. Since one computer works as a workstation as well as a server, it is slower and requires more memory. The (small) networks using such a server are known as peer-to-peer networks.

Dedicated Servers: On bigger network installations, there is a computer reserved for server's job and its only job is to help workstations access data, software and hardware resources. It does not double

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up as a workstation and such a server is known as a dedicated server. The network using such a server is known as a master-slave network.

On a network, there may be several servers that allow workstations to share specific resources. For example, there may be a server exclusively for serving files-related requests like storing files, deciding about their access privileges and regulating the amount of space allowed for each user. This server is known as a file server. Similarly, there may be a printer server and a modem server. The printer server takes care of the printing requirements of a number of workstations and the modem server helps a group of network users use a modem to transmit long distance messages.

Network Interface Unit (NIU)

Def: A NETWORK INTERFACE UNIT is an interpreter that helps to establish communication between the server and workstations.

A standalone computer (a computer that is not attached to a network) lives in its own world and carries out its tasks with its own in-built resources. But as soon as it becomes a workstation, it needs an interface to help establish a connection with the network because without this, the workstation will not be able to share network resources.

The network-interface-unit is a device that is attached to each of the workstations and the server, and helps the workstation to establish the all-important connection with the network. Each network-interface-unit that is attached to a workstation has a unique number identifying it which is known as the node address. The NIU is also called Terminal Access Point (TAP). Different manufacturers have different names for the interface.

Computer networks can be used for numerous services, both for companies and for individuals. For companies, networks of personal computers using shared servers often provide access to corporate information. Typically, they follow the client-server model, with client workstations on employee desktops accessing powerful servers in the main room. For individuals, networks offer access to a variety of information and entertainment resources. Individuals often access the Internet by calling up an ISP using a modem, although increasingly many people have a fixed connection at home. An up-and-coming area is wireless networking with new applications such as mobile e-mail access and m-commerce.

Applications & Uses of Networks

In the short time they have been around, data communication networks have become an indispensable part of business, industry, and entertainment. Some of the network applications in different fields are the following:

Marketing and sales. Computer networks are used extensively in both marketing and sales organizations. Marketing professionals use them to collect, exchange, and analyze data relating to customer needs and product development cycles. Sales applications include telemarketing, which uses order-entry computers or telephones connected to an order-processing network, and on-line reservations services for hotels, airlines, and so on.

Financial services. Today's financial services are totally dependent on computer networks. Applications include credit history searches, foreign exchange and investment services, and electronic funds transfer (EFT), which allows a user to transfer money without going into a bank (an automated teller machine is a kind of electronic funds transfer; automatic paycheck deposit is another).

Manufacturing. Computer networks are used today in many aspects of manufacturing, including the manufacturing process itself. Two applications that use networks to provide essential services are computer-assisted design (CAD) and computer-assisted manufacturing (CAM), both of which allow multiple users to work on a project simultaneously.

Electronic messaging: Probably the most widely used network application is electronic mail (e-mail).

Directory services: Directory services allow lists of files to be stored in a central location to speed worldwide search operations.

Information services: Network information services include bulletin boards and data banks. A World Wide Web site offering the technical specifications for a new product is an information service.

Electronic data interchange (EDI): EDI allows business information (including documents such as purchase orders and invoices) to be transferred without using paper.

Teleconferencing: Teleconferencing allows conferences to occur without the participants being in the same place. Applications include simple text conferencing (where participants communicate through their keyboards and computer monitors), voice conferencing (where participants at a number of locations communicate simultaneously over the phone) and video conferencing (where participants can see as well as talk to one another).

Cellular telephone: In the past two parties wishing to use the services of the telephone company had to be linked by a fixed physical connection. Today's cellular networks make it possible to maintain wireless phone connections even while traveling over large distances.

Cable television: Future services provided by cable television networks may include video on request, as well as the same information, financial, and communications services currently provided by the telephone companies and computer networks.

History (Development) of Computer Networks

Each of the past three centuries has been dominated by a single technology. The 18th century was the era of the great mechanical systems accompanying the Industrial Revolution. The 19th century was the age of the steam engine. During the 20th century, the key technology was information gathering, processing, and distribution. Among other developments, we saw the installation of worldwide telephone networks, the invention of radio and television, the birth and unprecedented growth of the computer industry, and the launching of communication satellites.

As a result of rapid technological progress, these areas are rapidly converging and the differences between collecting, transporting, storing, and processing information are quickly disappearing. Organizations with hundreds of offices spread over a wide geographical area routinely expect to be able to examine the current status of even their most remote output at the push of a button. As our ability to gather, process and distribute information grows, the demand for ever more sophisticated information processing grows even faster.

Although the computer industry is still young compared to other industries (e.g., automobiles and air transportation), computers have made spectacular progress in a short time. During the first two decades of their existence, computer systems were highly centralized, usually within a single large room. Not infrequently, this room had glass walls, through which visitors could gawk at the great electronic wonder inside. A medium-sized company or university might have had one or two computers, while large institutions had at most a few dozens. The idea that within twenty years equally powerful computers smaller than postage stamps would be mass produced by the millions was pure science fiction.

The merging of computers and communications has had a profound influence on the way computer systems are organized. The concept of the "computer center" as a room with a large computer to which users bring

their work for processing is now totally obsolete. The old model of a single computer serving all of the organization's computational needs has been replaced by one in which a large number of separate but inter-connected computers do the job. These systems are called computer networks. The design and organization of these networks are the subjects of this book.

Throughout the book we will use the term "computer network" to mean a collection of autonomous computers interconnected by a single technology. Two computers are said to be interconnected if they are able to exchange

information. The connection need not be via a copper wire; fiber optics, microwaves, infrared, and communication satellites can also be used. Networks come in many sizes, shapes and forms, as we will see later. Although it may sound strange to some people, neither the Internet nor the World Wide Web is a computer network. By the end of this book, it should be clear why. The quick answer is - the Internet is not a single network but a network of networks and the Web is a distributed system that runs on top of the Internet.

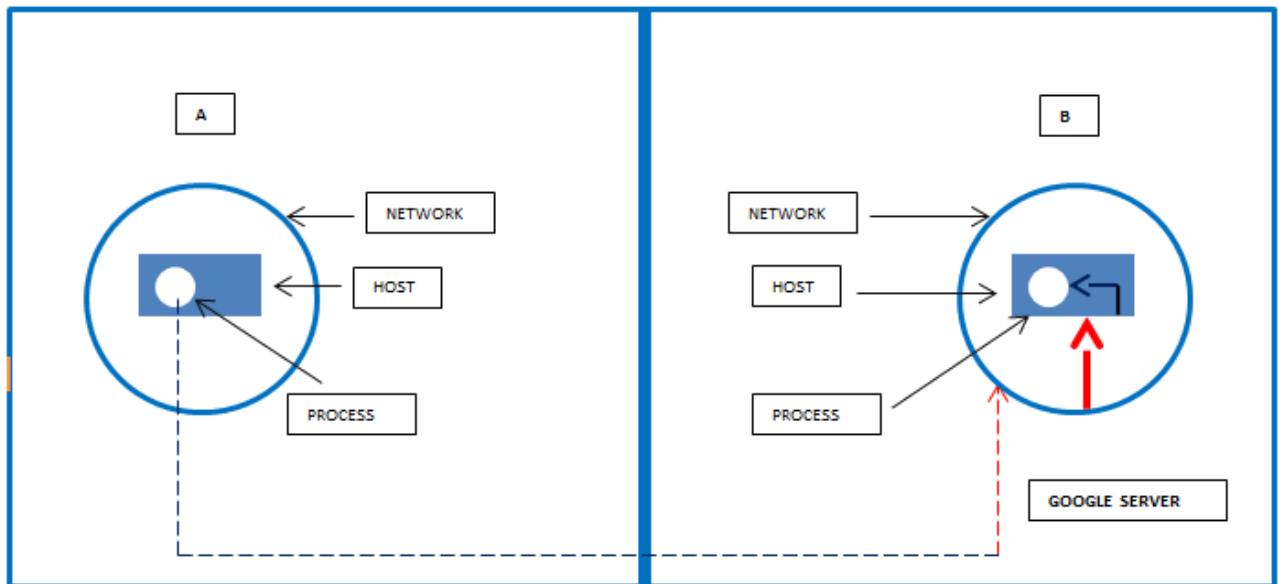
There is considerable confusion in the literature, between a computer network and a distributed system. The key distinction is that in a distributed system, a collection of independent computers appears to its users as a single coherent system. Usually, it has a single model or paradigm that it presents to the users. Often a layer of software on top of the operating system, called middleware, is responsible for implementing this model. A well-known example of a distributed system is the World Wide Web, in which everything looks like a document (Web page).

In a computer network, this coherence, model, and software are absent. Users are exposed to the actual machines, without any attempt by the system to make the machines look and act in a coherent way. If the machines have different hardware and different operating systems, that is fully visible to the users. If a user wants to run a program on a remote machine, he has to log on to that machine and run it there.

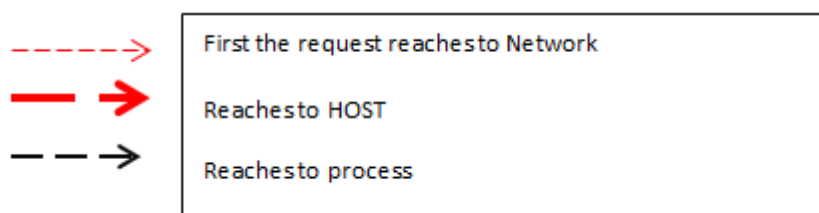
In effect, a distributed system is a software system built on top of a network. The software gives it a high degree of cohesiveness and transparency. Thus, the distinction between a network and a distributed system lies with the software (especially the operating system), rather than with the hardware.

Data Communication: When we communicate, we are sharing information. This sharing can be local or remote. Between individuals, local communication usually occurs face to face, while remote communication takes place over distance.

Let's see an example of a Computer Network for overall understanding, this explanation will make readers understand how things work



Within a network, there is a host (many hosts can be there) and within a host, there is a process (many processes can be there).



So the process is Mr. Ron wants to open www.google.com (domain name) on his web browser. He has to connect to the Google server to get the web page on his system. See the dotted lines that emerge from A and reaches to B (image description is available).

Now with using only the domain name, we have to identify the network, the host and the process which is the entire thing we will see happening.

The domain name (i.e. www.google.com) must be converted to IP address so that it can be understood by the receiving host and networks.

So, IP address has 2 parts

1. HOST ID
2. NETWORK ID.

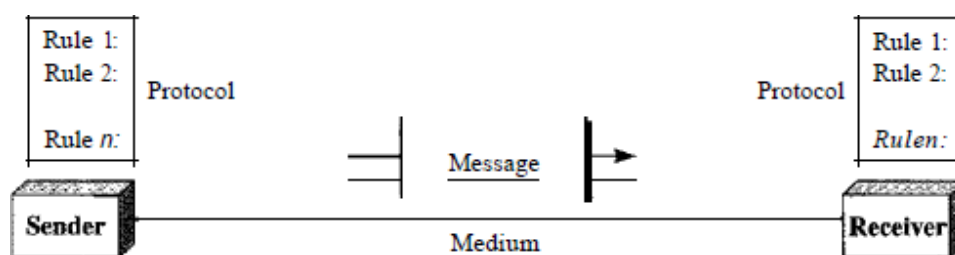


So using the network ID we can reach the target network and using the Host ID we can reach the Target host. After this, we need Port Number to reach the process. This is how the connection works for web services.

1.3 COMPONENTS OF COMPUTER NETWORK

Components: There are five basic components of a computer network

A data communication system has five components.



1. Message. The message is the information (data) to be communicated. Popular forms of information include text, numbers, pictures, audio, and video.
2. Sender. The sender is the device that sends the data message. It can be a computer, workstation, telephone handset, video camera, and so on.
3. Receiver. The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and so on.

4. 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

5. 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 Representation:

Information today comes in different forms such as text, numbers, images, audio, and video.

Text: In data communications, text is represented as a bit pattern, a sequence of bits (0s or 1s). Different sets of bit patterns have been designed to represent text symbols. Each set is called a code, and the process of representing symbols is called coding. Today, the prevalent coding system is called Unicode, which uses 32 bits to represent a symbol or character used in any language in the world. The American Standard Code for Information Interchange (ASCII), developed some decades ago in the United States, now constitutes the first 127 characters in Unicode and is also referred to as Basic Latin.

Numbers: Numbers are also represented by bit patterns. However, a code such as ASCII is not used to represent numbers; the number is directly converted to a binary number to simplify mathematical operations. Appendix B discusses several different numbering systems.

Images: Images are also represented by bit patterns. In its simplest form, an image is composed of a matrix of pixels (picture elements), where each pixel is a small dot. The size of the pixel depends on the *resolution*. For example, an image can be divided into 1000 pixels or 10,000 pixels. In the second case, there is a better representation of the image (better resolution), but more memory is needed to store the image. After an image is divided into pixels, each pixel is assigned a bit pattern. The size and the value of the pattern depend on the image. For an image made of only black and white dots (e.g., a chessboard), a 1-bit pattern is enough to represent a pixel. If an image is not made of pure white and pure black pixels, you can increase the size of the bit pattern to include gray scale. For example, to show

four levels of gray scale, you can use 2-bit patterns. A black pixel can be represented by 00, a dark gray pixel by 01, a light gray pixel by 10, and a white pixel by 11. There are several methods to represent color images. One method is called RGB, so called because each color is made of a combination of three primary colors: *red*, green, and blue. The intensity of each color is measured, and a bit pattern is assigned to it. Another method is called YCM, in which a color is made of a combination of three other primary colors: yellow, cyan, and magenta.

Audio: Audio refers to the recording or broadcasting of sound or music. Audio is by nature different from text, numbers, or images. It is continuous, not discrete. Even when we use a microphone to change voice or music to an electric signal, we create a continuous signal. In Chapters 4 and 5, we learn how to change sound or music to a digital or an analog signal.

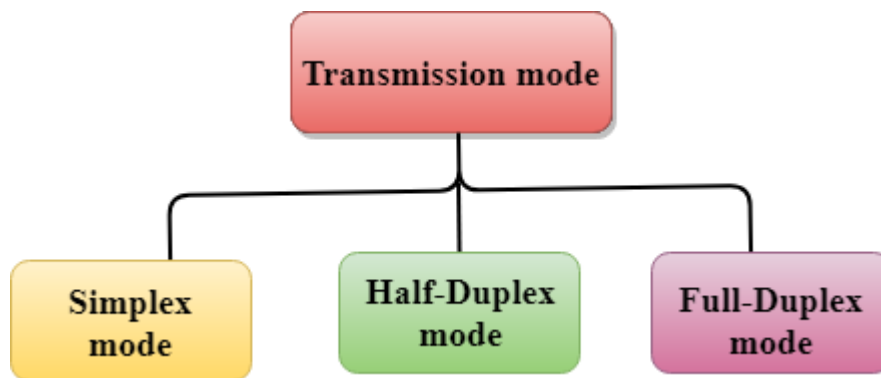
Video: Video refers to the recording or broadcasting of a picture or movie. Video can either be produced as a continuous entity (e.g., by a TV camera), or it can be a combination of images, each a discrete entity, arranged to convey the idea of motion. Again we can change video to a digital or an analog signal.

1.4 DIRECTION OF DATA FLOW

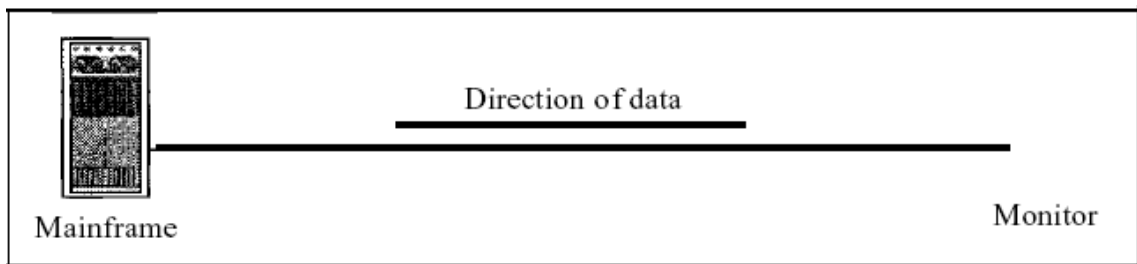
- The way in which data is transmitted from one device to another device is known as **transmission mode**.
- The transmission mode is also known as the communication mode.
- Each communication channel has a direction associated with it, and transmission media provide the direction. Therefore, the transmission mode is also known as a directional mode.
- The transmission mode is defined in the physical layer.

The Transmission mode is divided into three categories:

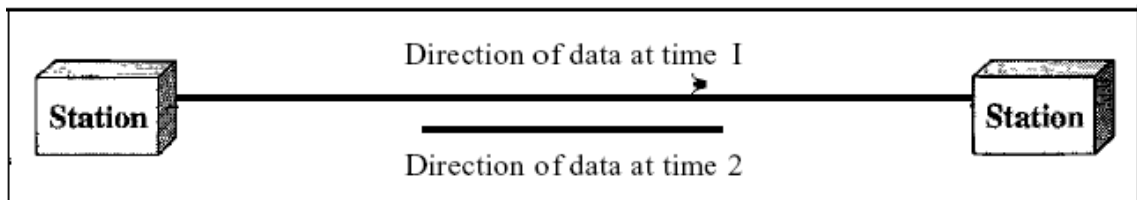
- Simplex mode
- Half-duplex mode
- Full-duplex mode



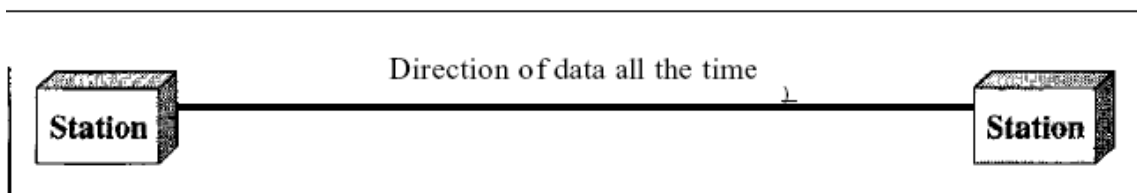
Communication between two devices can be simplex, half-duplex, or full-duplex as shown in figure:



a. Simplex



b. Half-duplex



c. 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 (see Figure a).

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. Walkie-talkies and CB (citizens band) radios are both half-duplex systems.

The half-duplex mode is used in cases where there is no need for communication in both directions at the same time; the entire capacity of the channel can be utilized for each direction.

Full-Duplex:

In full-duplex both stations can transmit and receive simultaneously (see Figure c). The full-duplex mode is like a two-way street with traffic flowing in both directions at the same time. In full-duplex mode, signals going in one direction share the capacity of the link; with signals going in the other direction. This sharing can occur in two ways: Either the link must contain two physically separate transmission paths, one for sending and the other for receiving; or the capacity of the channel is divided between signals traveling in both directions. 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. The full-duplex mode is used when communication in both directions is required all the time. The capacity of the channel, however, must be divided between the two directions.

1.5 NETWORKS

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 computers (usually a personal computer or workstation) handle a subset.

1.6 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 transit time and response time.

Transit 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 throughput and less delay. However, these two criteria are often contradictory. If we try to send more data to the network, we may increase throughput but we increase the delay because of traffic congestion in the network.

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, and the network's robustness in a catastrophe.

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.

1.7 SUMMARY

In this unit we have discussed about basics of computer networks. We also explained components of computer networks and direction of data flow. At the end of this unit reader are able to figure out the concept of networks and network criteria.

1.8 KEYWORDS

Computer networks, Simplex, Half-duplex, Full-duplex, Sender, Receiver, NIU, Server.

1.9 QUESTIONS

1. Define computer networks. Write the advantages of computer networks.
2. Briefly explain direction of data flow.
3. Explain components of computer network.
4. Write a short note on network criteria.
5. Discuss advantages of computer networks.

1.10 REFERENCES

- William Stallings, Data and Computer Communication, Prentice Hall of India
- Behrouz A. Forouzan, Data Communication and Networking, McGraw-Hill
- Andrew S Tanenbaum, Computer Networks, Prentice Hall.

UNIT 2: TYPES OF NETWORKS AND REFERENCE MODELS

Structure:

2.0 Objectives

2.1 Introduction

2.2 Physical structure

2.3 Categories of Network

2.4 Protocols and Standards

2.5 Reference Models

2.5.1 OSI Reference Model

2.5.2 TCP/IP Reference Model

2.5.3 Comparative Study

2.6 Summary

2.7 Keywords

2.8 Questions

2.9 Reference

2.0

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OBJECTIVES

After going through this lesson you will be able to

- lucidate Physical structures E
- Discuss Categories of network D
- Describe Protocols and standards D
- lucidate Reference models such as OSI Reference Model and TCP/IP Reference Model. Also discuss comparative study. E

2.1 INTRODUCTION

In this unit we will learn about the different types of networks, their applications and networking models. We will also examine reference models, its various layers and functions of each layer. The networking model describes the architecture, components, and design used to establish communication between the source and destination systems. Aliases for network models include protocol stacks, protocol suites, network stacks, and network protocols. There are 2 predominant models available. Let us take a look at them

1. Open Systems Interconnection (OSI) Model
2. Transmission Control Protocol/Internet Protocol (TCP/IP) Model

2.2 PHYSICAL STRUCTURES

Type of Connection

A network is two or more devices connected through links. A link is a communication 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. For communication to occur, two devices must be connected in some way to the same link at the same time. There are two possible types of connections: point-to-point and multipoint.

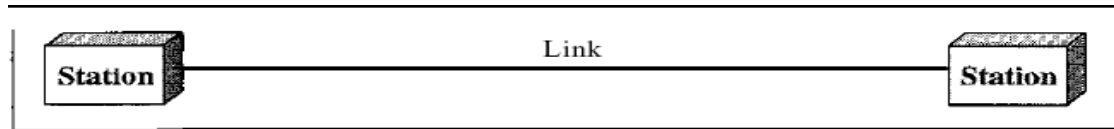
Point-to-Point

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. Most point-to-point connections use an actual length of wire or cable to connect the two ends, but other options, such as microwave or satellite links, are also possible. When you change television channels by infrared remote control, you are establishing a point-to-point connection between the remote control and the television's control system.

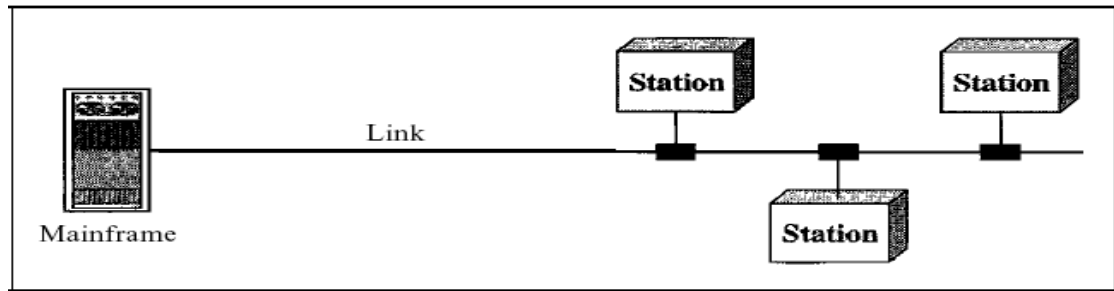
Multipoint

A multipoint (also called multidrop) 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. If several devices can use the link

simultaneously, it is a *spatially shared connection*. If users must take turns, it is a *time shared connection*.



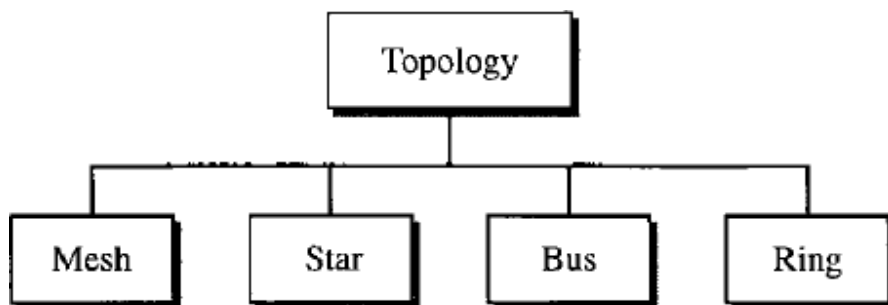
a. Point-to-point



b. Multipoint

Physical Topology

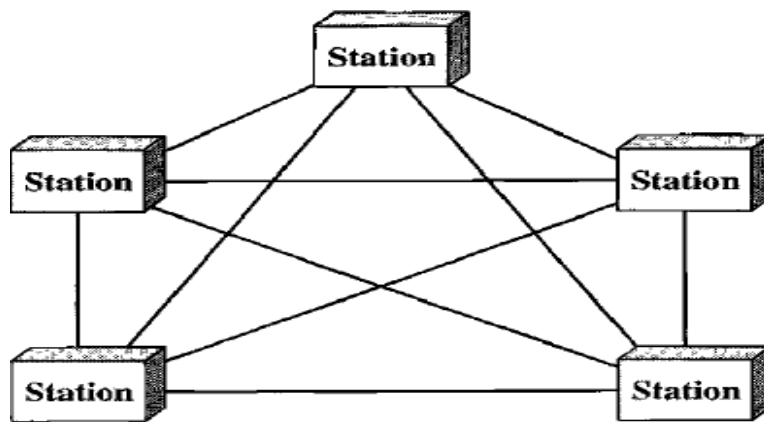
The term *physical topology* refers to the way in which a network is laid out physically. One or more devices connect to a link; two or more links form a topology. The topology of a network is the geometric representation of the relationship of all the links and linking devices (usually called nodes) to one another. There are four basic topologies possible: mesh, star, bus, and ring.



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. To find the number of physical links in a fully connected mesh network with n nodes, we first consider that each node must be connected to every other node. Node 1 must be connected to $n - 1$ nodes, node 2 must be connected to $n - 1$ node, and finally node n must be connected to $n - 1$ nodes. We need $n(n - 1)$ physical links. However, if each physical link allows communication in both directions (duplex mode), we can divide the number of links by 2. In other words, we can say that in a mesh topology, we need $n(n - 1) / 2$ duplex-mode links.

To accommodate that many links, every device on the network must have $n - 1$ input/output (VO) ports to be connected to the other $n - 1$ stations.



Advantages:

1. The use of dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problem that can occur when links must be shared by multiple devices.
2. A mesh topology is robust. If one link becomes unusable, it does not incapacitate the entire system. Third, there is the advantage of privacy or security. When every message travels along a dedicated line, only the intended recipient sees it. Physical boundaries prevent other users from gaining access to messages. Finally, point-to-point links make fault identification and fault isolation easy. Traffic can be routed to avoid links with suspected problems. This facility enables the network manager to discover the precise location of the fault and aids in finding its cause and solution.

Disadvantages:

1. Disadvantage of a mesh is related to the amount of cabling because every device must be

connected to every other device, installation and reconnection are difficult.

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

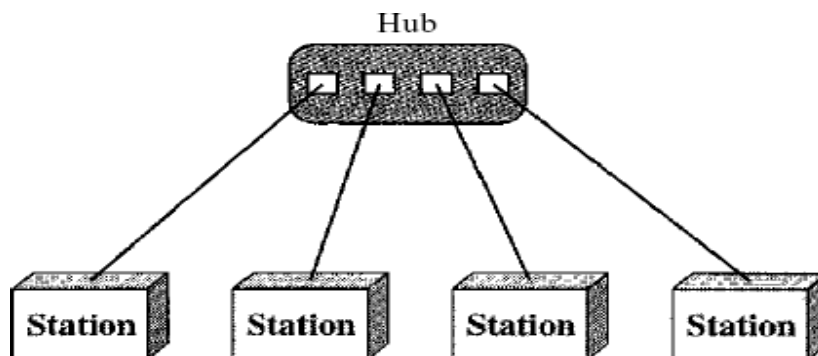
For these reasons a mesh topology is usually implemented in a limited fashion, for example, as a backbone connecting the main computers of a hybrid network that can include several other topologies.

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.

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. This factor also lends itself to easy fault identification and fault isolation. As long as the hub is working, it can be used to monitor link problems and bypass defective links.

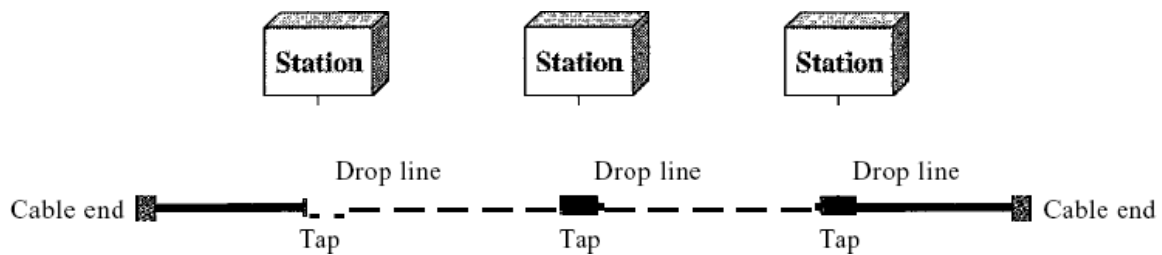


One big disadvantage of a star topology is the dependency of the whole topology on one

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. For this reason, often more cabling is required in a star than in some other topologies (such as ring or bus).

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



Nodes are connected to the bus cable by drop lines and taps. A drop line is a connection running between the device and the main cable. A tap is a connector that either splices into the main cable or punctures the sheathing of a cable to create a contact with the metallic core. As a signal travels along the backbone, some of its energy is transformed into heat. Therefore, it becomes weaker and weaker as it travels farther and farther. For this reason there is a limit on the number of taps a bus can support and on the distance between those taps.

Advantages of a bus topology include ease of installation. Backbone cable can be laid along the most efficient path, then connected to the nodes by drop lines of various lengths. In this way, a bus uses less cabling than mesh or star topologies. In a star, for example, four network devices in the same room require four lengths of cable reaching all the way to the hub. In a bus, this redundancy is eliminated. Only the backbone cable stretches through the entire facility. Each drop line has to reach only as far as the nearest point on the backbone.

Disadvantages include difficult reconnection and fault isolation. A bus is usually designed to be optimally efficient at installation. It can therefore be difficult to add new devices. Signal reflection at the taps can cause degradation in quality. This degradation can be controlled

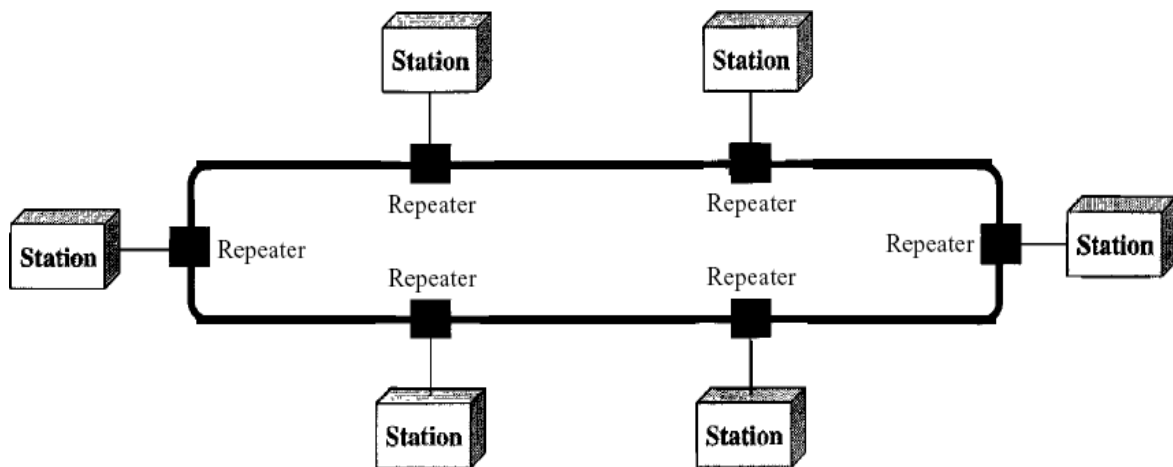
by limiting the number and spacing of devices connected to a given length of cable. Adding new devices may therefore require modification or replacement of the backbone.

In addition, a fault or break in the bus cable stops all transmission, even between devices on the same side of the problem. The damaged area reflects signals back in the direction of origin, creating noise in both directions.

Bus topology was the one of the first topologies used in the design of early local area networks. Ethernet LANs can use a bus topology, but they are less popular.

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. Each device in the ring incorporates a repeater. When a device receives a signal intended for another device, its repeater regenerates the bits and passes them along

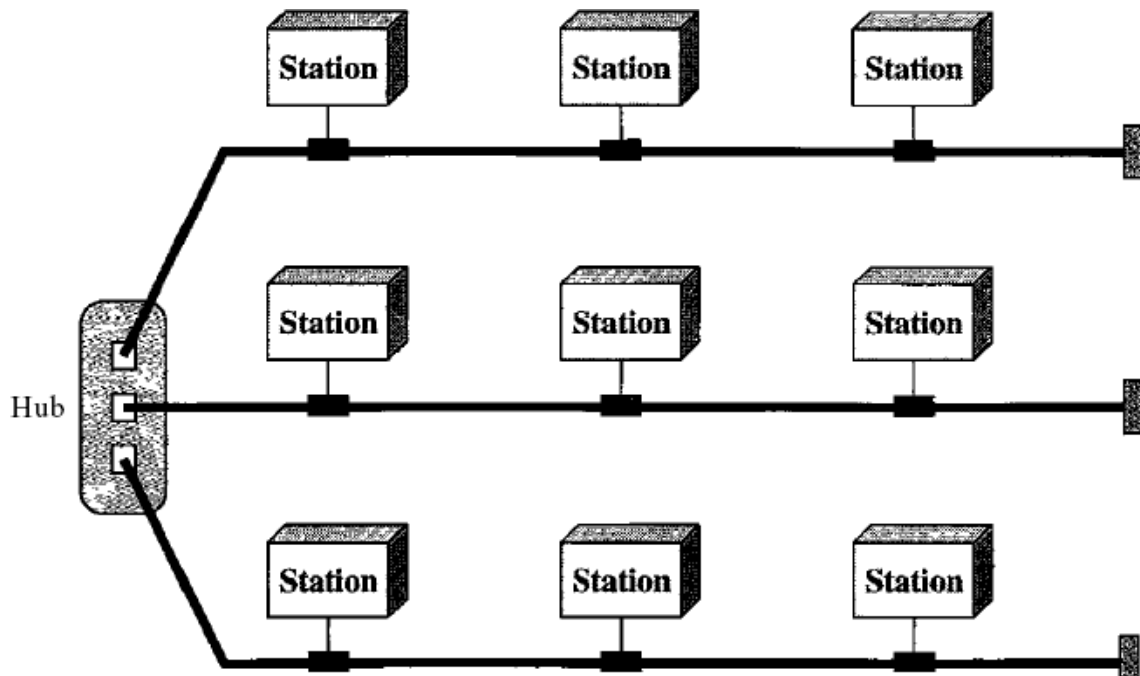


A ring is relatively easy to install and reconfigure. Each device is linked to only its immediate neighbors (either physically or logically). To add or delete a device requires changing only two connections. The only constraints are media and traffic considerations (maximum ring length and number of devices).

In addition, fault isolation is simplified. Generally in a ring, a signal is circulating at all times. If one device does not receive a signal within a specified period, it can issue an alarm. The alarm alerts the network operator to the problem and its location.

However, 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. This weakness can be solved by using a dual ring or a switch capable of closing off the break. Ring topology was prevalent when IBM introduced its local-area network Token Ring.

Today, the need for higher-speed LANs has made this topology less popular. 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 as shown in Figure



2.3: CATEGORIES OF NETWORKS

Local Area Networks:

Local area networks, generally called LANs, are privately-owned networks within a single building or campus of up to a few kilometers in size. They are widely used to connect personal computers and workstations in company offices and factories to share resources (e.g.,

printers) and exchange information. LANs are distinguished from other kinds of networks by three characteristics:

- (1) Their size,
- (2) Their transmission technology, and
- (3) Their topology.

LANs are restricted in size, which means that the worst-case transmission time is bounded and known in advance. Knowing this bound makes it possible to use certain kinds of designs

that would not otherwise be possible. It also simplifies network management. LANs may use a transmission technology consisting of a cable to which all the machines are attached, like the telephone company party lines once used in rural areas. Traditional LANs run at speeds of 10 Mbps to 100 Mbps, have low delay (microseconds or nanoseconds), and make very few errors. Newer LANs operate at up to 10 Gbps. Various topologies are possible for broadcast LANs. Figure 1 shows two of them. In a bus (i.e., a linear cable) network, at any instant at most one machine is the master and is allowed to transmit. All other machines are required to refrain from sending. An arbitration mechanism is needed to resolve conflicts when two or more machines want to transmit simultaneously. The arbitration mechanism may be centralized or distributed. IEEE 802.3, popularly called Ethernet, for example, is a bus-based broadcast network with decentralized control, usually operating at 10 Mbps to 10 Gbps. Computers on an Ethernet can transmit whenever they want to; if two or more packets collide, each computer just waits a random time and tries again later.

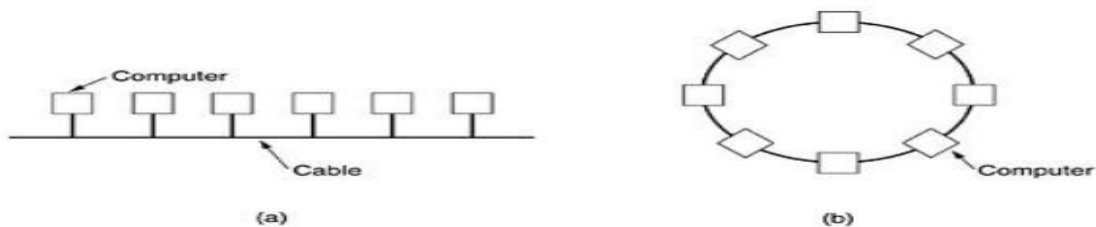


Fig.1: Two broadcast networks. (a) Bus. (b) Ring.

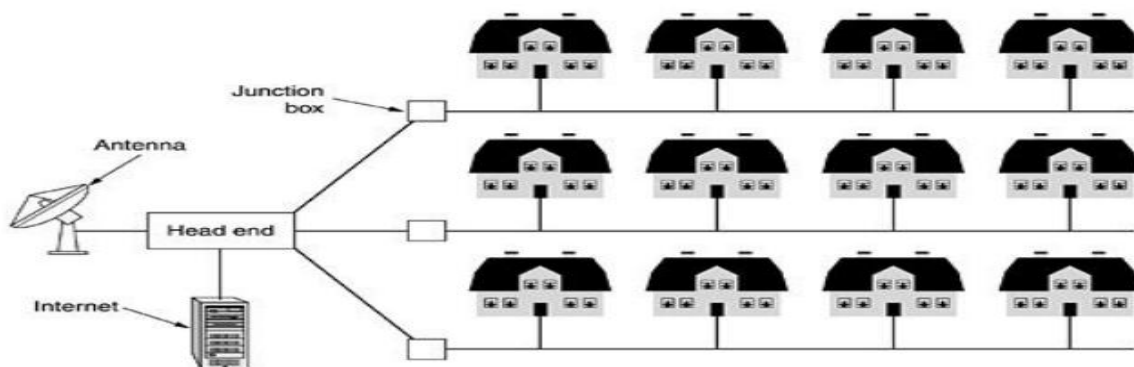
A second type of broadcast system is the ring. In a ring, each bit propagates around on its own, not waiting for the rest of the packet to which it belongs. Typically, each bit

circumnavigates the entire ring in the time it takes to transmit a few bits, often before the complete packet has even been transmitted. As with all other broadcast systems, some rule is needed for arbitrating simultaneous accesses to the ring. Various methods, such as having the machines take turns, are in use. IEEE 802.5 (the IBM token ring), is a ring-based LAN operating at 4 and 16 Mbps. FDDI is another example of a ring network.

Metropolitan Area Network (MAN):

Metropolitan Area Network:

A metropolitan area network, or MAN, covers a city. The best-known example of a MAN is the cable television network available in many cities. This system grew from earlier community antenna systems used in areas with poor over-the-air television reception. In these early systems, a large antenna was placed on top of a nearby hill and signal was then piped to the subscribers' houses. At first, these were locally-designed, ad hoc systems. Then companies began jumping into the business, getting contracts from city governments to wire up an entire city. The next step was television programming and even entire channels designed for cable only. Often these channels were highly specialized, such as all news, all sports, all cooking, all gardening, and so on. But from their inception until the late 1990s, they were intended for television reception only. To a first approximation, a MAN might look something like the system shown in Fig. In this figure both television signals and Internet are fed into the centralized head end for subsequent distribution to people's homes. Cable television is not the only MAN. Recent



developments in high-speed wireless Internet access resulted in another MAN, which has been standardized as IEEE 802.16.

Fig.2: Metropolitan area network based on cable TV.

A MAN is implemented by a standard called DQDB (Distributed Queue Dual

Bus) or IEEE 802.16. DQDB has two unidirectional buses (or cables) to which all the computers are attached.

Wide Area Network (WAN).

Wide Area Network: A wide area network, or WAN, spans a large geographical area, often a country or continent. It contains a collection of machines intended for running user (i.e., application) programs. These machines are called as hosts. The hosts are connected by a communication subnet, or just subnet for short. The hosts are owned by the customers (e.g., people's personal computers), whereas the communication subnet is typically owned and operated by a telephone company or Internet service provider. The job of the subnet is to carry

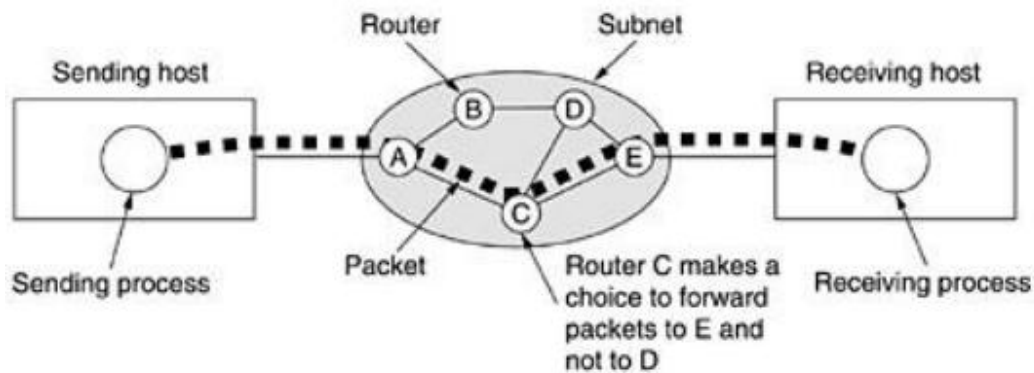
messages from host to host, just as the telephone system carries words from speaker to listener.

Separation of the pure communication aspect of the network (the subnet) from the application aspects (the hosts), greatly simplifies the complete network design. In most wide area networks, the subnet consists of two distinct components: transmission lines and switching elements. Transmission lines move bits between machines. They can be made of copper wire, optical fiber, or even radio links. In most WANs, the network contains numerous transmission lines, each one connecting a pair of routers. If two routers that do not share a transmission line wish to communicate, they must do this indirectly, via other routers. When a packet is sent from one router to another via one or more intermediate routers, the packet is received at each intermediate router in its entirety, stored there until the required output line is free, and then forwarded. A subnet organized according to this principle is called a store-and-forward or packet-

switched subnet. Nearly all wide area networks (except those using satellites) have store-and-forward subnets. When the packets are small and all the same size, they are often called cells.

The principle of a packet-switched WAN is so important. Generally, when a process on some host has a message to be sent to a process on some other host, the sending host first cuts the message into packets, each one bearing its number in the sequence. These packets are then injected into the network one at a time in quick succession. The packets are transported individually over the network and deposited at the receiving host, where they are reassembled into the original message and delivered to the receiving process. A stream of packets resulting from some initial message is illustrated in Fig.

In this figure, all the packets follow the route ACE, rather than ABDE or ACDE. In some networks all packets from a given message must follow the same route; in others each packet is routed separately. Of course, if ACE is the best route, all packets may be sent along it, even if each



packet is individually routed.

Fig.3.1: A stream of packets from sender to receiver.

Not all WANs are packet switched. A second possibility for a WAN is a satellite system. Each router has an antenna through which it can send and receive. All routers can hear the output from the satellite, and in some cases they can also hear the upward transmission of their fellow routers to the satellite as well. Sometimes the routers are connected to a substantial point-to-point subnet, with only some of them having a satellite antenna. Satellite networks are inherently broadcast and are most useful when the broadcast property is important.

THE INTERNET

The Internet has revolutionized many aspects of our daily lives. It has affected the way we do business as well as the way we spend our leisure time. Count the ways you've used the Internet recently. Perhaps you've sent electronic mail (e-mail) to a business associate, paid a utility bill, read a newspaper from a distant city, or looked up a local movie schedule—all by using the Internet. Or maybe you researched a medical topic, booked a hotel reservation, chatted with a fellow Trekkie, or comparison-shopped for a car. The Internet is a communication system that has brought a wealth of information to our fingertips and organized it for our use.

A Brief History

A network is a group of connected communicating devices such as computers and printers. An internet (note the lowercase letter i) is two or more networks that can communicate with each other. The most notable internet is called the Internet (uppercase letter I), a collaboration of more than hundreds of thousands of interconnected networks. Private individuals as well as various organizations such as government agencies, schools, research facilities, corporations, and libraries in more than 100 countries use the Internet. Millions of people are users. Yet this extraordinary communications system only came into being in 1969.

In the mid-

1960s, mainframe computers in research organizations were stand-alone devices. Computers from different manufacturers were unable to communicate with one another. The Advanced Research Projects Agency (ARPA) in the Department of Defense (DoD) was interested in finding a way to connect computers so that the researchers they funded could share their findings, thereby reducing costs and eliminating duplication of effort.

In 1967, at an Association for Computing Machinery (ACM) meeting, ARPA presented its ideas for ARPANET, a small network of connected computers. The idea was that each host computer (not necessarily from the same manufacturer) would be attached to a specialized computer, called an *interface message processor* (IMP). The IMPs, in turn, would be connected to one another. Each IMP had to be able to communicate with other IMPs as well as with its own attached host. By 1969, ARPANET was a reality. Four nodes, at the University of California at Los Angeles (UCLA), the University of California at Santa Barbara (UCSB), Stanford Research Institute (SRI), and the University of Utah, were connected via the IMPs to form a network. Software called the *Network Control Protocol* (NCP) provided communication between the hosts.

In 1972, Vint Cerf and Bob Kahn, both of whom were part of the core ARPANET group, collaborated on what they called the *Internetworking Project*. Cerf and Kahn's landmark 1973 paper outlined the protocols to achieve end-to-end delivery of packets. This paper on Transmission Control Protocol (TCP) included concepts such as encapsulation, the datagram, and the function of a gateway. Shortly thereafter, authorities made a decision to split TCP into two protocols: Transmission Control Protocol (TCP) and Internetworking Protocol (IP). IP would have

ndledatagramroutingwhileTCPwouldberesponsibleforhigher-levelfunctionssuchas segmentation,reassembly,anderrordetection.Theinternetworkingprotocolbecameknownas TCPIIP.

TheInternetToday

TheInternethascomealongway

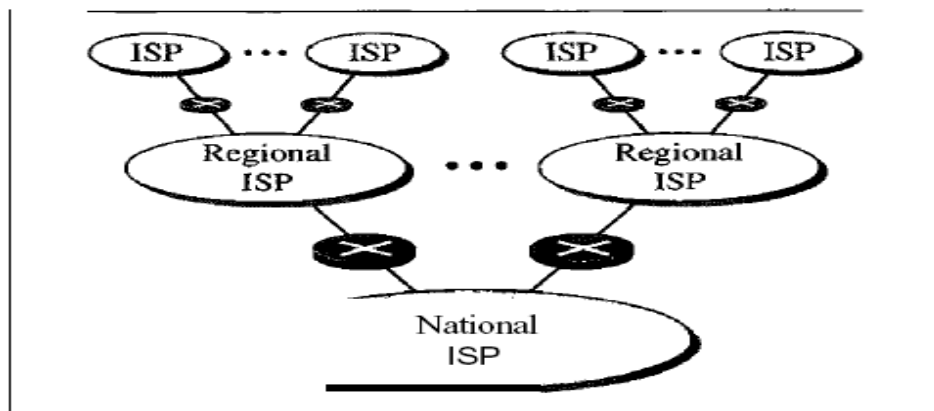
since the 1960s. The Internet today is not a simple hierarchical structure. It is made up of many wide- and local-

areanetworksjoinedbyconnectingdevicesandswitchingstations.ItisdifficulttogiveanaccuraterepresentationoftheInternetbecauseitiscontinuallychanging-

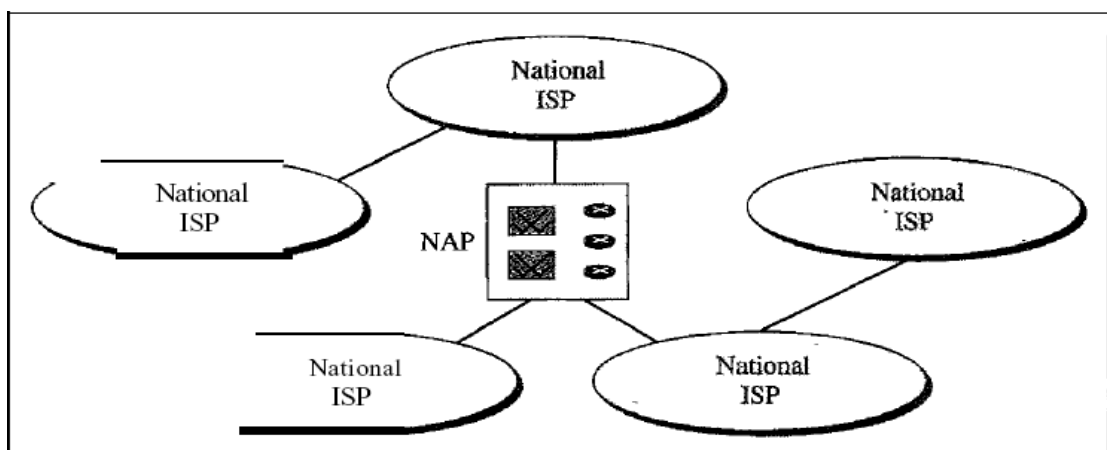
newnetworksarebeingadded,existingnetworksareaddingaddresses,andnetworksofdefunctcompaniesarebeingremoved.TodaymostenduserswhowantInternetconnectionusethe service

sofInternetserviceproviders(ISPs).Thereareinternationalserviceproviders,nationalservice providers,regionalserviceproviders,andlocalserviceproviders.TheInternettodayisrunby

privatecompanies,notthegovernment.Figureshowsaconceptual(notgeographic)viewoftheInternet.



a. Structure of a national ISP



b. Interconnection of national ISPs

International Internet Service Providers:

At the top of the hierarchy are the international service providers that connect nations together.

National Internet Service Providers:

The national Internet service providers are backbone networks created and maintained by specialized companies. There are many national ISPs operating in North America; some of the most well known are Sprint Link, PSINet, UUNet Technology, AGIS, and Internet Mel. To provide connectivity between the end users, these backbone networks are connected by complex switching stations (normally run by a third party) called network access points (NAPs). Some national ISP networks are also connected to one another by private switching stations called *peering points*. These normally operate at a high data rate (up to 600 Mbps).

Regional Internet Service Providers:

Regional Internet service providers or regional ISPs are smaller ISPs that are connected to one or more national ISPs. They are at the third level of the hierarchy with a smaller data rate. *Local Internet Service Providers:*

Local Internet service providers provide direct service to the end users. The local ISP can be connected to regional ISPs or directly to national ISPs. Most end users are connected to the local ISPs. Note that in this sense, a local ISP can be a company that just provides Internet services, a corporation with a network that supplies services to its own employees, or a nonprofit organization, such as a college or a university, that runs its own network. Each of these local ISPs can be connected to a regional or national service provider.

2.4 PROTOCOLS AND STANDARDS

Protocols:

In computer networks, communication occurs between entities in different systems. An entity is anything capable of sending or receiving information. However, two entities cannot simply send bit streams to each other and expect to be understood. For communication to occur, the entities must agree on a protocol. A protocol is a set of rules that govern data communication.

ions. A protocol defines what is communicated, how it is communicated, and when it is communicated. The key elements of a protocol are syntax, semantics, and timing.

- o Syntax. The term *syntax* refers to the structure or format of the data, meaning the order in which they are presented. For example, a simple protocol might expect the first 8 bits of data to be the address of the sender, the second 8 bits to be the address of the receiver, and the rest of the stream to be the message itself.

- o Semantics. The word *semantics* refers to the meaning of each section of bits. How is a particular pattern to be interpreted, and what action is to be taken based on that interpretation? For example, does an address identify the route to be taken or the final destination of the message?

- o Timing. The term *timing* refers to two characteristics: when data should be sent and how fast they can be sent. For example, if a sender produces data at 100 Mbps but the receiver can process data at only 1 Mbps, the transmission will overload the receiver and some data will be lost.

Standards

Standards are essential in creating and maintaining an open and competitive market for equipment manufacturers and in guaranteeing national and international interoperability of data and telecommunication technology and processes. Standards provide guidelines to manufacturers, vendors, government agencies, and other service providers to ensure the kind of interconnectivity necessary in today's marketplace and in international communications. Data communication standards fall into two categories: *de facto* (meaning "by fact" or "by convention") and *de jure* (meaning "by law" or "by regulation").

- o De facto. Standards that have not been approved by an organized body but have been adopted as standards through widespread use are de facto standards. De facto standards are often established originally by manufacturers who seek to define the functionality of a new product or technology.

- o De jure. Those standards that have been legislated by an officially recognized body are de jure standards.

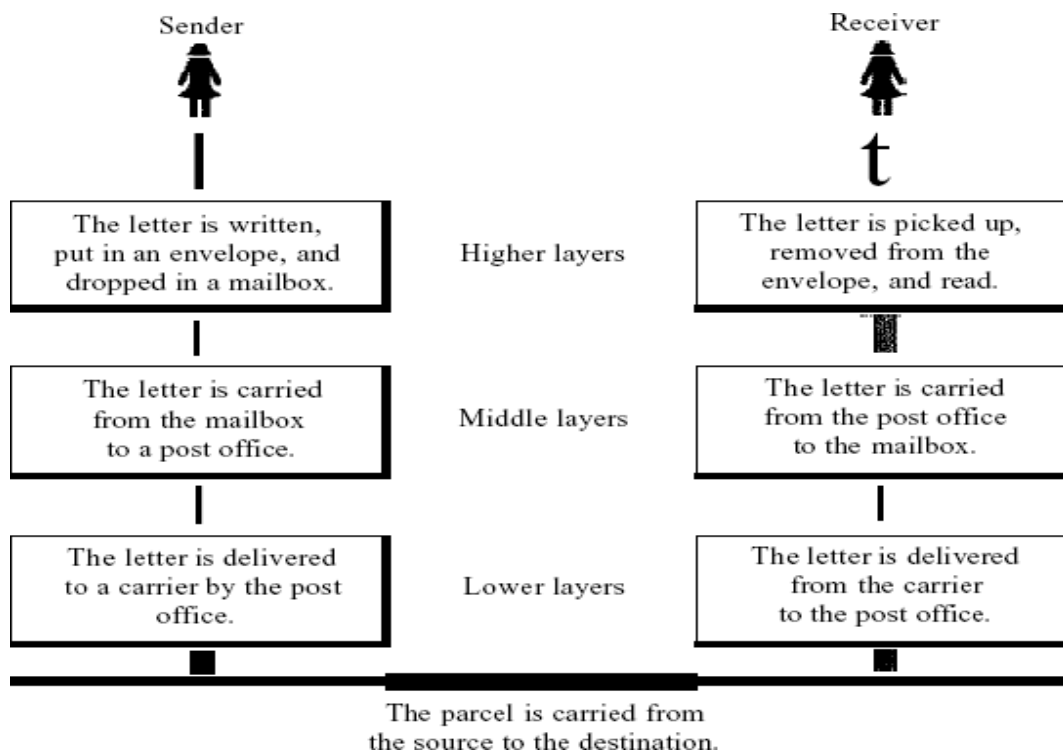
LAYERED TASKS

We use the concept of layers in our daily life. As an example, let us consider two friends who

communicate through postal mail. The process of sending a letter to a friend would become complex if there were no services available from the post office. Below Figure shows the steps in this task.

Sender, Receiver, and Carrier

In Figure we have a sender, a receiver, and a carrier that transport the letter. There is a hierarchy of



asks.

At the Sender Site

Let us first describe, in order, the activities that take place at the sender site.

- o Higher layer. The sender writes the letter, inserts the letter in an envelope, writes the sender and receiver addresses, and drops the letter in a mailbox.
- o Middle layer. The letter is picked up by a letter carrier and delivered to the post office.
- o Lower layer. The letter is sorted at the post office; a carrier transports the letter.

On the Way: The letter is then on its way to the recipient. On the way to the recipient's local post office, the letter may actually go through a central office. In addition, it may be transported by truck, train, airplane, boat, or a combination of these.

At the Receiver Site

- o Lower layer. The carrier transports the letter to the post office.
- o Middle layer. The letter is sorted and delivered to the recipient's mailbox.
- o Higher layer. The receiver picks up the letter, opens the envelope, and reads it.

2.5 REFERENCE MODELS

The OSI Reference Model:

The OSI model (minus the physical medium) is shown in Fig. This model is based on a proposal developed by the International Standards Organization (ISO) as a first step toward international standardization of the protocols used in the various layers (Day and Zimmermann, 1983). It was revised in 1995 (Day, 1995). The model is called the ISO-OSI (Open Systems Interconnection) Reference Model because it deals with connecting open systems—that is, systems that are open for communication with other systems.

The OSI model has seven layers. The principles that were applied to arrive at these seven layers can be briefly summarized as follows:

1. A layer should be created where a different abstraction is needed.
2. Each layer should perform a well-defined function.
3. The function of each layer should be chosen with an eye toward defining internationally standardized protocols.
4. The layer boundaries should be chosen to minimize the information flow across the interfaces.
5. The number of layers should be large enough that distinct functions need not be thrown together in the same layer out of necessity and small enough that the architecture does not become unwieldy.

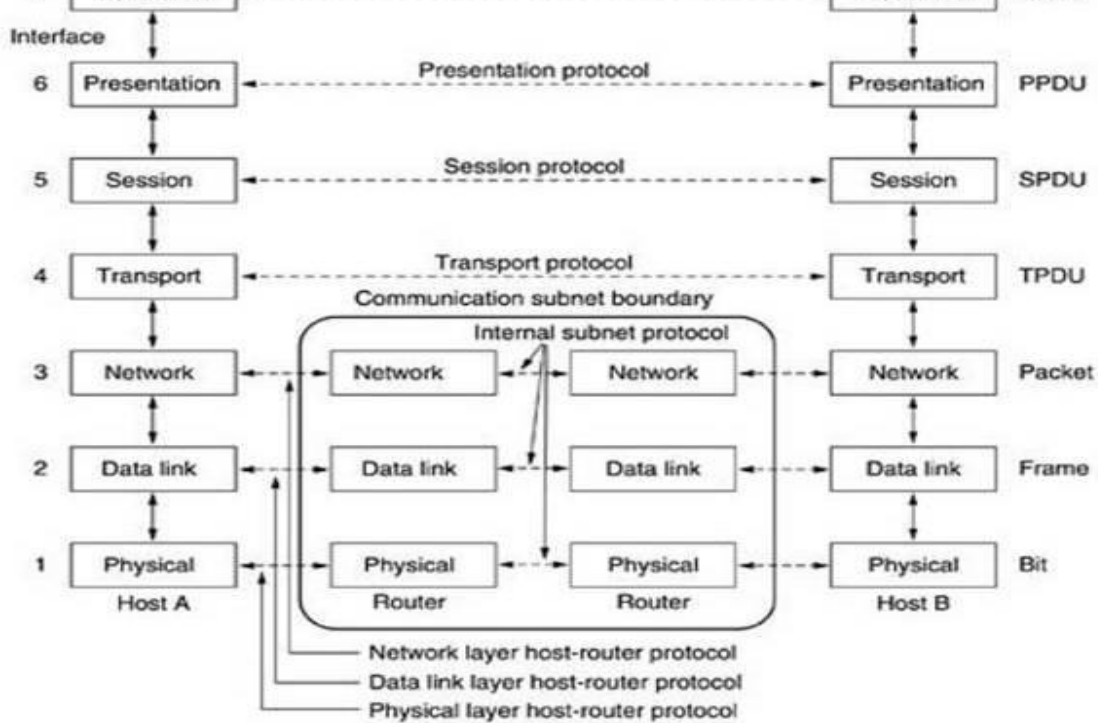


Fig.4: The OSI reference model

The Physical Layer:

The physical layer is concerned with transmitting raw bits over a communication channel. The design issues have to do with making sure that when one side sends a 1 bit, it is received by the other side as a 1 bit, not as a 0 bit.

The Data Link Layer:

The main task of the data link layer is to transform a raw transmission facility into a line that appears free of undetected transmission errors to the network layer. It accomplishes this task by having the sender break up the input data into data frames (typically a few hundred or a few thousand bytes) and transmits the frames sequentially. If the service is reliable, the receiver confirms correct receipt of each frame by sending back an acknowledgement frame.

Another issue that arises in the data link layer (and most of the higher layers as well) is how

to keep a fast transmitter from drowning a slow receiver in data. Some traffic regulation mechanisms are often needed to let the transmitter know how much buffer space the receiver has at the moment. Frequently, this flow regulation and the error handling are integrated.

The Network Layer:

The network layer controls the operation of the subnet. A key design issue is determining how packets are routed from source to destination. Routes can be based on static tables that

are

"wired into" the network and rarely changed. They can also be determined at the start of each conversation, for example, a terminal session (e.g., a login to a remote machine). Finally, they can be highly dynamic, being determined anew for each packet, to reflect the current network load.

If too many packets are present in the subnet at the same time, they will get in one another's way, forming bottlenecks. The control of such congestion also belongs to the network layer. More generally, the quality of service provided (delay, transit time, jitter, etc.) is also a network layer issue.

When a packet has to travel from one network to another to get to its destination, many problems can arise. The addressing used by the second network may be different from the first one. These two may not accept the packet at all because it is too large. The protocols may differ, and so on. It is up to the network layer to overcome all these problems to allow heterogeneous networks to be interconnected. In broadcast networks, the routing problem is simple, so the network layer is often thin or even nonexistent.

The Transport Layer:

The basic function of the transport layer is to accept data from above, split it up into smaller units if needed, pass these to the network layer, and ensure that the pieces all arrive correctly at the other end. Furthermore, all this must be done efficiently and in a way that isolates the upper layers from the inevitable changes in the hardware technology. The transport layer also determines what type of service to provide to the session layer, and, ultimately, to the users of the network. The most popular type of transport connection is an error-free point-to-point channel that delivers messages or bytes in the order in which they were sent. However, other possible kinds of transport service are the transporting of isolated messages, with no guarantee about the order of delivery, and the broadcasting of messages to multiple destinations. The type of service is determined when the connection is established.

The transport layer is a true end-to-end layer, all the way from the source to the destination. In other words, a program on the source machine carries on a conversation with a similar program on the destination machine, using the message headers and control messages. In the lower layers, the protocols are between each machine and its immediate neighbours, and not between the ultimate source

and destination machines, which may be separated by many routers.

The Session Layer:

The session layer allows users on different machines to establish sessions between them. Sessions offer various services, including dialog control (keeping track of whose turn it is to transmit), token management (preventing two parties from attempting the same critical operation at the same time), and synchronization (checkpointing long transmissions to allow them to continue from where they were after a crash).

The Presentation Layer:

The presentation layer is concerned with the syntax and semantics of the information transmitted. In order to make it possible for computers with different data representations to communicate, the data structures to be exchanged can be defined in an abstract way, along with a standard encoding to be used "on the wire." The presentation layer manages these abstract data structures and allows higher-level data structures (e.g., banking records), to be defined and exchanged.

The Application Layer:

The application layer contains a variety of protocols that are commonly needed by users. One widely-used application protocol is HTTP (Hypertext Transfer Protocol), which is the basis for the World Wide Web. When a browser wants a Web page, it sends the name of the page it wants to the server using HTTP. The server then sends the page back. Other application protocols are used for file transfer, electronic mail, and network news.

2.6 The TCP/IP Reference Model:

The TCP/IP reference model was developed prior to the OSI model. The major design goals of this model were,

1. To connect multiple networks together so that they appear as a single network.
2. To survive after partial subnetwork hardware failures.
3. To provide a flexible architecture.

Unlike the OSI reference model, the TCP/IP reference model has only 4 layers. They are,

1. Host-to-Network Layer

2. InternetLayer
3. TransportLayer
4. Application
Layer
ApplicationLayer
TransportLayer
InternetLayer
Host-to-NetworkLayer

Host-to-NetworkLayer:

The TCP/IP reference model does not really say much about what happens here, except to point out that the host has to connect to the network using some protocol so it can send IP packets to it. This protocol is not defined and varies from host to host and network to network.

InternetLayer:

This layer, called the internet layer, is the linchpin that holds the whole architecture together. Its job is to permit hosts to inject packets into any network and have them travel independently to the destination (potentially on a different network). They may even arrive in a different order than they were sent, in which case it is the job of higher layers to rearrange them, if in-order delivery is desired. Note that "internet" is used here in a generic sense, even though this layer is present in the Internet.

The internet layer defines an official packet format and protocol called IP (Internet Protocol). The job of the internet layer is to deliver IP packets where they are supposed to go. Packet routing is clearly the major issue here, as is avoiding congestion. For these reasons, it is reasonable to say that the TCP/IP internet layer is similar in functionality to the OSI network layer. Fig. shows this correspondence.

The TransportLayer:

The layer above the internet layer in the TCP/IP model is now usually called the transport layer. It is designed to allow peer entities on the source and destination hosts to carry on a conversation, just as in the OSI transport layer. Two end-to-end transport protocols have been defined here. The first one, TCP (Transmission Control Protocol), is a reliable connection-oriented protocol that allows a byte stream originating on one machine to be

delivered without error on any other machine in the internet. It fragments the incoming byte stream into discrete messages and passes each one on to the internet layer. At the destination, the receiving TCP process reassembles the received messages into the output stream. TCP also handles flow control to make sure a fast sender cannot swamp a slow receiver with more messages than it can handle.

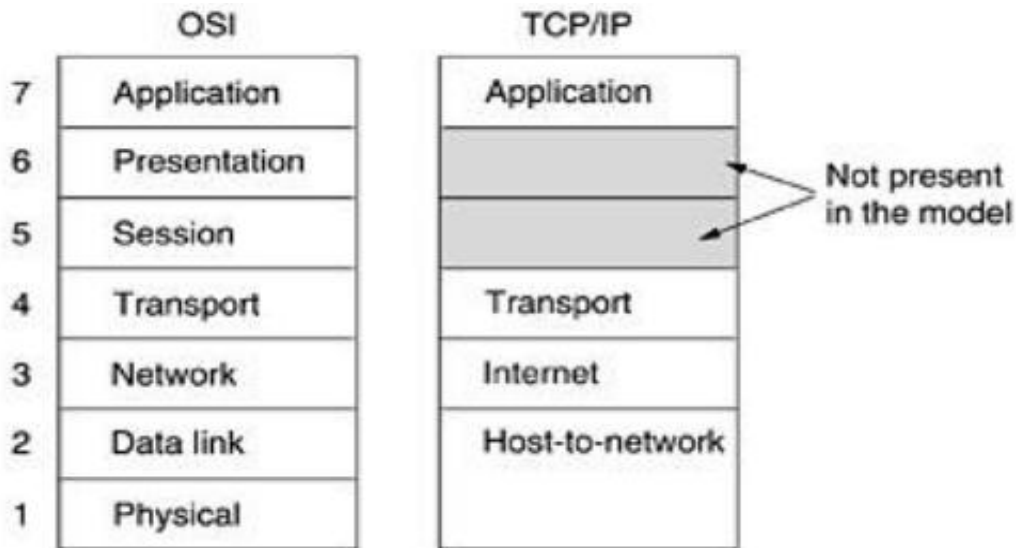


Fig.1: The TCP/IP Preference model.

The second protocol in this layer, UDP (User Datagram Protocol), is an unreliable, connectionless protocol for applications that do not want TCP's sequencing or flow control and wish to provide their own. It is also widely used for one-shot, client-server-type request-reply queries and applications in which prompt delivery is more important than accurate delivery, such as transmitting speech or video. The relation of IP, TCP, and UDP is shown in Fig.2. Since the model was developed, IP has been implemented on many other networks.

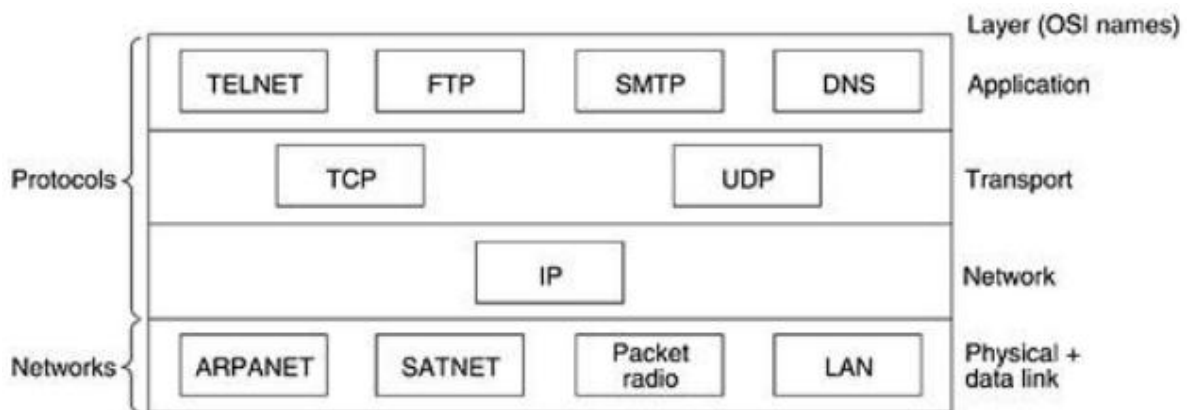


Fig.2:ProtocolsandnetworksintheTCP/IPmodelinitially.

TheApplicationLayer:

The TCP/IP model does not have session or presentation layers. On top of the transport layer is the application layer. It contains all the higher-level protocols. The early ones included virtual terminal (TELNET), file transfer (FTP), and electronic mail (SMTP), as shown in Fig.6.2. The virtual terminal protocol allows a user on one machine to log onto a distant machine and work there. The file transfer protocol provides a way to move data efficiently from one machine to another. Electronic mail was originally just a kind of file transfer, but later a specialized protocol (SMTP) was developed for it. Many other protocols have been added to these over the years: the Domain Name System (DNS) for mapping host names onto their network addresses, NNTP, the protocol for moving USENET news articles around, and HTTP, the protocol for fetching pages on the World Wide Web, and many others.

ComparisonoftheOSIandTCP/IPReferenceModels:

The OSI and TCP/IP reference models have much in common. Both are based on the concept of a stack of independent protocols. Also, the functionality of the layers is roughly similar. For example, in both models the layers up through and including the transport layer are there to provide an end-to-end, network-independent transport service to processes wishing to communicate. These layers form the transport provider. Again in both models, the layers above transport are application-oriented users of the transport service. Despite these fundamental similarities, the two models also have many differences. Three concepts are central to the OSI model:

1. Services.
2. Interfaces.
3. Protocols.

Probably the biggest contribution of the OSI model is to make the distinction between these three concepts explicit. Each layer performs some services for the layer above it. The service definition tells what the layer does, not how entities above it access it or how the layer works. It defines the layer's semantics.

A layer's interface tells the processes above it how to access it. It specifies what the parameters are and what results to expect. It, too, says nothing about how the layer works inside.

Finally, the peer protocols used in a layer are the layer's own business. It can use any protocols it wants to, as long as it gets the job done (i.e., provides the offered services). It can also change them at will without affecting software in higher layers.

The TCP/IP model did not originally clearly distinguish between service, interface, and protocol, although people have tried to retrofit it after the fact to make it more OSI-like.

For example, the only real services offered by the internet layer are SEND_IP_PACKET and RECEIVE_IP_PACKET.

As a consequence, the protocols in the OSI model are better hidden than in the TCP/IP model and can be replaced relatively easily as the technology changes. Being able to make such changes is one of the main purposes of having layered protocols in the first place. The OSI reference model was devised before the corresponding protocols were invented. This ordering means that the model was not biased toward one particular set of protocols, a fact that made it quite general. The downside of this ordering is that the designers did not have much experience with the subject and did not have a good idea of which functionality to put in which layer.

Another difference is in the area of connectionless versus connection-oriented communication. The OSI model supports both connectionless and connection-oriented communication in the network layer, but only connection-oriented communication in the transport layer, where it counts (because the transport service is visible to the users). The TCP/IP model has only one mode in the network layer (connectionless) but supports both modes in the transport layer, giving the users a choice. This choice is especially important for simple request-response protocols.

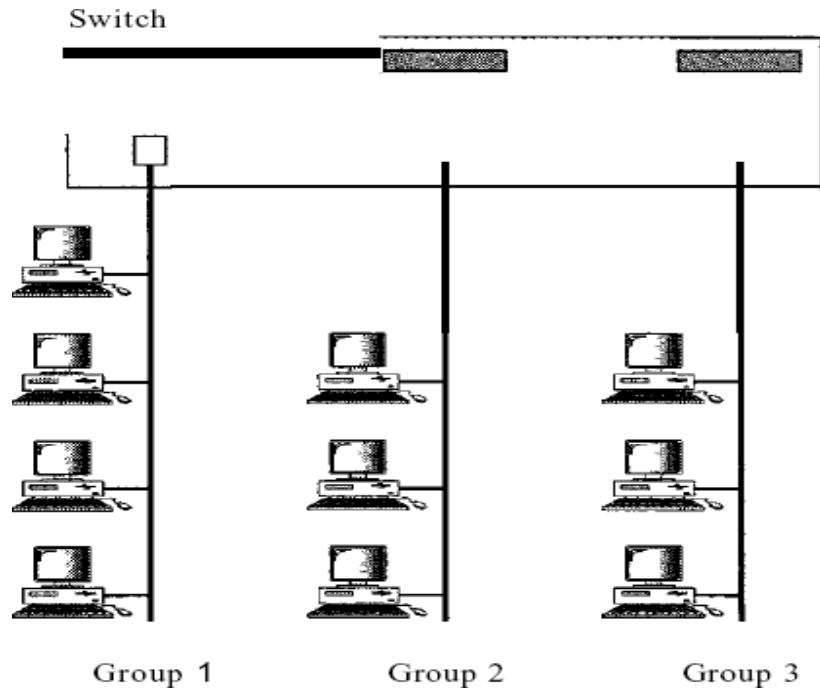


Figure 6 A switch connecting three LANs

2.6 SUMMARY

At the end of this unit we have learnt about physical structure which includes topologies of networks; we dealt with the concept of categories of networks (LAN, MAN, WAN). In the later stage we focused protocols and standards. In the last section of this unit we have discussed reference models which includes OSI reference model and TCP/IP reference model.

2.7 KEYWORDS

Mesh topology, Star topology, Bus topology, Ring topology, LAN, WAN, MAN, Internet.

2.8 QUESTIONS

1. Briefly explain physical structure.
2. Discuss protocols and standards.
3. With neat diagram explain OSI reference model.
4. Differentiate OSI and TCP/IP reference model.

5. Describe categories of network.

2.9 REFERENCES

- William Stallings, Data and Computer Communication, Prentice Hall of India
- Behrouz A. Forouzan, Data Communication and Networking, McGraw-Hill
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UNIT 3: PHYSICAL LAYER

Structure:

3.0 Objectives

3.1 Introduction

3.2 Overview of data

3.3 Data and Signals

- 3.4 Digital Transmission
- 3.5 Analog Transmission
- 3.6 Summary
- 3.7 Keywords
- 3.8 Questions
- 3.9 Reference

3.0 OBJECTIVES

After going through this lesson you will be able to

- describe Overview of data D
- discuss data and signals. D
- elucidate digital transmission E
- explain analog transmission E

3.1 INTRODUCTION

Physical Layer is the bottom-most layer in the **Open System Interconnection (OSI) Model** which is a physical and electrical representation of the system. It consists of various network components such as power plugs, connectors, receivers, cable types, etc. Physical Layer sends data bits from one device(s) (like a computer) to another device(s). Physical Layer defines the types of encoding (that is how the 0's and 1's is encoded in a signal). Physical Layer is responsible for the communication of the unstructured raw data streams over a physical medium.

PHYSICAL LAYER

Physical layer in the OSI model plays the role of interacting with actual hardware and signaling mechanism. Physical layer is the only layer of OSI network model which actually

deals with the physical connectivity of two different stations. This layer defines the hardware equipment, cabling, wiring, frequencies, pulses used to represent binary signals etc.

Physical layer provides its services to Data-link layer. Data-link layer hands over frames to physical layer. Physical layer converts them to electrical pulses, which represent binary data. The binary data is then sent over the wired or wireless media.

Signals

When data is sent over physical medium, it needs to be first converted into electromagnetic signals. Data itself can be analog such as human voice, or digital such as file on the disk. Both analog and digital data can be represented in digital or analog signals.

- **Digital Signals**

Digital signals are discrete in nature and represent sequence of voltage pulses. Digital signals are used within the circuitry of a computer system.

- **Analog Signals**

Analog signals are in continuous wave form in nature and represented by continuous electromagnetic waves.

Transmission Impairment

When signals travel through the medium they tend to deteriorate. This may have many reasons as given:

- **Attenuation**

For the receiver to interpret the data accurately, the signal must be sufficiently strong. When the signal passes through the medium, it tends to get weaker. As it covers distance, it loses strength.

- **Dispersion**

As signal travels through the media, it tends to spread and overlaps. The amount of dispersion depends upon the frequency used.

- **Delay distortion**

Signals are sent over media with pre-defined speed and frequency. If the signal speed and frequency do not match, there are possibilities that signal reaches destination in arbitrary fashion. In digital media, this is very critical that some bits reach earlier than the previously sent ones.

- **Noise**

Random disturbance or fluctuation in analog or digital signal is said to be Noise in signal, which may distort the actual information being carried. Noise can be characterized in one of the following class:

- **Thermal Noise**

Heat agitates the electronic conductors of a medium which may introduce noise in the media. Up to a certain level, thermal noise is unavoidable.

- **Intermodulation**

When multiple frequencies share a medium, their interference can cause noise in the medium. Intermodulation noise occurs if two different frequencies are sharing a medium and one of them has excessive strength or the component itself is not functioning properly, then the resultant frequency may not be delivered as expected.

- **Crosstalk**

This sort of noise happens when a foreign signal enters into the media. This is because signal in one medium affects the signal of second medium.

- **Impulse**

This noise is introduced because of irregular disturbances such as lightening, electricity, short-circuit, or faulty components. Digital data is mostly affected by this sort of noise.

Transmission Media

The media over which the information between two computer systems is sent, called transmission media. Transmission media comes in two forms.

- **Guided Media**

All communication wires/cables are guided media, such as UTP, coaxial cables, and fiber Optics. In this media, the sender and receiver are directly connected and the information is send (guided) through it.

- **Unguided Media**

Wireless or open air space is said to be unguided media, because there is no connectivity between the sender and receiver. Information is spread over the air, and anyone including the actual recipient may collect the information.

Channel Capacity

The speed of transmission of information is said to be the channel capacity. We count it as data rate in digital world. It depends on numerous factors such as:

- **Bandwidth:** The physical limitation of underlying media.
- **Error-rate:** Incorrect reception of information because of noise.
- **Encoding:** The number of levels used for signaling.

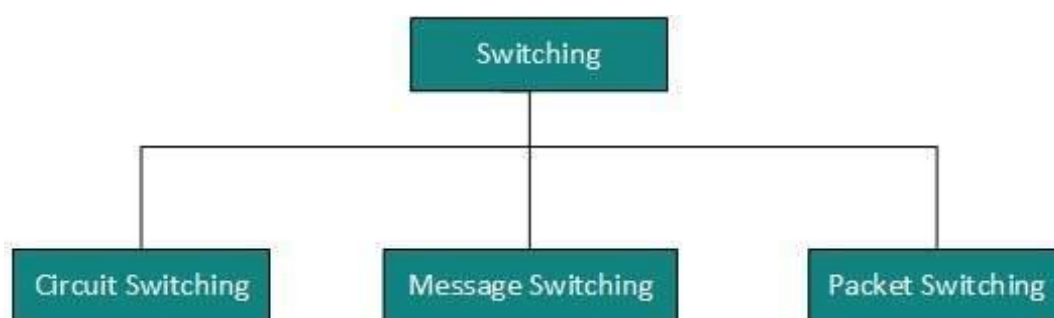
Multiplexing

Multiplexing is a technique to mix and send multiple data streams over a single medium. This technique requires system hardware called multiplexer (MUX) for multiplexing the streams and sending them on a medium, and de-multiplexer (DMUX) which takes information from the medium and distributes to different destinations.

Switching

Switching is a mechanism by which data/information sent from source towards destination which are not directly connected. Networks have interconnecting devices, which receives data from directly connected sources, stores data, analyze it and then forwards to the next interconnecting device closest to the destination.

Switching can be categorized as:



Data or information can be stored in two ways, analog and digital. For a computer to use the data, it must be in discrete digital form. Similar to data, signals can also be in analog and digital form. To transmit data digitally, it needs to be first converted to digital form.

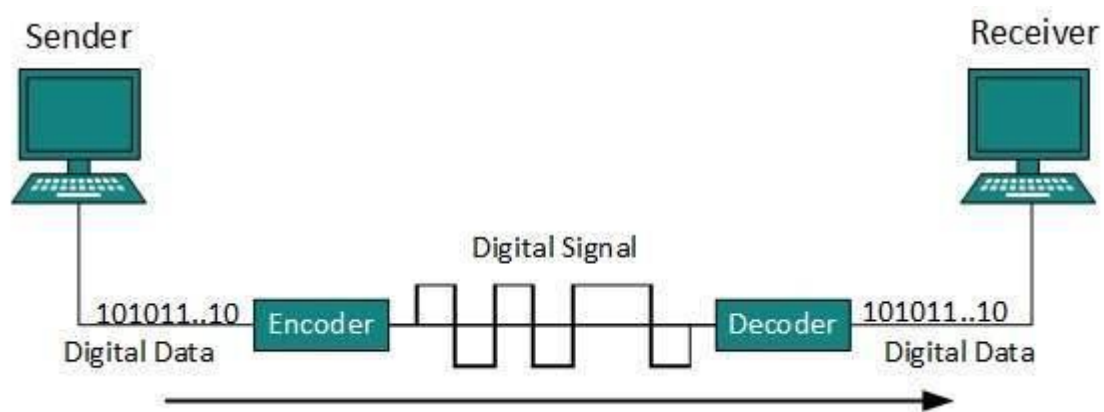
3.4 DIGITAL TRANSMISSION

Digital-to-Digital Conversion

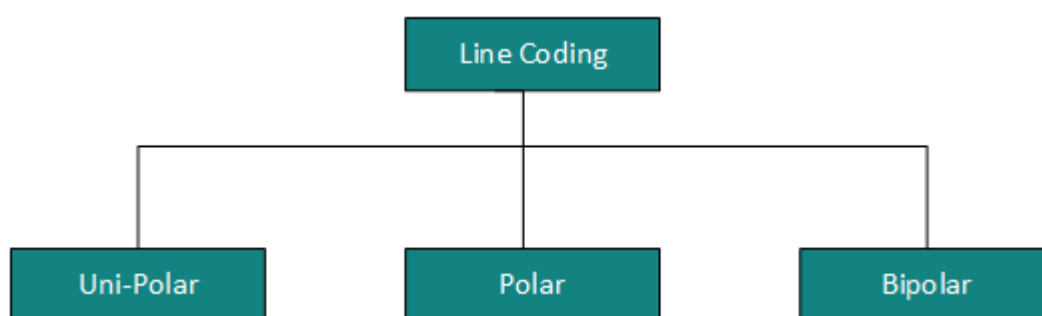
This section explains how to convert digital data into digital signals. It can be done in two ways, line coding and block coding. For all communications, line coding is necessary whereas block coding is optional.

Line Coding

The process for converting digital data into digital signal is said to be Line Coding. Digital data is found in binary format. It is represented (stored) internally as series of 1s and 0s.

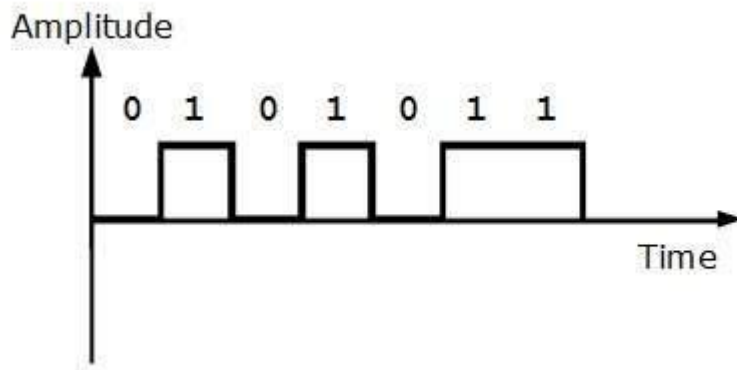


Digital signal is denoted by discrete signal, which represents digital data. There are three types of line coding schemes available:



Uni-polar Encoding

Unipolar encoding schemes use single voltage level to represent data. In this case, to represent binary 1, high voltage is transmitted and to represent 0, no voltage is transmitted. It is also called Unipolar-Non-return-to-zero, because there is no rest condition i.e. it either represents 1 or 0.



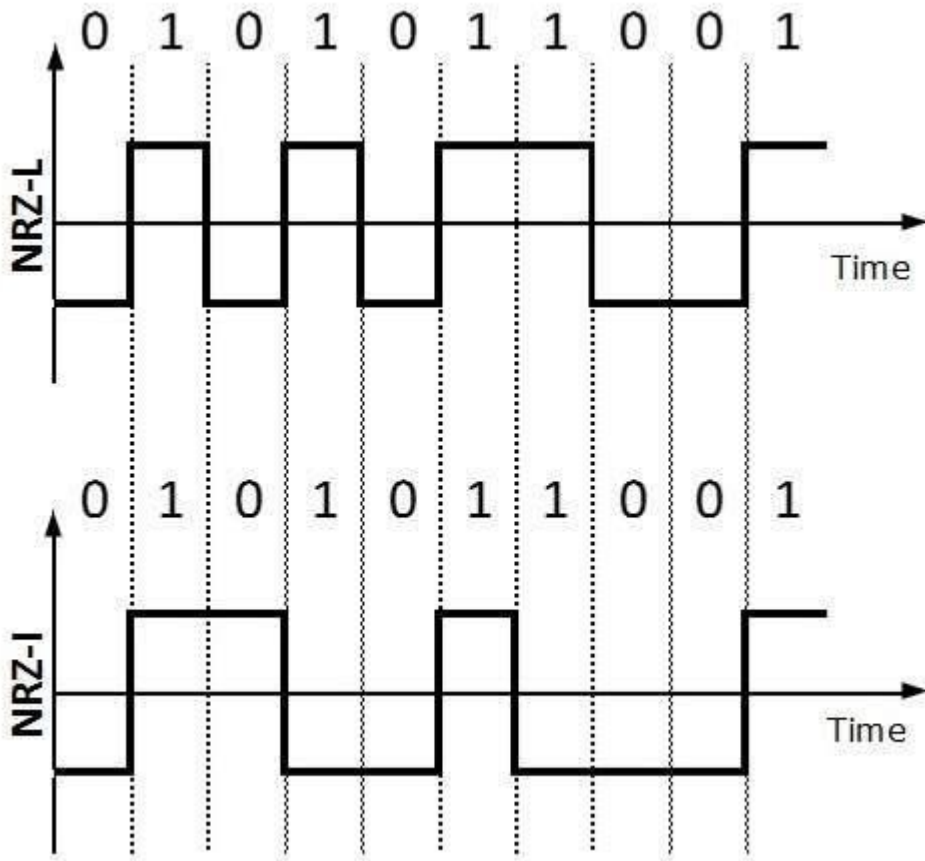
Polar Encoding

Polar encoding scheme uses multiple voltage levels to represent binary values. Polar encodings is available in four types:

- Polar Non-Return to Zero (Polar NRZ)

It uses two different voltage levels to represent binary values. Generally, positive voltage represents 1 and negative value represents 0. It is also NRZ because there is no rest condition.

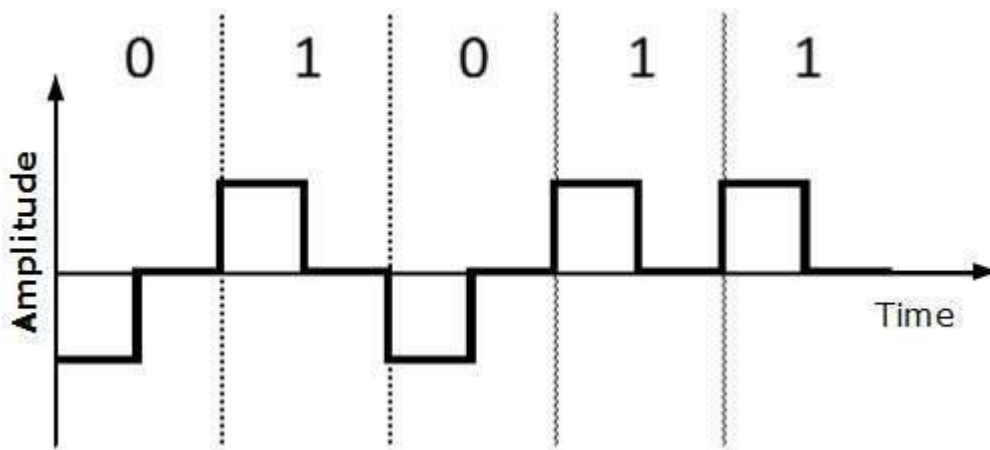
NRZ scheme has two variants: NRZ-L and NRZ-I.



NRZ-L changes voltage level at when a different bit is encountered whereas NRZ-I changes voltage when a 1 is encountered.

- Return to Zero (RZ)

Problem with NRZ is that the receiver cannot conclude when a bit ended and when the next bit is started, in case when sender and receiver's clock are not synchronized.



RZ uses three voltage levels, positive voltage to represent 1, negative voltage to represent 0 and zero voltage for none. Signals change during bits not between bits.

- Manchester

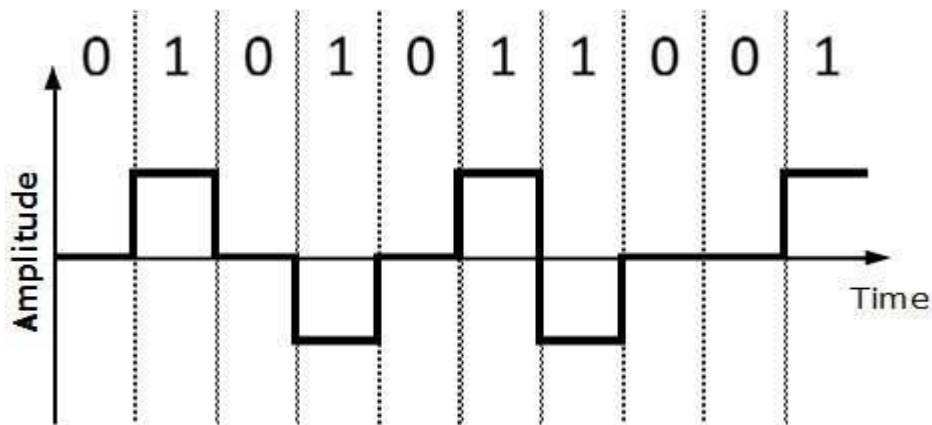
This encoding scheme is a combination of RZ and NRZ-L. Bit time is divided into two halves. It transits in the middle of the bit and changes phase when a different bit is encountered.

- Differential Manchester

This encoding scheme is a combination of RZ and NRZ-I. It also transit at the middle of the bit but changes phase only when 1 is encountered.

Bipolar Encoding

Bipolar encoding uses three voltage levels, positive, negative and zero. Zero voltage represents binary 0 and bit 1 is represented by altering positive and negative voltages.



Block Coding

To ensure accuracy of the received data frame redundant bits are used. For example, in even-parity, one parity bit is added to make the count of 1s in the frame even. This way the original number of bits is increased. It is called Block Coding.

Block coding is represented by slash notation, mB/nB . Means, m -bit block is substituted with n -bit block where $n > m$. Block coding involves three steps:

- Division,
- Substitution
- Combination.

After block coding is done, it is line coded for transmission.

Analog-to-Digital Conversion

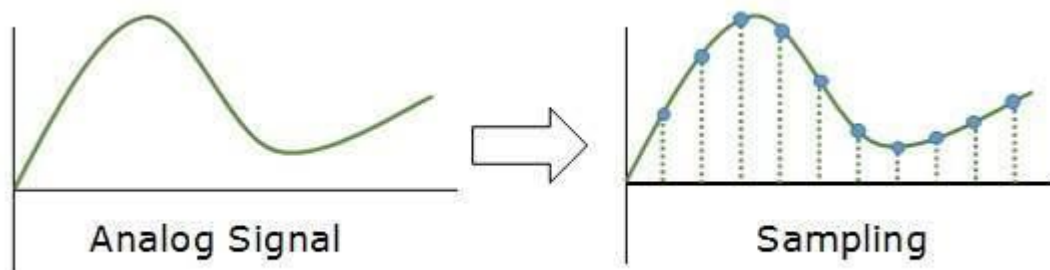
Microphones create analog voice and camera creates analog videos, which are treated as analog data. To transmit this analog data over digital signals, we need analog to digital conversion.

Analog data is a continuous stream of data in the wave form whereas digital data is discrete. To convert analog wave into digital data, we use Pulse Code Modulation (PCM).

PCM is one of the most commonly used methods to convert analog data into digital form. It involves three steps:

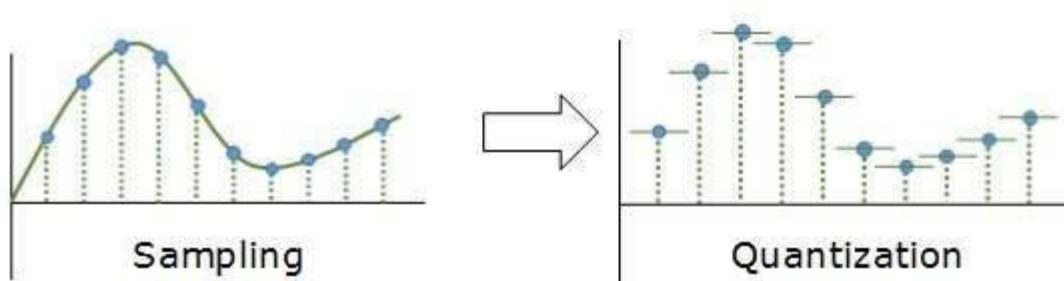
- Sampling
- Quantization
- Encoding.

Sampling



The analog signal is sampled every T interval. Most important factor in sampling is the rate at which analog signal is sampled. According to Nyquist Theorem, the sampling rate must be at least two times of the highest frequency of the signal.

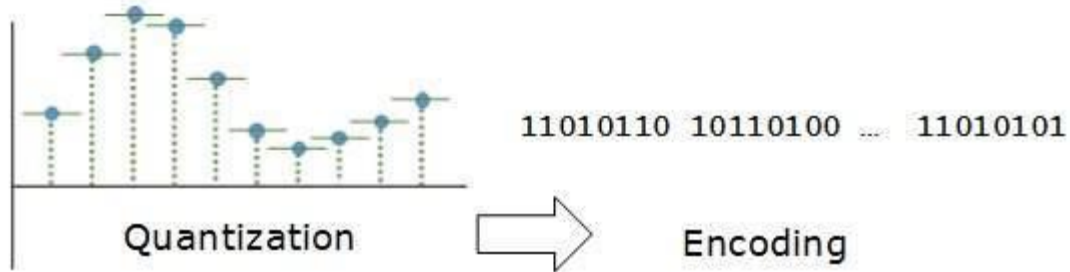
Quantization



Sampling yields discrete form of continuous analog signal. Every discrete pattern shows the amplitude of the analog signal at that instance. The quantization is done between the

maximum amplitude value and the minimum amplitude value. Quantization is approximation of the instantaneous analog value.

Encoding

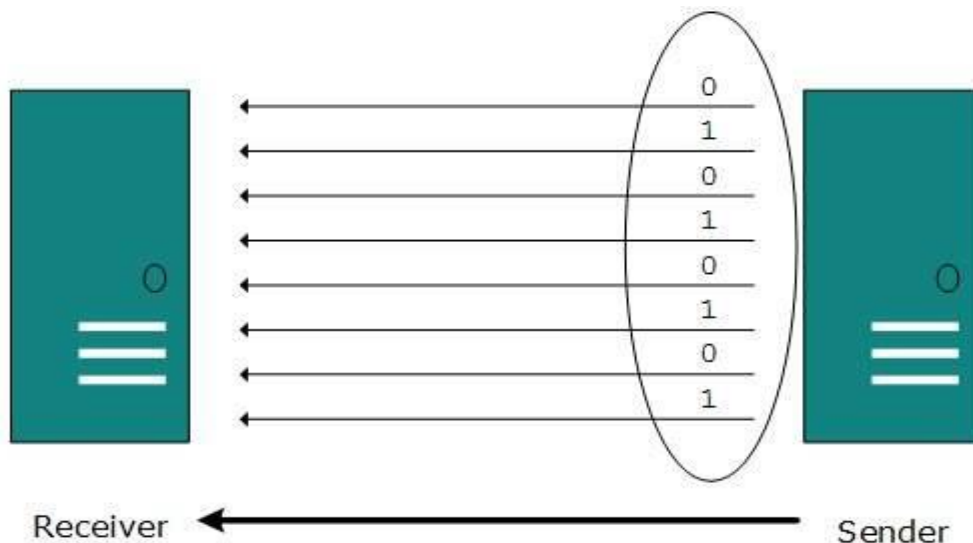


In encoding, each approximated value is then converted into binary format.

Transmission Modes

The transmission mode decides how data is transmitted between two computers. The binary data in the form of 1s and 0s can be sent in two different modes: Parallel and Serial.

Parallel Transmission

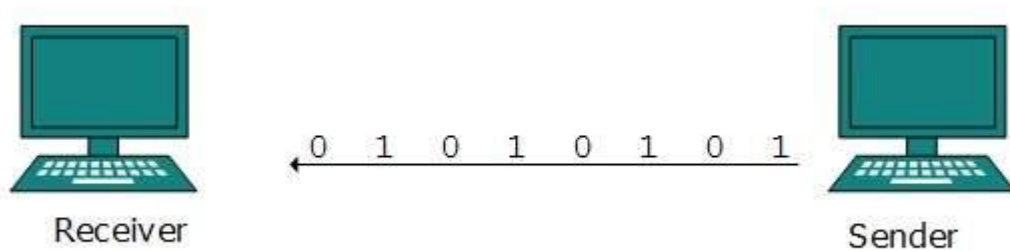


The binary bits are organized in-to groups of fixed length. Both sender and receiver are connected in parallel with the equal number of data lines. Both computers distinguish between high order and low order data lines. The sender sends all the bits at once on all lines. Because the data lines are equal to the number of bits in a group or data frame, a complete group of bits (data frame) is sent in one go. Advantage of Parallel transmission is

high speed and disadvantage is the cost of wires, as it is equal to the number of bits sent in parallel.

Serial Transmission

In serial transmission, bits are sent one after another in a queue manner. Serial transmission requires only one communication channel.



Serial transmission can be either asynchronous or synchronous.

Asynchronous Serial Transmission

It is named so because there's no importance of timing. Data-bits have specific pattern and they help receiver recognize the start and end data bits. For example, a 0 is prefixed on every data byte and one or more 1s are added at the end.

Two continuous data-frames (bytes) may have a gap between them.

Synchronous Serial Transmission

Timing in synchronous transmission has importance as there is no mechanism followed to recognize start and end data bits. There is no pattern or prefix/suffix method. Data bits are sent in burst mode without maintaining gap between bytes (8-bits). Single burst of data bits may contain a number of bytes. Therefore, timing becomes very important.

It is up to the receiver to recognize and separate bits into bytes. The advantage of synchronous transmission is high speed, and it has no overhead of extra header and footer bits as in asynchronous transmission.

3.5 ANALOG TRANSMISSION

To send the digital data over an analog media, it needs to be converted into analog signal. There can be two cases according to data formatting.

Bandpass:The filters are used to filter and pass frequencies of interest. A bandpass is a band of frequencies which can pass the filter.

Low-pass: Low-pass is a filter that passes low frequencies signals.

When digital data is converted into a bandpass analog signal, it is called digital-to-analog conversion. When low-pass analog signal is converted into bandpass analog signal, it is called analog-to-analog conversion.

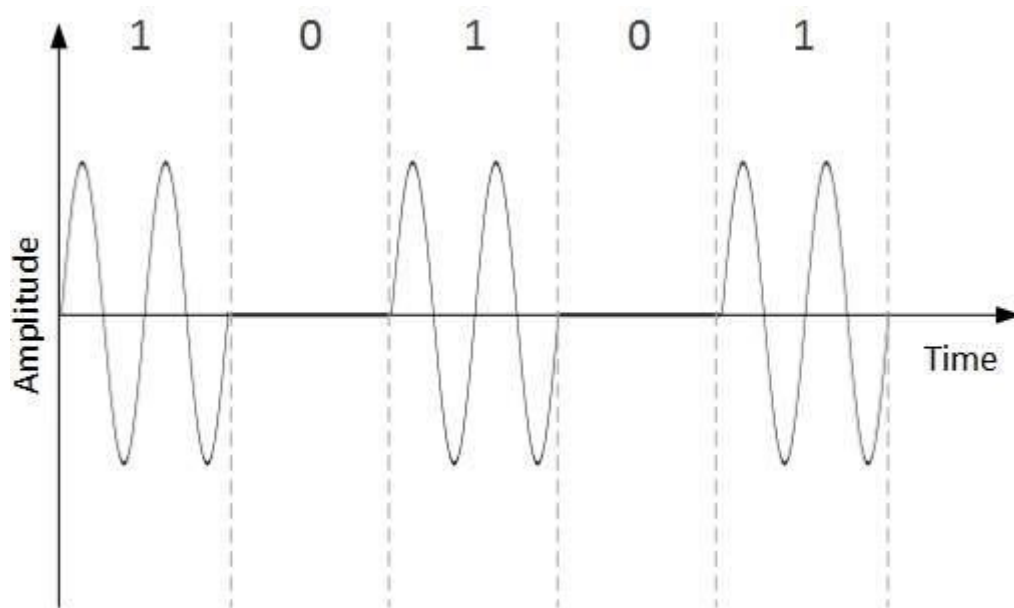
Digital-to-Analog Conversion

When data from one computer is sent to another via some analog carrier, it is first converted into analog signals. Analog signals are modified to reflect digital data.

An analog signal is characterized by its amplitude, frequency, and phase. There are three kinds of digital-to-analog conversions:

- **Amplitude Shift Keying**

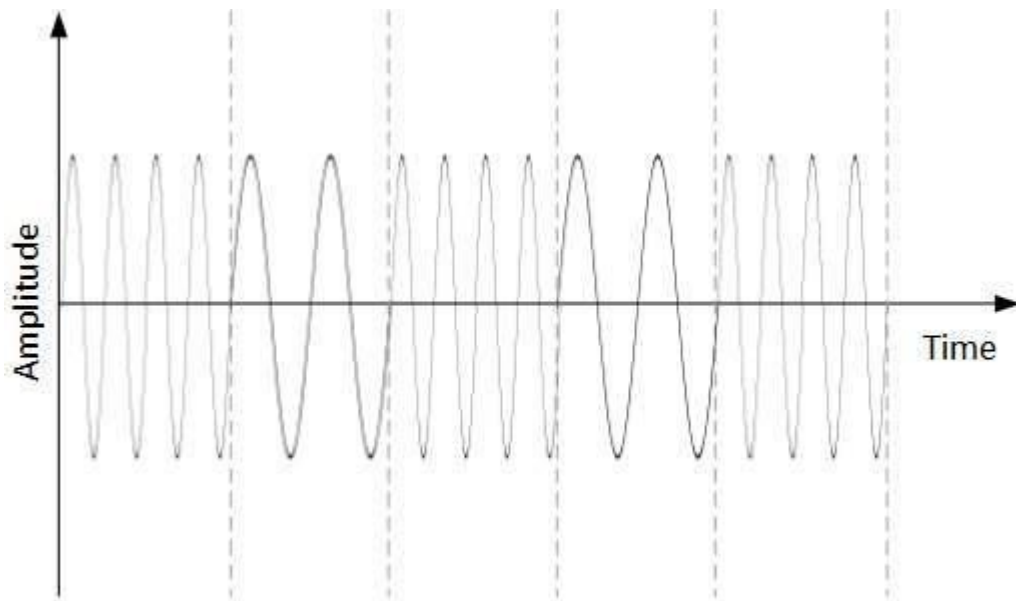
In this conversion technique, the amplitude of analog carrier signal is modified to reflect binary data.



When binary data represents digit 1, the amplitude is held; otherwise it is set to 0. Both frequency and phase remain same as in the original carrier signal.

- **Frequency Shift Keying**

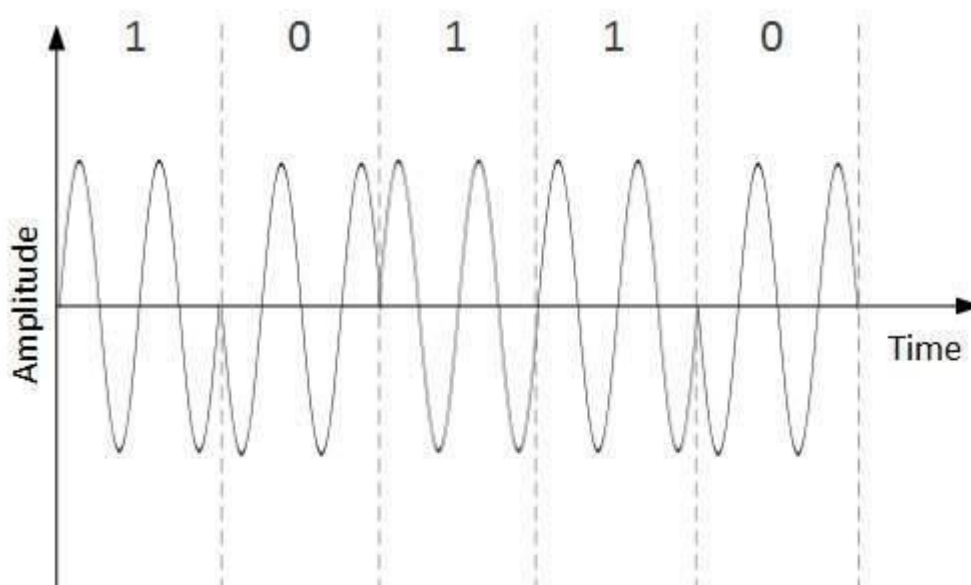
In this conversion technique, the frequency of the analog carrier signal is modified to reflect binary data.



This technique uses two frequencies, f_1 and f_2 . One of them, for example f_1 , is chosen to represent binary digit 1 and the other one is used to represent binary digit 0. Both amplitude and phase of the carrier wave are kept intact.

- **Phase Shift Keying**

In this conversion scheme, the phase of the original carrier signal is altered to reflect the binary data.



When a new binary symbol is encountered, the phase of the signal is altered. Amplitude and frequency of the original carrier signal is kept intact.

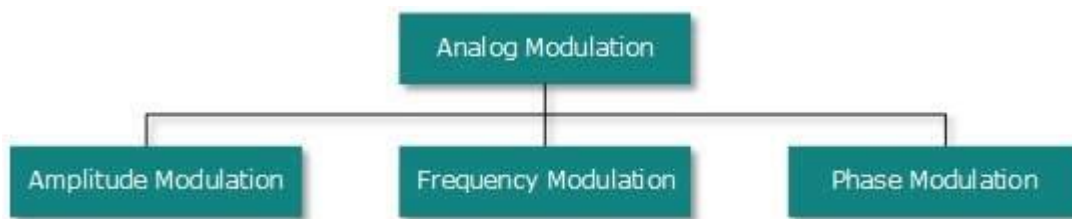
- **Quadrature Phase Shift Keying**

QPSK alters the phase to reflect two binary digits at once. This is done in two different phases. The main stream of binary data is divided equally into two sub-

streams. The serial data is converted in to parallel in both sub-streams and then each stream is converted to digital signal using NRZ technique. Later, both the digital signals are merged together.

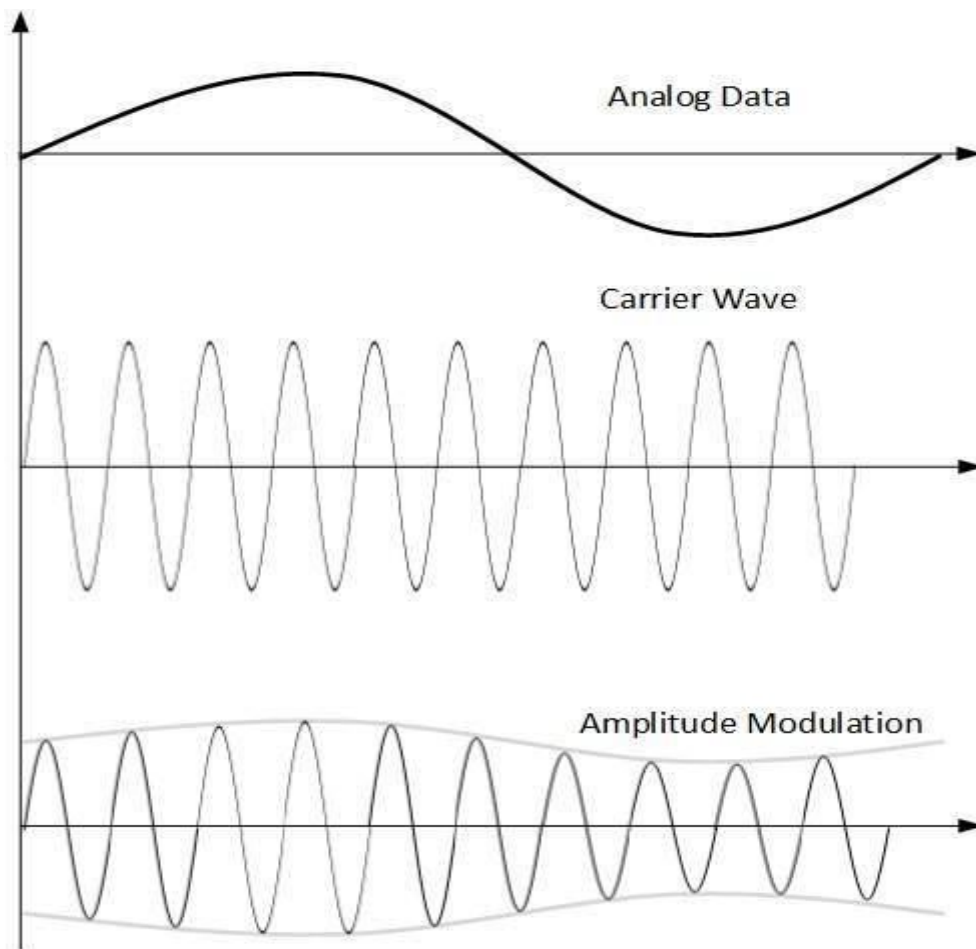
Analog-to-Analog Conversion

Analog signals are modified to represent analog data. This conversion is also known as Analog Modulation. Analog modulation is required when bandpass is used. Analog to analog conversion can be done in three ways:



- **Amplitude Modulation**

In this modulation, the amplitude of the carrier signal is modified to reflect the analog data.

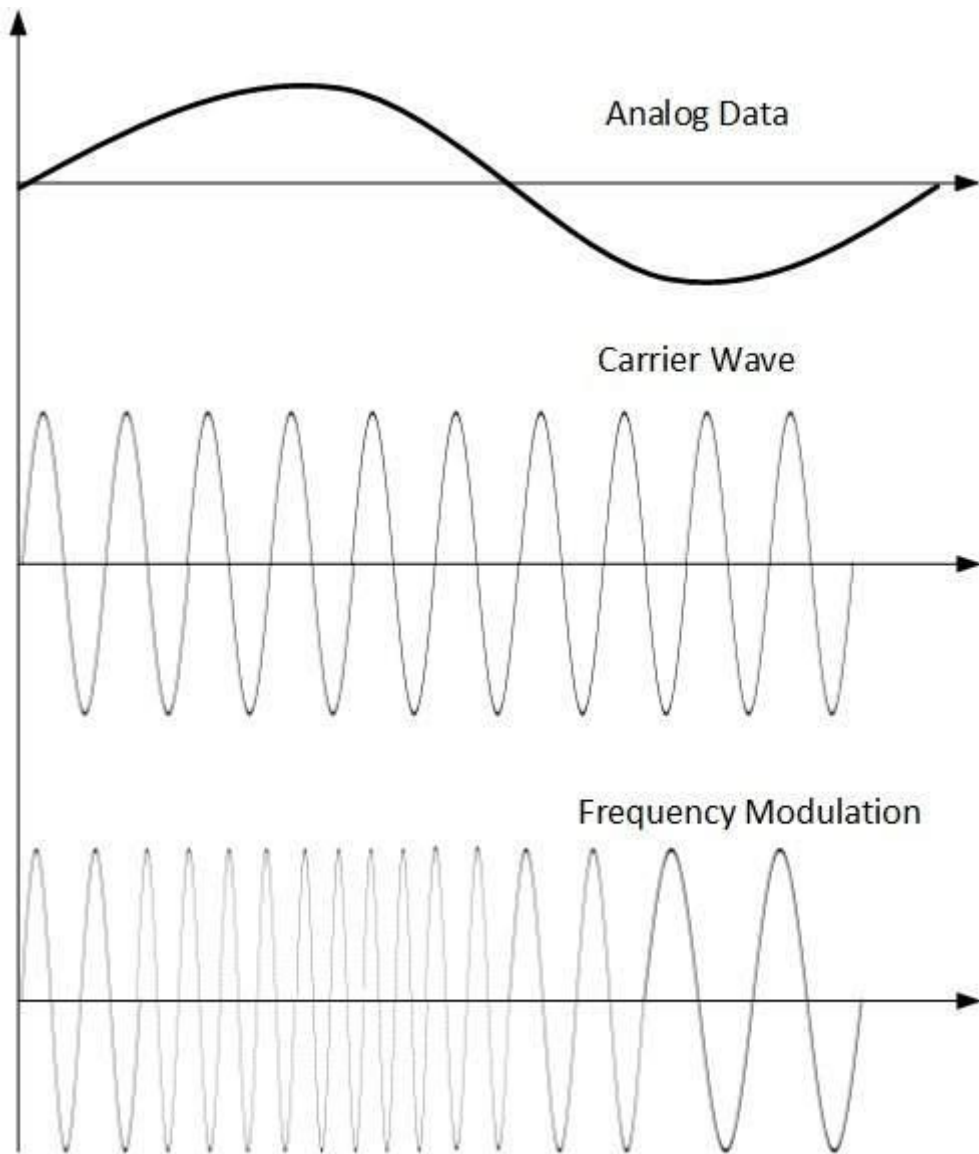


Amplitude modulation is implemented by means of a multiplier. The amplitude of modulating signal (analog data) is multiplied by the amplitude of carrier frequency, which then reflects analog data.

The frequency and phase of carrier signal remain unchanged.

- **Frequency Modulation**

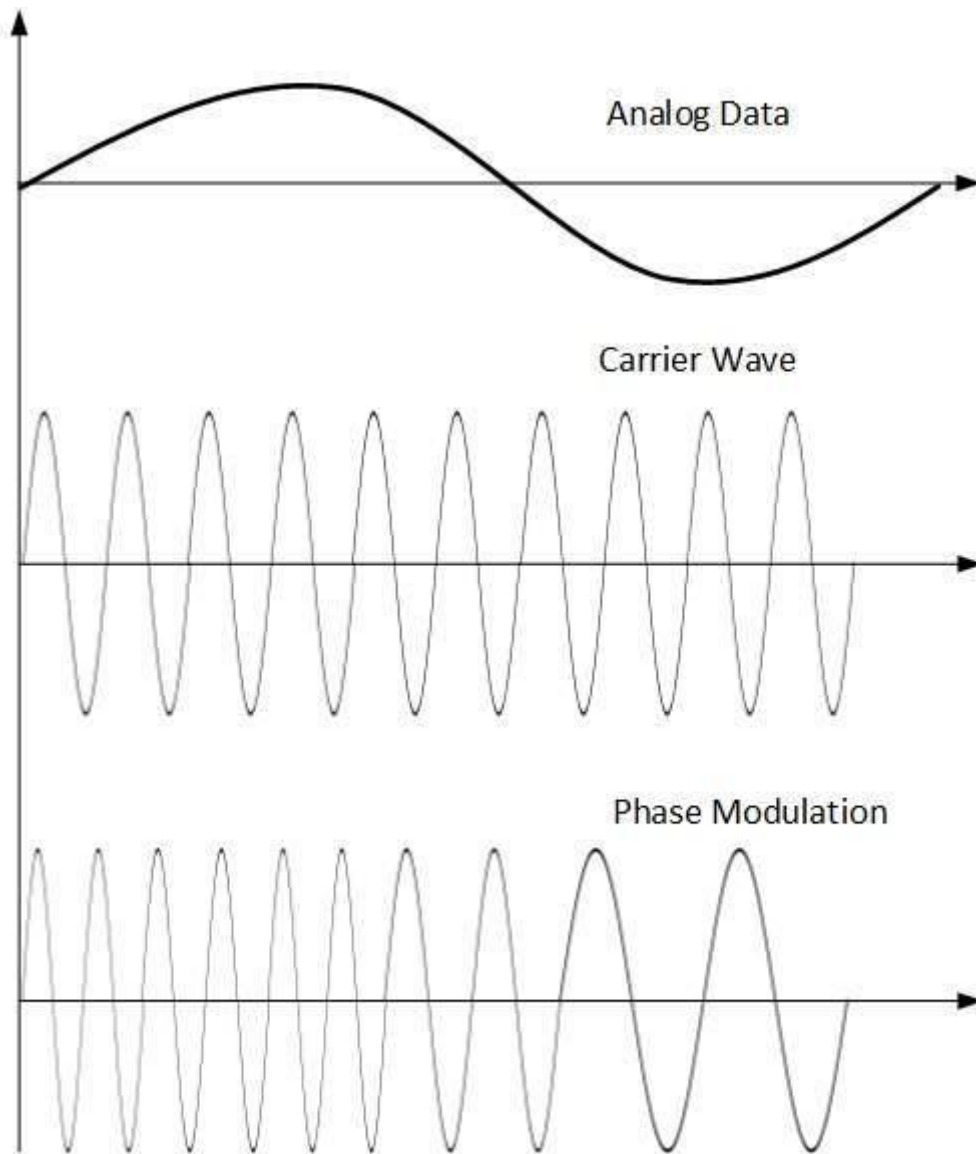
In this modulation technique, the frequency of the carrier signal is modified to reflect the change in the voltage levels of the modulating signal (analog data).



The amplitude and phase of the carrier signal are not altered.

- **Phase Modulation**

In the modulation technique, the phase of carrier signal is modulated in order to reflect the change in voltage (amplitude) of analog data signal.



Phase modulation is practically similar to Frequency Modulation, but in Phase modulation frequency of the carrier signal is not increased. Frequency of carrier is signal is changed (made dense and sparse) to reflect voltage change in the amplitude of modulating signal.

3.6 SUMMARY

In this unit we have discussed in detail about physical layer and also covered data and signals. We also explained about digital transmission. At the end of this unit also covered in detail analog transmission.

3.7 KEYWORDS

Analog signal, Digital signal, Noise, Switching, Multiplexing, Sampling, Quantization and Encoding.

3.8 QUESTIONS

1. Define analog and digital signal.
2. Describe line coding.
3. Elucidate amplitude shift keying.
4. Briefly explain analog to digital conversion.

3.9 REFERENCES

- William Stallings, Data and Computer Communication, Prentice Hall of India
- Behrouz A. Forouzan, Data Communication and Networking, McGraw-Hill
- Andrew S. Tanenbaum, Computer Networks, Prentice Hall.

Structure

- 4.0 Objectives
- 4.1 Introduction
- 4.2 Data Link Layer
- 4.3 Error detection and Correction
- 4.4 Data link Control
- 4.5 Multiple Access Protocols
- 4.6 Virtual Circuit Switching
- 4.7 Frame relay
- 4.8 ATM
- 4.9 Summary
- 4.10 Keywords
- 4.11 Questions and answers
- 4.12 References

4.0 OBJECTIVES

At the end of this unit, you should be able to

- Discuss data link layer
- Describe error detection and correction
- Explain multiple access protocols
- Elucidate virtual circuit switching
- Discuss frame relay and ATM

4.1 INTRODUCTION

The Data-link layer is the second layer from the bottom in the OSI (Open System Interconnection) network architecture model. It is responsible for the node-to-node delivery of data. Its major role is to ensure error-free transmission of information. DLL is also responsible to encode, decode and organize the outgoing and incoming data. This is considered the most complex layer of the OSI model as it hides all the underlying complexities of the hardware from the other above layers.

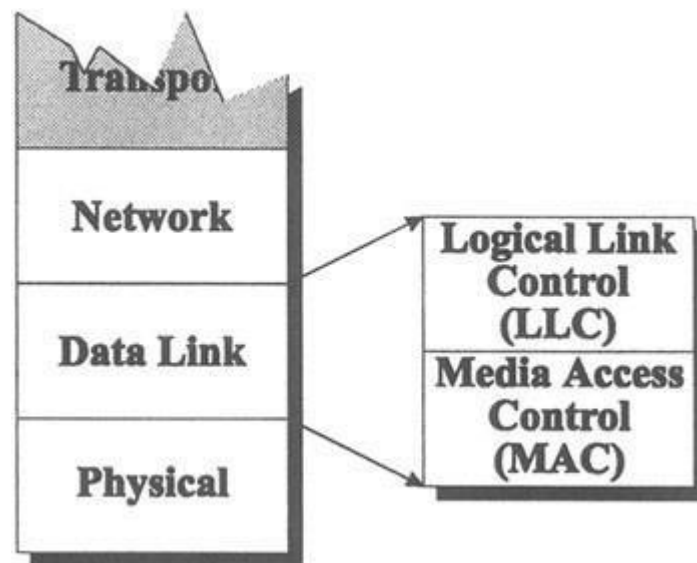
4.2 DATA LINK LAYER

FunctionsofDataLinkLayer:

The datalink layer transforms the physical layer, a raw transmission facility, to a link responsible for

ode-to-node (hop-to-hop) communication. Specific responsibilities of the data link layer include framing, addressing, flow control, error control, and media access control. The data link layer divides the stream of bits received from the network layer into manageable data units called frames. The data link layer adds a header to the frame to define the addresses of the sender and receiver of the frame. 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. The data link layer also adds reliability to the physical layer by adding mechanisms to detect and retransmit damaged, duplicate, or lost frames. 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.

Overview of Logical Link Control (LLC) and Media Access Control (MAC):



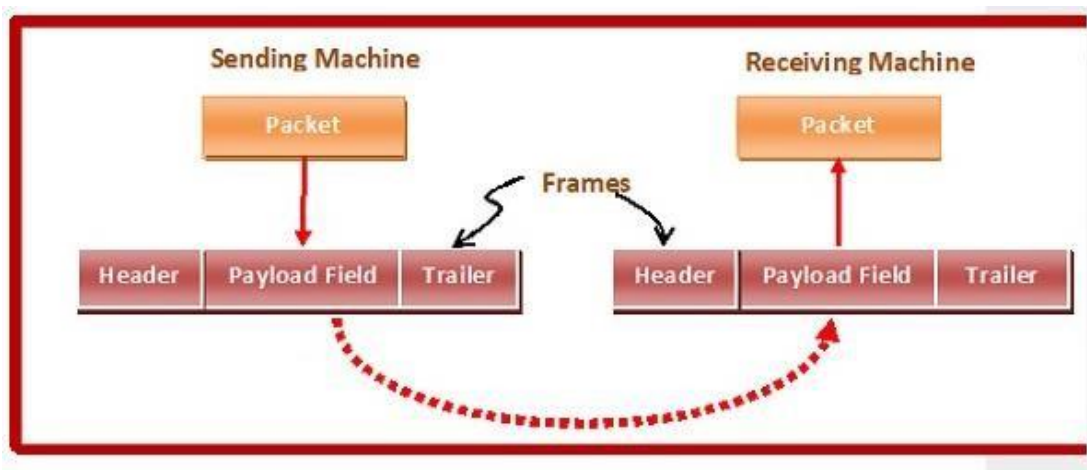
The data link layer is divided into two sublayers:

- The LLC sublayer acts as an interface between the media access control (MAC) sublayer and the network layer. Logical Link Control (LLC) sublayer provides the logic for the data link. The LLC sublayer provides multiplexing mechanisms that make it possible for several network protocols to coexist within a multipoint network and to be transported over the same network medium. Thus, it controls the synchronization, flow control, and error checking functions of the data link layer.
- Media Access Control (MAC) sublayer provides control for accessing the transmission medium. It is responsible for moving data packets from one network

interface to another, across a shared transmission medium. Physical addressing is handled at the MAC sublayer. When sending data to another device on the network, the MAC sublayer encapsulates higher-level frames into frames appropriate for the transmission medium, adds a frame check sequence to identify transmission errors, and then forwards the data to the physical layer. Additionally, the MAC is also responsible for compensating for collisions by initiating retransmission if a jam signal is detected.

Framing:

In the physical layer, data transmission involves synchronized transmission of bits from the source to the destination. The data link layer packs these bits into frames. Data-link layer takes the packets from the Network Layer and encapsulates them into frames. If the frame size becomes too large, then the packet may be divided into small sized frames. Smaller sized frames make flow control and error control more efficient. Then, it sends each frame bit-by-bit on the hardware. At receiver's end, data link layer picks up signals from hardware and assembles them into frames.



Partsofa Frame

A frame has the following parts—

Frame Header— It contains the source and the destination addresses of the frame.

Payload field – It contains the message to be delivered.

Trailer – It contains the error detection and error correction bits. Flag – It marks the beginning and end of the frame.



Types of Framing

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

Fixed-sized Framing

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

Example – ATM cells.

Variable-Sized Framing

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

It is used in local area networks.

Flow Control Mechanisms:

Flow control coordinates the amount of data that can be sent before receiving an acknowledgment and is one of the most important duties of the data link layer. In most protocols, flow control is a set of procedures that tell the sender how much data it can transmit before it must wait for an acknowledgment from the receiver. The flow of data must not be allowed to overwhelm the receiver. Any receiving device has a limited speed at which it can process incoming data and a limited amount of memory in which to store incoming data. The receiving device must be able to inform the sending device before those limits are reached and to request that the transmitting device

send fewer frames or stop temporarily. Incoming data must be checked and processed before they can be used. The rate of such processing is often slower than the rate of transmission. For this reason, each receiving device has a block of memory, called a buffer, reserved for storing incoming data until they are processed. If the buffer begins to fill up, the receiver must be able to tell the sender to halt transmission until it is once again able to receive.

When a data frame is sent from one host to another over a single medium, it is required that the sender and receiver should work at the same speed. That is, sender sends at a speed on which the receiver can process and accept the data. If sender is sending too fast, the receiver may be overloaded and data may be lost.

Two types of mechanisms can be deployed to control the flow:

- A simple stop and wait Protocol

Sliding Window Protocol

4.3 ERROR DETECTION AND CORRECTION TECHNIQUES:

Networks must be able to transfer data from one device to another with acceptable accuracy. For most applications, a system must guarantee that the data received are identical to the data transmitted.

Any time data are retransmitted from one node to the next, they can become corrupted in passage. Many factors can alter one or more bits of a message. Some applications require a mechanism for detecting and correcting errors.

Some applications can tolerate a small level of error. For example, random errors in audio or video transmissions may be tolerable, but when we transfer text, we expect a very high level of accuracy.

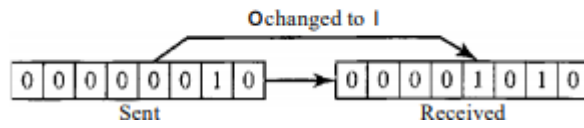
Types of Errors:

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. In a single-bit error, a 0 is changed to a 1 or a 1 to a 0. In a burst error, multiple bits are changed.

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.

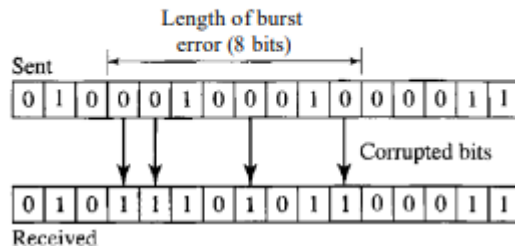
Single-bit error



BurstError:

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.

Burst error of length 8



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. We are not even interested in the number of errors. A single-bit error is the same for us as a burst error. 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. If we need to correct one single error in an 8-bit data unit, we need to consider eight possible error locations; if we need to correct two errors in a data unit of the same size, we need to consider 28 possibilities.

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 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.

Error Detecting Codes

Whenever a message is transmitted, it may get scrambled by noise or data may get corrupted. To avoid this, we use error-detecting codes which are additional data added to a given digital message to help us detect if any error has occurred during transmission of the message.

Some popular techniques for error detection are:

- Parity
- Checksum
- Cyclic redundancy check

Parity check

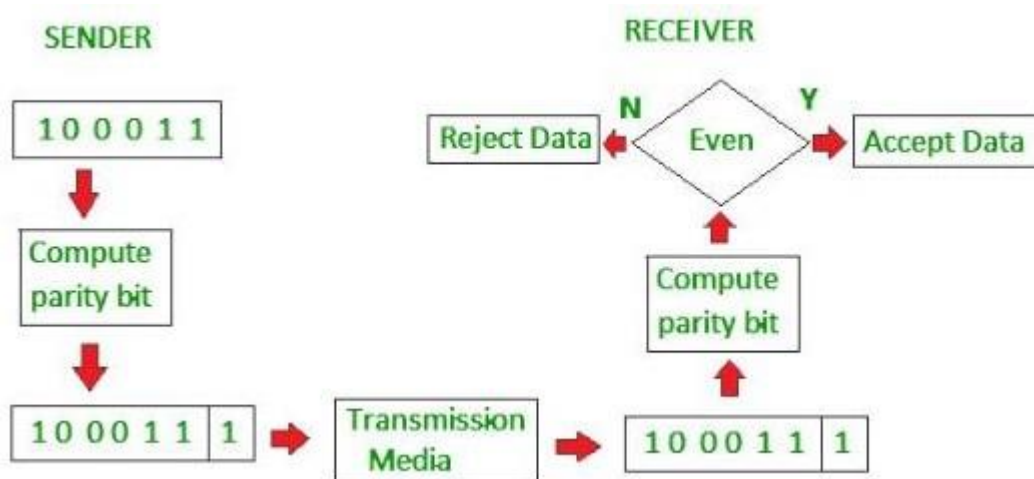
The most common and least expensive mechanism for error-detection is the parity check. The parity check is done by adding an extra bit, called parity bit to the data to make a number of 1s either even in case of even parity or odd in case of odd parity. While creating a frame, the sender counts the number of 1s in it and adds the parity bit in the following way:

- In case of even parity: If a number of 1s is even then parity bit value is 0. If the number of 1s is odd, then parity bit value is 1.
- In case of odd parity: If a number of 1s is odd then parity bit value is 0. If a number of 1s is even, then parity bit value is 1.

then parity bit value is 1.

On receiving a frame, the receiver counts the number of 1s in it. In case of even parity check, if the count of 1s is even, the frame is accepted, otherwise, it is rejected. A similar rule is adopted for odd parity check.

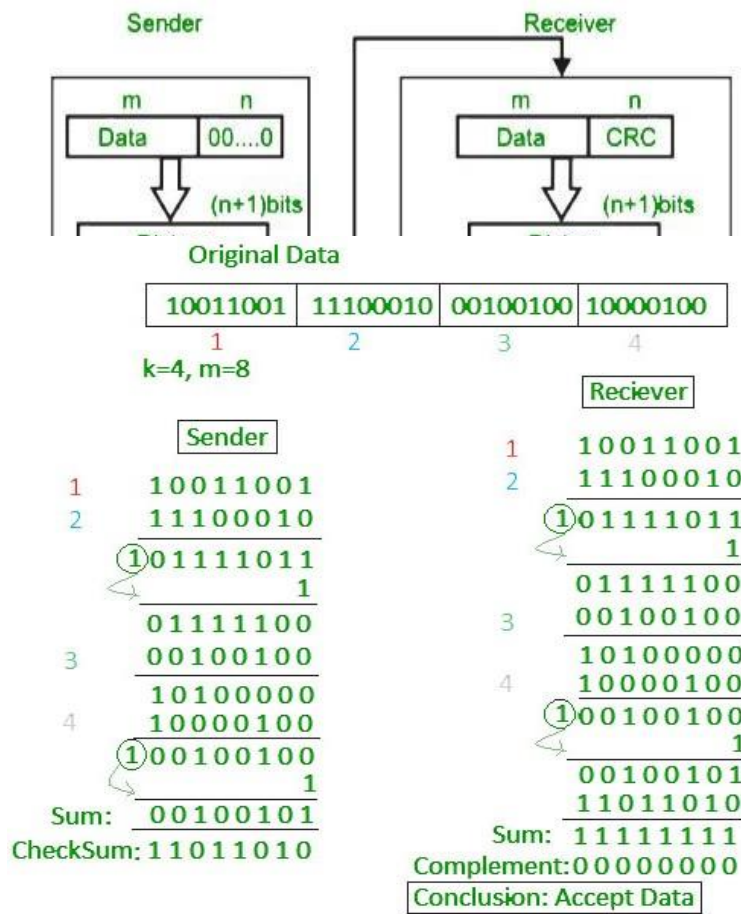
The parity check is suitable for single bit error detection only.



Checksum

In this error detection scheme, the following procedure is applied:

- Data is divided into fixed sized frames or segments. (k segment each of m bits)
- The sender adds these segments using 1's complement arithmetic to get the sum. It then complements the sum to get the checksum and sends it along with the data frames.
- The receiver adds the incoming segments along with the checksum using 1's complement arithmetic to get the sum and then complements it.



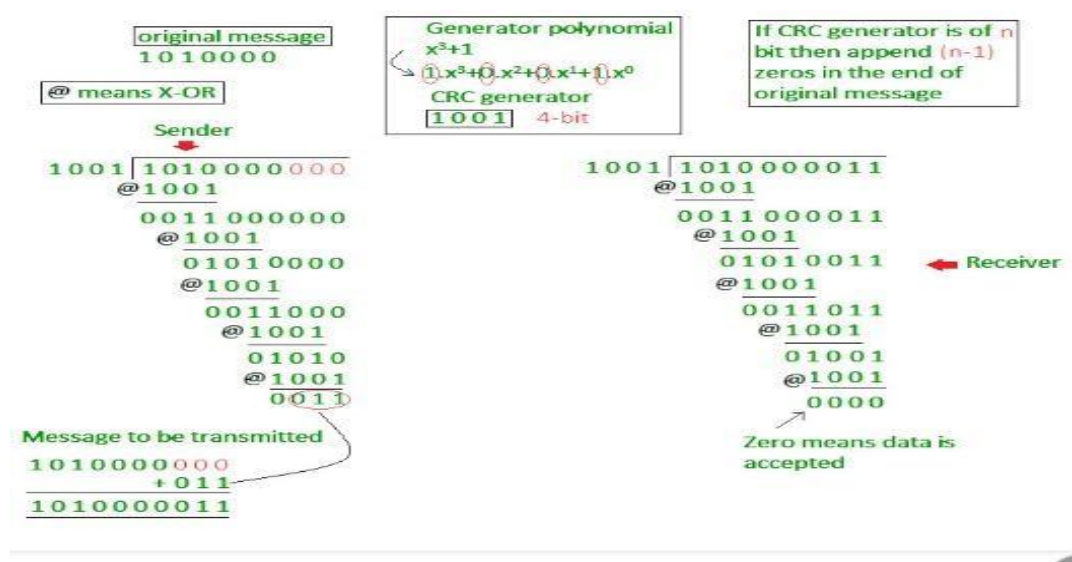
- If the result is zero, the received frames are accepted; otherwise, they are discarded.

Cyclic Redundancy Check:

A cyclic redundancy check (CRC) is an error-

detecting code commonly used in digital networks and storage devices to detect accidental changes to raw data. Unlike checksum scheme, which is based on addition, CRC is based on binary division. In CRC, a sequence of redundant bits, called cyclic redundancy check bits, are appended to the end of data units so that the resulting data unit becomes exactly divisible by a second, predetermined binary number. At the destination, the incoming data unit is divided by the same number. If at this step there is no remainder, the data unit is assumed to be correct and is therefore accepted. A remainder indicates that the data unit has been damaged in transit and therefore must be rejected. At the sender's side, the data unit to be transmitted is divided by a predetermined divisor (binary number) in order to obtain the remainder. This remainder is called CRC. The CRC has one bit less than the divisor. It means that if CRC is of n bits, divisor is of $n+1$ bit. The sender appends this CRC to the end of data units such that the resulting data unit becomes exactly divisible by predetermined divisor i.e. remainder becomes zero. At the destination, the incoming data unit i.e. data + CRC is divided by the same number (predetermined binary divisor). If the remainder after division is zero then there is no error in the data unit &

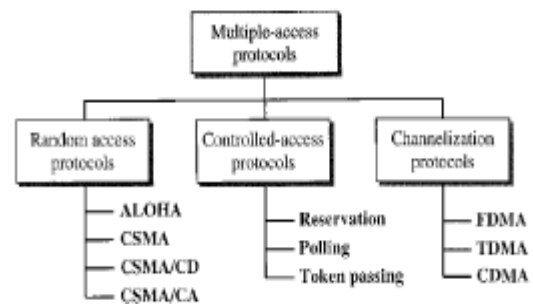
receiver accepts it. If remainder after division is not zero, it indicates that the data unit has been damaged in transit and therefore it is rejected. This technique is more powerful than the parity check and checksum error detection.



4.5 MULTIPLE ACCESS PROTOCOLS (CHANNEL ALLOCATION TECHNIQUES)

When nodes or stations are connected and use a common link, called a multipoint or broadcast link, we need a multiple access protocol to coordinate access to the link. Many formal protocols have been

Taxonomy of multiple-access protocols discussed in this chapter



designed to handle access to a shared link. We categorize them into three groups.

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 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 testing of the state of the medium.

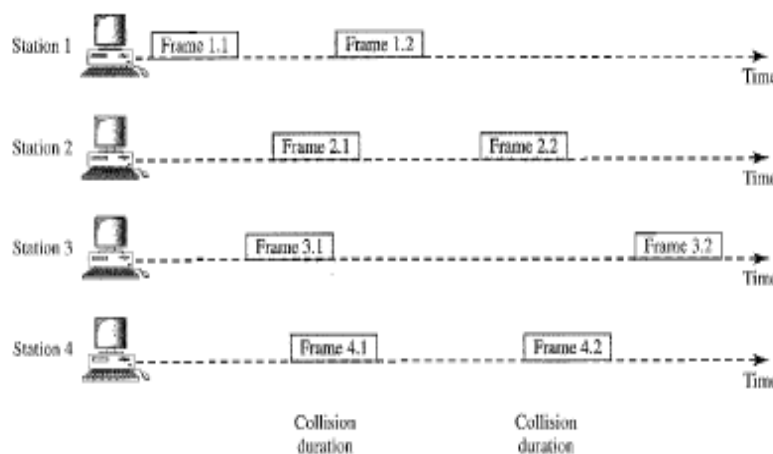
ALOHA:

ALOHA is the earliest random-access method developed for wireless LAN but can be used on any shared medium. In this, multiple stations can transmit data at the same time and can hence lead to collision. The data from the two stations collide.

Pure ALOHA:

The idea behind this protocol 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 possibility of collision between frames from different stations. Even if one bit of a frame coexists on the channel with one bit from

Figure 12.3 *Frames in a pure ALOHA network*



another frame, there is a collision and both will be destroyed.

The pure ALOHA protocol relies on acknowledgements from the receiver. When a station sends a frame, it expects the receiver to send an acknowledgement. If the acknowledgement does not arrive after the time out period, the station assumes that the frame (or the acknowledgement) has been destroyed and resends the frame. A collision involves two or more stations. If all these stations try to resend their

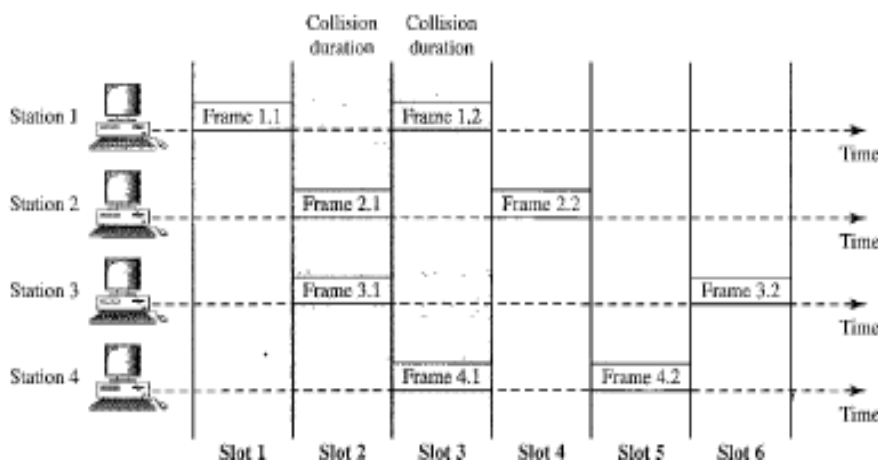
frames after the timeout period passes, each station waits a random amount of time before resending its frame. The randomness will help avoid more collisions, called back-off time. Since different stations may wait for different amount of time, the probability of further collision decreases.

Slotted ALOHA:

In pure ALOHA, there is no rule that defines when the station can send. A station may send soon after another station has started or soon before another station has finished. So, still the collision may occur.

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

Figure 12.6 Frames in a slotted ALOHA network



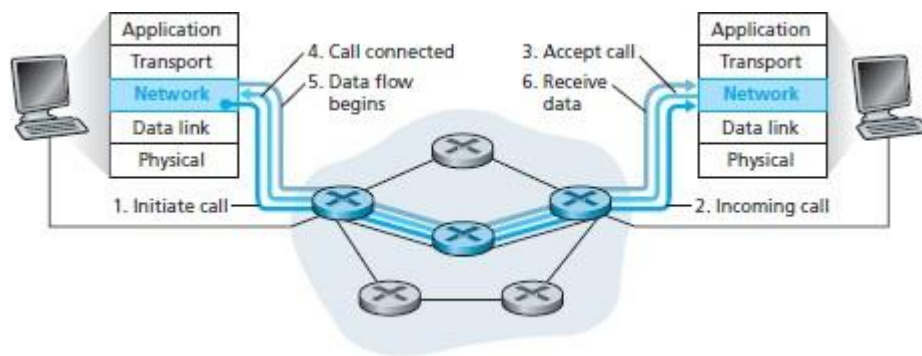
Allowing a station to send only at the beginning of the time slot means that the station sending in the previous slot has finished sending its frame already. However, there is still possibility of collision if two stations try to send at the beginning of the same time slot.

4.6 VIRTUAL CIRCUIT SWITCHING

Virtual circuit switching is a packet switching methodology whereby a path is established between the source and the final destination through which all the packets will be routed during a call. This path is called a virtual circuit because to the user, the connection appears to be a dedicated physical

circuit. However, other communications may also be sharing the parts of the same path. So, virtual circuit packet switching is connection oriented.

Before the data transfer begins, the source and destination identify a suitable path for the virtual circuit. All intermediate nodes between the two points put an entry of the routing in their routing table for the call. Additional parameters, such as the maximum packet size, are also exchanged between the source and the destination during call setup. The virtual circuit is cleared after the data transfer is completed.



Advantages of virtual circuit switching are:

- Packets are delivered in order, since they all take the same route;
- The overhead in the packets is smaller, since there is no need for each packet to contain the full address;
- The connection is more reliable, network resources are allocated at call setup so that even during times of congestion, provided that a call has been setup, the subsequent packets should get through;
- Billing is easier, since billing records need only be generated per call and not per packet.

4.7 FRAMERELAY

Frame Relay is a virtual-circuit wide-area network that was designed in response to demands for a new type of WAN. Frame Relay is a wide area network with the following features:

1. Frame Relay operates at a higher speed (1.544 Mbps and recently 44.376 Mbps).
2. Frame Relay operates in just the physical and data link layers. This means it can easily

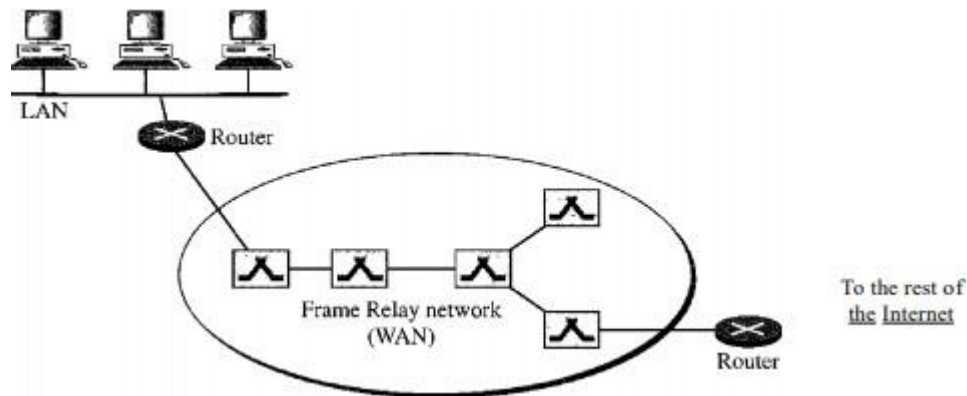
be used as a backbone network to provide services to protocols that already have a network layer protocol, such as the Internet.

3. Frame Relay allows bursty data.

4. Frame Relay allows a frame size of 9000 bytes, which can accommodate all local area network frame sizes.

5. Frame Relay is less expensive than other traditional WANs.

6. Frame Relay has error detection at the data link layer only. There is no flow control or error control. There is not even a retransmission policy if a frame is damaged; it is silently dropped. Frame Relay was designed in this way to provide fast transmission capability for more reliable media and for those protocols that have flow and error control at the higher layers.



Frame relay is a virtual circuit packet switching technology that fragments data into transmission units called frames and sends them in high-speed bursts through a digital network. It establishes an exclusive connection during the transmission period called a virtual connection. Frame relay puts data in a variable-size unit called a frame and leaves any necessary error correction (retransmission of data) up to the endpoints, which speeds up overall data transmission. Configuring user equipment in a Frame Relay network is extremely simple. The connection-oriented service provided by Frame Relay has properties like non-duplication of frames, preservation of the frame transfer order and a small probability of frame loss. The features provided by Frame Relay make it one of the best choices for interconnecting local area networks using a wide area network. However, the drawback in this method is that it becomes prohibitively expensive with growth of the network.

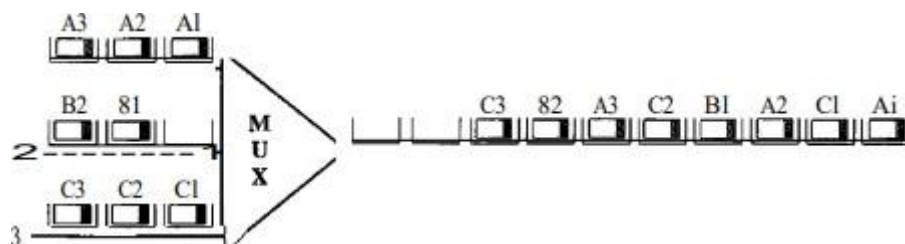
There are certain benefits which are associated with Frame Relay.

- It helps in reducing the cost of internet working, as there is considerable reduction in the number of circuits required and the associated bandwidths.
- It helps in increasing the performance due to reduced network complexity.
- It increases the interoperability with the help of international standards.
- Frame Relay is protocol independent and can easily be used to combine traffic from other network working protocols.

In business scenarios, where there is a slow connection or continuous traffic flow due to applications like multimedia, Frame Relay is not a recommended choice.

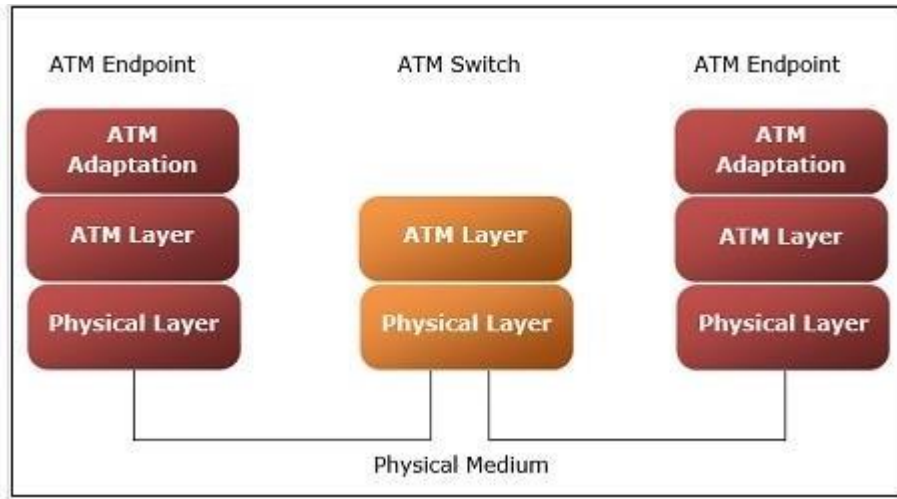
4.8 ATM

Asynchronous transfer mode (ATM) is a switching technique used by telecommunication networks that uses asynchronous time-division multiplexing to encode data into small, fixed-sized cells.



ATM networks are connection-oriented networks that support voice, video and data communications. It encodes data into small fixed-size cells so that they are suitable for TDM and transmits them over a physical medium.

The size of an ATM cell is 53 bytes: 5-byte header and 48-byte payload.



Physical Layer—This layer corresponds to physical layer of OSI model. At this layer, the cells are converted into bit streams and transmitted over the physical medium.

ATM Layer—This layer is comparable to data link layer of OSI model. It accepts the 48-byte segments from the upper layer, adds a 5-byte header to each segment and converts into 53-byte cells. This layer is responsible for routing of each cell, traffic management, multiplexing and switching.

ATM Adaptation Layer—This layer corresponds to network layer of OSI model. It provides facilities to the existing packet switched networks to connect to ATM network and use its services. It accepts the data and converts them into fixed sized segments.

Benefits of ATM Networks are

- It provides the dynamic bandwidth that is particularly suited for bursty traffic.
- Since all data are encoded into identical cells, data transmission is simple, uniform and predictable.
- Uniform packet size ensures that mixed traffic is handled efficiently.
- Small sized header reduces packet overhead, thus ensuring effective bandwidth usage.
- ATM networks are scalable both in size and speed.

4.9 SUMMARY

In this unit, we were able to see about the concept of data link layer. We were also able to clearly discuss on the importance error correction and detection. And also know n about the concepts of data link control and multiple access. At the end of this unit discussed about virtual circuit networks such as frame relay and ATM.

4.10 KEYWORDS

LLC, Framing, Burst error, CRC, ALOHA, Frame relay and ATM

4.11 QUESTIONS

1. Describe cyclic redundancy check.
2. Explain briefly ALOHA.
3. Write a short note on ATM
4. Discuss frame relay.
5. Explain virtual-circuit switching

4.12 REFERENCES

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