Contents

Unit 2
Protocol Layering

Structure of the Unit

2.1 Unit Outcomes

After the successful completion of this unit, the student will be able to:

- Describe the OSI reference model and the TCP/IP protocol stack.
- Differentiate between the Circuit and Packet Switching networks.
- Define the different data transmission delays.

2.2 Layered Communication Models

We already know that a communication system consists of hardware, the physical components constituting the system, and the software or the programs used to control the data communication through it. This system is very complex, and implementing it as a single and unstructured component is very difficult. To overcome this complexity, the International Organization of Standardization (ISO) has developed a layered approach and has given a reference model on which the present-day networks are built. This model conceptualizes and divides the network functionality into layers, where each layer is assigned a specific task.

Fig. 2.1 shows a generic layered communication model. Here every layer is built to perform some part of the communication process and every layer uses a protocol that is suitable and is needed to communicate the data. The protocols at every layer at the sending end communicate with the protocols at the corresponding layer at the receiving end, and data is sent transferred between these protocols with the help of the protocols in the layers below it.

At the sending end, the top-layer protocol receives its data from the application program, processes it, and sends it to the same protocol running at the corresponding layer at the receiving end. This protocol passes the data to the application program at the receiver. (Server in the figure). This is basically a logical connection. The protocols running at the intermediate layers receive the data from the layer above, process it, and pass it to the layer below. The protocol at the bottom layer converts the data into suitable signals for transmission on the medium. The signals are electrical in nature for Ethernet LANs, electromagnetic signals for WiFi, or pulses of light for optical fibers.

At the receiving end, the protocol at the bottom layer converts the signals back into data and passes them to the layer above it. The protocols running at the intermediate layers receive the data from the layer below, process it, and pass it to the layer above. The top-layer protocol receives the data from the layer below it and passes it to the appropriate application program.

Protocol Layering is discussed in detail in section 2.3.

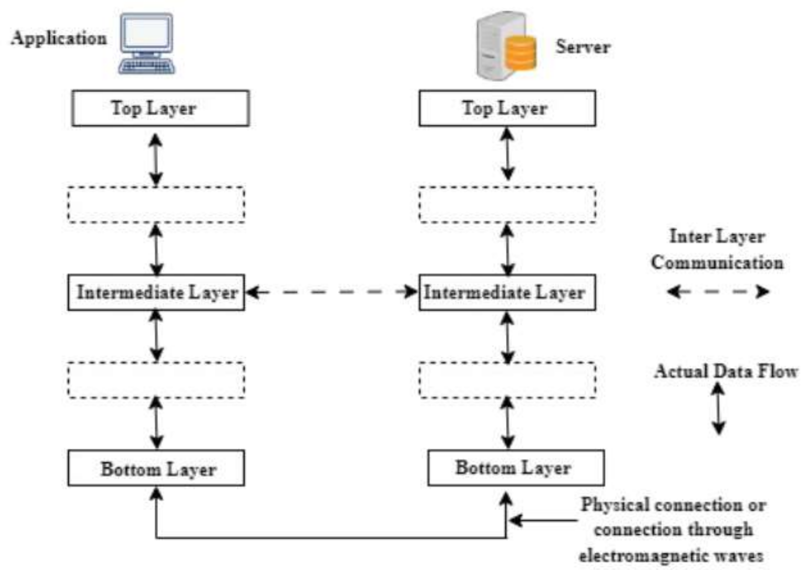

Fig. 2.1: Layered Model

In the following subsections, we study the following topics:

- Two Network models, the OSI reference model, and the TCP/IP Internet model
- Layered protocol architecture
- Different Internet Access networks
- Circuit and Packet Switching

2.2.1 Open Systems Interconnection (OSI) Reference Model

A reference model was given by the International Organization of Standardization (ISO) in the late 1970s covering all aspects of network communication. This model was given to allow communication between different systems without changing the logic underlying the hardware and software and thereby called Open Systems Interconnection.

OSI is a standard specifying the functionality of the layers and the interfaces between them. It is a layered framework consisting of 7 separate but related layers. The framework helps in the design of network systems that allow communication between all types of computer systems. It also helps us understand and design a network architecture that is flexible, robust, and interoperable.

Each layer defines only a part of the whole process of moving information across a network.

Fig. 2.2 shows the OSI Reference Model.

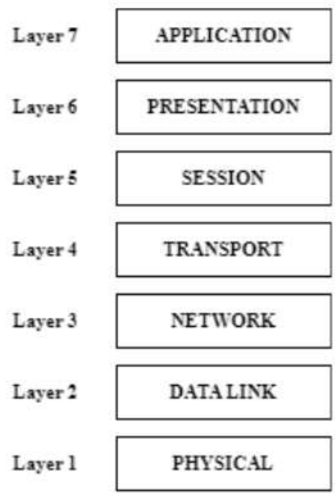

Fig. 2.2 : OSI Model

The important function of each layer is given below:

1. Physical Layer: This layer is responsible for the physical characteristics of the interface and medium, representation of bits (smallest unit of data in the digital format), data rate, synchronization of bits, line configuration, physical topology, and transmission mode.
2. Data Link Layer: This layer is responsible for moving the datagrams across a link, creating a frame (data unit at layer 2), physical addressing, error control, and access control.
3. Network Layer: This layer creates a connection between the source and destination computer and is responsible for the delivery of packets (data unit at layer 3) across the same or different networks, providing internetworking, and combining heterogeneous networks logically together to look like a single network to upper layers.
4. Transport Layer: This layer is responsible for an end-to-end logical connection and provides services such as end-to-end delivery, addressing, reliable delivery, flow control, and multiplexing.
5. Session layer: This layer is responsible for opening, managing, and closing sessions between the computers.
6. Presentation Layer: This layer is responsible for translating the message between the layers and maintaining its integrity.
7. Application Layer: This layer provides various functions for applications to access services provided by the network. The data input and output take place at this layer.

2.2.2 The TCP/IP Internet Model

This model uses the Transmission Control Protocol / Internet Protocol (TCP/IP) Protocol Suite also known as TCP/IP protocol stack. The TCP/IP is a protocol suite that is a set of protocols or rules organized in different layers used on the Internet today. It is derived from the OSI model where the presentation and session layer functionality is implemented in the transport and application layers. The data link and physical layer functionality are merged into one layer called as link layer or network interface layer and here the network layer is called as Internet layer.

There are two versions of the TCP/IP model. RFC 1122, defines a model with four layers. However, a second version with 5 layers was preferred by the users for a better understanding in comparison with the OSI model. Here the Internet Layer was referred to as the Network Layer and the Link layer was divided into Data link and Physical layers. The two versions of the models are shown in Fig. 2.3

It is also a hierarchical model like the OSI model and is made up of interactive modules. Each module provides a specific functionality, and each upper-layer protocol is supported by the services provided by the lower-layer protocol.

Layer 4	APPLICATION	APPLICATION	Layer 5
Layer 3	TRANSPORT	TRANSPORT	Layer 4
Layer 2	INTERNET	NETWORK	Layer 3
Layer 1	LINK	DATA LINK	Layer 2
		PHYSICAL	Layer 1

Fig. 2.3: 4 Layered and 5 Layered TCP/IP Models

The functionality of TCP/IP or Internet Protocol Stack and the protocols used are summarized below:

1. Application Layer - Supports application processes that generate messages. This layer uses protocols based on the application being used.
   Eg: An E-mail application uses SMTP along with IMAP or POP3, A Web browser uses HTTP, a file transfer application uses FTP, and so on.
2. Transport Layer - Supervises process-to-process communication (multiplexing/demultiplexing messages).
   Eg: TCP is a reliable transfer protocol, and UDP is an unreliable transfer protocol.
3. Network/Internet Layer - Enable end-to-end routing of messages (from source to destination)
   Eg: IP protocol uses IP addresses to route IP packets from source to destination.
4. Link Layer – Combines the functionality of the Data Link layer and Physical layer of the OSI model. Enable hop-to-hop transfer of messages (between neighbors) and Enable bit transmission on medium(air/wire)
   Eg: Ethernet is a link layer used to transfer packets in a Local Area Network on a shared medium, and 802.11 is a wireless transfer protocol.
   Eg: 10-base-T is an Ethernet standard for transmission on twisted pair cables.

2.3 Protocol Layering

For effective communication, the sender, the receiver, and all intermediate devices need to follow a set of rules, called Protocols. A Protocol defines the format and the order of the messages exchanged between the two end devices. Also, they specify what action is to be taken during the transmission and reception of a message.

Simple communication between only a few devices may need only one simple protocol, but when communication is complex, we need protocol layering or a protocol at each layer.

To understand protocol layering, let us consider two simple scenarios, one with simple communication between two persons X and Y happening at one layer and the second happening at three layers.

Scenario 1:

Assume X and Y are neighbors and communicate face-to-face in the same language.

Rules and steps followed for effective communications are as below:

1. Greetings should be exchanged before a conversation.
2. Conversation happens through a dialog, but when one is talking the other should listen.
3. Greetings should be exchanged after the conversation before they part.

Fig. 2.4 demonstrates Scenario 1, where the communication process happens in a single layer.

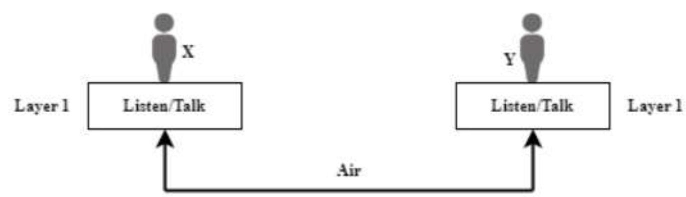

Fig. 2.4: A Single layer protocol

Scenario 2:

Assume X and Y have moved to different locations but are connected through a common business. X wants to communicate with Y for sharing business information through the postal service. For maintaining the security of the information, both X and Y agree upon a common encryption (encoding) and decryption (decoding) technique. There are three entities at each end to perform the tasks at each layer.

Rules and steps followed for communications are as below:

1. X creates the mail content in plain text for encryption or next-level processing.
2. The encrypter or encryption process generates the cipher text for mailing.
3. The encrypted mail is posted.
4. The postman delivers the mail.
5. Decrypter or a decrypting process decodes the contents.
6. Y receives the information.

Fig. 2.5 demonstrates Scenario 2, where the communication process happens in three layers.

In this scenario, the first three tasks are performed at the sending end and the next three at the receiving end. The steps are reversed when Y wants to send information to X.

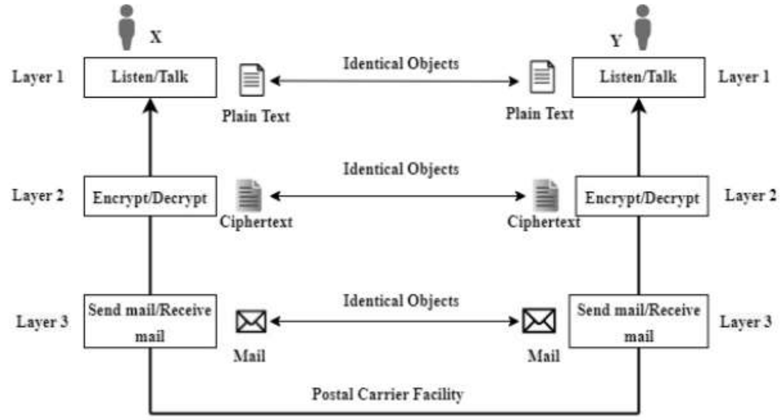

Fig. 2.5: A Three-layer protocol

The protocol layering benefits communication in the following ways:

- It is seen that the complex task at each end is divided into three smaller and simpler ones, performed at different layers.
- Every layer receives services from lower layers and provides services to the above layers. These services are provided by following different protocols at every layer.
- Services are separated from their implementation, i.e how the inputs are changed to outputs is not considered but what inputs and outputs are needed are of concern.
- The design of intermediate devices between the sender and receiver becomes simpler and less expensive.

2.3.1 Principles of Protocol Layering

Protocol layering is based on two principles:

1 In the case of bidirectional communication, each layer should be built to perform two opposite tasks, one in each direction.

2 Two objects under each layer at both sites (source and destination) should be identical.

The data units or objects at the different layers in the TCP/IP model are shown in Fig. 2.6. Here the 5-layer model is used for better understanding. The data units at the different layers from the top are messages, segments or user datagrams, datagrams, frames, and bits.

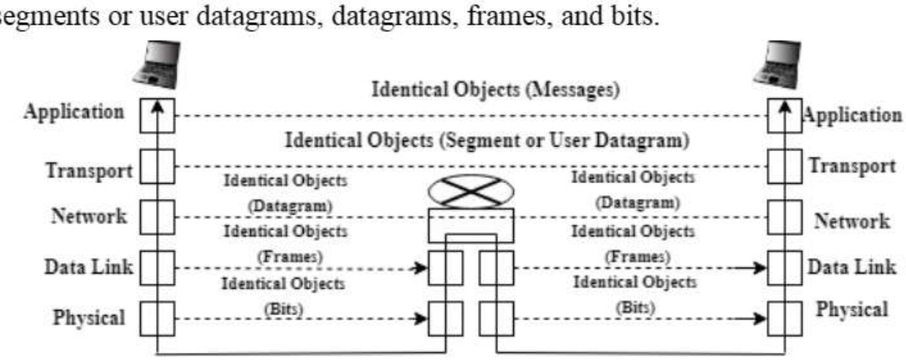

Fig. 2.6: Identical Objects in the TCP/IP Protocol Suite

2.3.2 Logical Connection between Peer Layers

There should be a logical connection between each layer to have layer-to-layer communication. In Scenario 2 discussed in the previous section, X and Y can assume that there is a logical connection at each layer through which they can send the object created from that layer.

In Fig. 2.5, there is a logical connection between Layer 1 at X and Layer 1 of Y, between Layer 2 at X and Layer 2 of Y, and so on.

Fig. 2.7 demonstrates the logical connections between layers in the TCP/IP

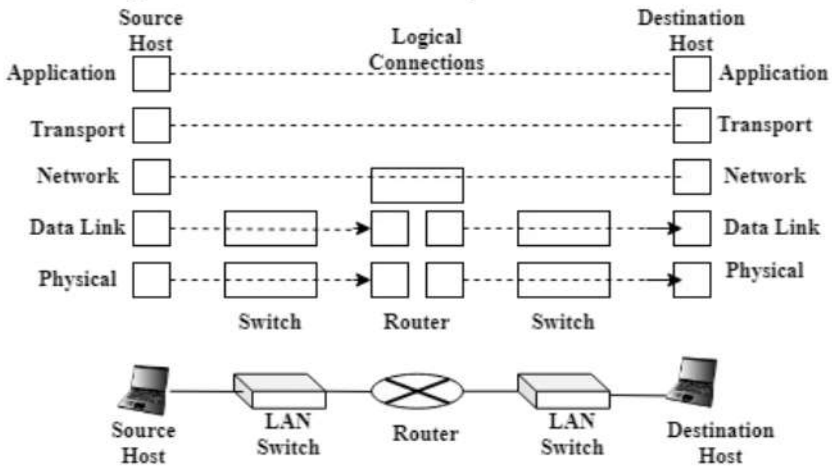

Fig. 2.7: Logical Connections between layers in TCP/IP

2.3.3 Encapsulation and Decapsulation

As discussed in Sections 2.2.1 and 2.2.2, network communication has been standardized by the use of network models like the OSI Reference Model and the TCP/IP Model. The TCP/IP is the new Internet model and has replaced the OSI.

When data is sent from one location to another on the Internet, it passes through different networks and networking devices. At each device, the data passes through different layers as shown in Fig.2.6. Each layer plays a specific role and provides services.

The application layer provides all the services such as messaging, web browsing, etc. to the end user. The transport layer provides host-to-host logical communications. Each device on the network has an IP address for identification. The network layer provides source and destination IP addresses to identify the location of the data unit on the network. The data link layer also known as the network access layer assigns source and destination MAC (media access control) addresses to the packets from the Network layer (also called a datagram) and sends them out on the network as frames. The MAC address is the physical address of the devices.

Encapsulation:

This process is performed at the Source Host. During transmission, data moves from the upper layer to the lower layer. At each layer, useful information called a header is added to the data or the payload. The header and the data now become the data to the next lower layer, which then adds its own header to it. The header is always added at the beginning of the data when it is transmitted. At the receiver, the header helps in extracting the data. This packaging of data at each layer is called encapsulation. Fig. 2.8 demonstrates the encapsulation process.

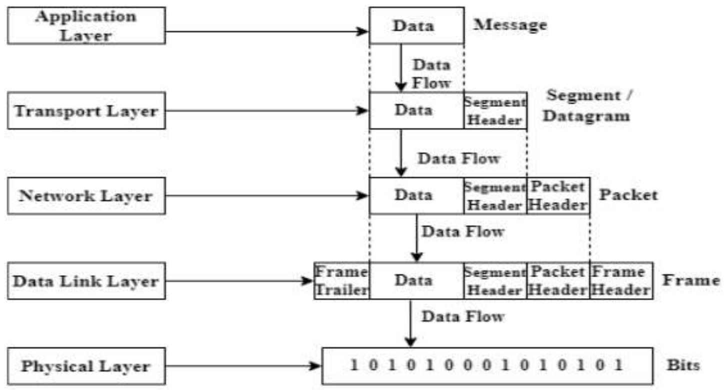

Fig. 2.8: The Encapsulation Process

Decapsulation:

This process is performed at the Destination Host. It is a reverse process of encapsulation. During the reception, data moves from the lower layer to the upper layer. Each layer unpacks the information from the corresponding header and uses it to deliver the data to the appropriate network application waiting for it. This unpackaging of data at each layer is called decapsulation. Fig. 2.9 demonstrates the decapsulation process.

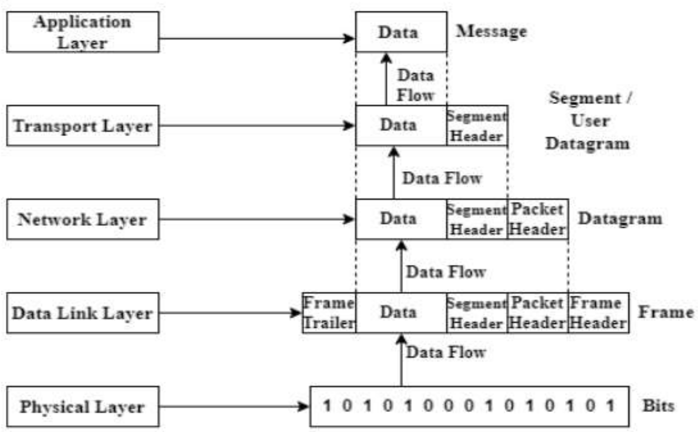

Fig. 2.9: The Decapsulation Process

2.3.4 Addressing

Addressing is needed whenever there is communication between two entities. As discussed in section 2.3.2, logical communication happens between the peer layers in the TCP/IP protocol stack. This is possible when they can address each other. Though there are five layers in the TCP/IP protocol stack, there are only four pairs of addresses (source and destination addresses at each layer).

- Physical layer data units are bits, so they **do not need** addressing.
- Application layer addresses are in the form of **names**, of the sites that provide the services.
  Eg: abc.org, xyz@mail.com
- Transport layer addresses are called **port numbers**. These are local addresses used to differentiate between programs running at the application layers at both the source and destination.

- Network layer addresses are logical addresses and are called **IP addresses**. These are global in nature and define the connection of a device to the Internet.
- Data Link layer addresses are called **MAC addresses** which define the physical address of the devices (Host or Router in a LAN or a WAN) connected to the Internet. So they are locally defined addresses.

2.3.5 Multiplexing and Demultiplexing in Protocol Layering

At every layer in the TCP/IP protocol stack, there is more than one protocol used to provide different services needed by the applications. So when the data unit passes from one layer to another, the protocol used at that layer has to be indicated by some means. This information is added in the header in the form of bit combinations in the fields.

Multiplexing is a process by which a protocol at a layer encapsulates a data unit from various higher-layer protocols at the source host. Demultiplexing is a process in which a protocol at a layer decapsulates a data unit and delivers it to the various higher-layer protocols at the destination host. To implement these processes, a field in the header is needed to identify the protocol to which the data unit belongs.

Fig. 2.10 shows multiplexing and demultiplexing at Transport and Network Layers.

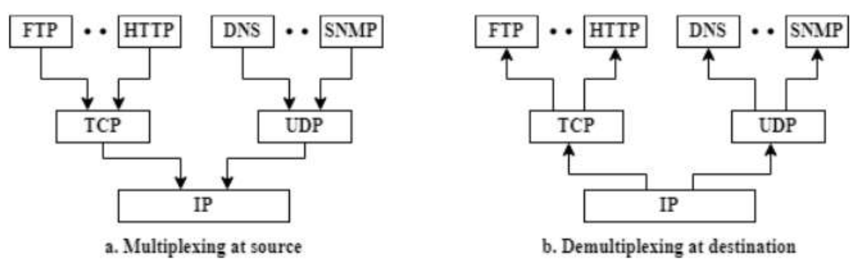

Fig. 2.10: Mutliplexing and Demultiplexing

At the transport layer, there are two protocols: Transmission Control (TCP) and User Datagram (UDP) protocols. At this layer, TCP or UDP can accept a message from many application layer protocols.

At the network layer, IP (Internet Protocol) can accept a segment from TCP or datagram from UDP and from other protocols such as ICMP, IGMP etc.

At the data link layer, the protocols can accept a packet from IP or other protocols like ARP (not shown in the figure).

2.4 The Network Edge

The end systems on the Internet also known as hosts can be either a client or a server. The client devices are phones, desktops, and other devices listed in the last section. The server systems are powerful machines that store and distribute Web pages, relay e-mail, stream video, etc.

Google hosts thousands of servers globally for storing Web pages, videos, etc.

In this section the following topics are discussed:

- Different ways of connecting to Internet Services or Access Technologies.
- The different physical mediums used for transmission.

2.4.1 Accessing the Internet

The Internet today allows any user to become part of it, but the needs to be physically connected to an ISP. This physical connection can be done through a point-to-point WAN in four different ways as given below:

1. Using Telephone Networks
2. Using Cable Networks
3. Using Wireless Networks
4. Using a Direct Connection

2.4.1.1 Using Telephone Network

Today all residences and small businesses use Telephone services, so they are connected to a Telephone Network. Telephone Networks are already connected to the Internet, So residents, individuals, and small businesses can connect to the Internet by changing the voice line between the residence and telephone center to a point-to-point WAN.

There are three ways to connect to the Internet through a telephone network:

1. Dial-up service
2. DSL service
3. Fiber to Home

1. **The Dial-up service** is obsolete as it is very slow and cannot be used for a telephone connection.

2. **Digital Subscriber Line (DSL) service**: Telephone companies have upgraded their telephone lines to provide high-speed Internet services to residences and small businesses. This is a broadband service that allows the line to be used simultaneously for voice and data communication. Fig. 2.11 shows a DSL Internet Service.

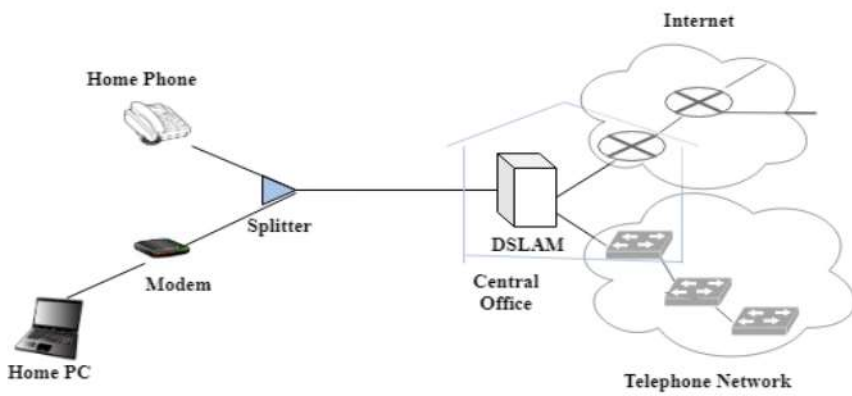

Fig. 2.11: DSL Internet Access

In this service, the customer's telephone service company is its Internet Service Provider (ISP).

DSL operation during upstream traffic from the customer to the Internet or the telephone network is as follows:

The DSL modem converts digital data to high-frequency analog signals to be transmitted to the central office on the telephone line. The splitter on the customer's side separates these signals from the

telephone signals. At the CO, the Digital Subscriber Line Access Multiplexer (DSLAM) converts these analog signals to digital format and sends the data to the Internet.
Telephone signals are separated from the signals from the PC again by the splitter and transmitted as low-frequency audio signals on the telephone lines to the Central Office (CO). At the CO, the DSLAM separates these signals from the data and sends them to the telephone network.

DSL operation during downstream traffic from the Internet or the telephone network to the customer, t is as follows:
Data from the Internet is converted to high-frequency analog signals by the DSLAM and forwarded to the customer's splitter through the telephone lines. The splitter separates these signals from the telephone signals and passes them to the modem, which converts them to digital data for the PC.
Telephone signals from the telephone network are separated from the data by the DSLAM and sent on the telephone lines. The splitter separates them from the data and forwards them to the customer's phone.

The upstream data, downstream data, and telephone signals are transmitted simultaneously on the telephone lines by encoding them at different frequencies.

3. Fiber to Home (FTTH)

This is an optical distribution network, where one fiber cable leaves the Central Office (CO) for every home. Fig. 2.12 shows FTTH Internet access.

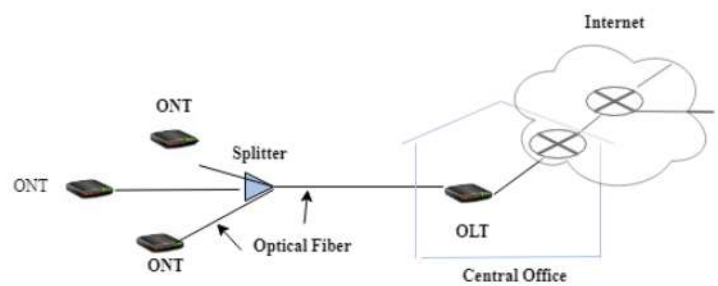

Fig. 2.12: FTTH Internet Access

Each home uses an optical network terminator (ONT) connected to a shared neighborhood splitter. The splitter combines individual fibers from the homes and connects them to an optical line terminator in the telephone company's CO. The Optical Line Terminator (OLT) converts optical signals to electrical and vice-versa and connects to the Internet through a telephone company router. In the homes, users connect a wireless home router to the ONT and access the internet through this router. The access rates are very high in Gigabits per second. The FTTH ISPs offer different data rates. Higher data rates cost more for the customers.

2.4.1.2 Cable Networks

The cable television network services have upgraded their cable network to provide Internet connection to residences and small businesses. In this service, the customer's cable television company is its

Internet Service Provider (ISP). Fig. 2.13 shows a Hybrid Fiber-Coaxial Cable Network.

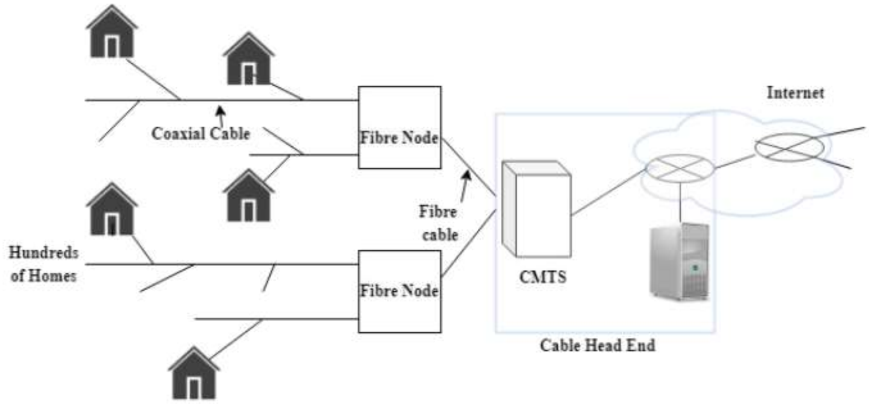

Fig. 2.13: A Hybrid Fiber-Coaxial Access Network

It uses both fiber and coaxial cable are used as transmission mediums, so is called a Hybrid Fiber-Cable Network (HFC). The optical fiber cable connects the Cable Head End to the neighborhood-level junction. From there the coaxial cable is used to reach the individual homes. The cable head end houses the cable modem termination system (CMTS) which works like a DSLAM as in the DSL Network discussed in the previous section. The modem used on the customer's side is a special cable modem, which divides the HFC network into two channels, a downstream and an upstream channel.

This cable Internet access is a shared broadcast medium, where every packet sent by a home travels on the upstream channel to the head end. Similarly, every packet sent by the head end travels downstream on every link to every home.

2.4.1.3 Wireless networks

This is the most popular way of accessing the Internet today. Here the customers can connect to the Internet using a combination of wired and wireless connections.

Eg: In a wireless LAN, the users transmit or receive a packet from an access point that will be connected to the enterprise network, which in turn will be connected to the wired Internet as shown in Fig. 2.2. Wireless LAN access uses IEEE 802.11 technology also known as Wi-fi. Here the user must remain within a few tens of meters of the access point.

Today Wi-fi is used in Universities, Cafeterias, Business Offices, Airports, Aircraft, and homes.

2.4.1.4 Direct Connection to the Internet

Large organizations or corporations can get connected to the Internet by leasing high-speed WAN from a carrier provider and connecting themselves to a regional ISP. This connection can act as a local internet service provider for all the people in the organization or the corporations.

Eg: A large University with several campuses can create internetwork and connect their internetwork to the INTERNET.

2.5 The Network Core

The core of a data communication network is the central part of a network also called a backbone network. It is designed to transfer network traffic at high speeds, with optimum performance, and reliability over long distances, and to upscale when needed. They connect WANs and LANs together. They provide a lot of services but their key function is to switch/route the data packets across different sub-networks in their path from source to destination.

In this section the following topics are discussed:

- The two switching technologies.

2.5.1 Switched Communication Networks

These are networks where the data to be transferred from a source node to a destination node is routed between various intermediate nodes. When there are multiple paths between a source to a destination, switching helps in finding the best route for data transmission.

Eg: Telephone Networks, and the Internet

The **two** most common types of switched networks are :

1. Circuit-switched network (CSN)
2. Packet-switched network (PSN)

2.5.1.1 Circuit-switched network

In this network, a dedicated connection called a circuit will be available between any two end systems. All the resources such as the link, transmission rate, and buffers will be reserved for the whole duration of the communication between the end systems. A switch can only make the connection active or inactive. The earlier telephone network was completely circuit-switched, but today part of it is packet-switched.

Fig. 2.14 shows a circuit-switched telephone network with two telephones at each end.

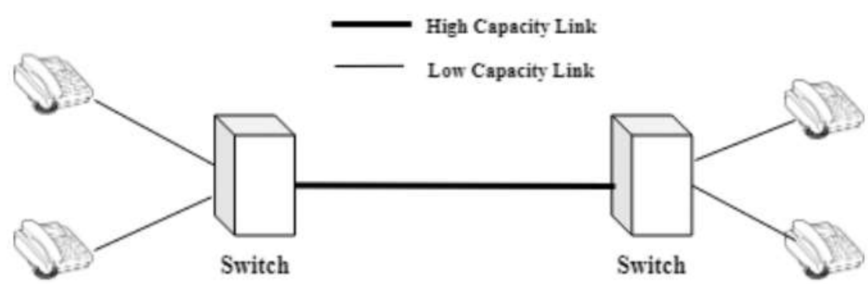

Fig. 2.14: Circuit Switched Telephone Network

Here the capacity of the link between the switches is shared between pairs of telephones and is two voice calls at the same time. The link is efficiently utilized when all the telephone sets are busy. When only one pair of phones are communicating, only half the link capacity is used. So this type of network is inefficient.

Multiplexing in Circuit-Switched Networks

Today, communication networks connect millions of nodes for exchanging information. In order to reduce the hardware cost and complexity, the transmission links are shared between multiple nodes as shown in Fig. 2.14.

The technique used by the transmitter to combine and transmit multiple data streams over a single medium is called Multiplexing, and the hardware used is called Multiplexer.

A multiplexer divides the high-capacity physical medium into a low-capacity logical medium which will be shared by different data streams. Multiplexing is used for all mediums of transmission. As multiplexing is used at the sending end, a reverse process or technique called demultiplexing is used on the receiving end to separate the data and distribute it to the appropriate receivers.

There are **two multiplexing techniques** used in Circuit Switching Links: Frequency-division and Time-division

Frequency-division multiplexing (FDM): This is an analog technology and the carrier is frequency. Here the frequency spectrum or carrier bandwidth of a link is divided among the connections established across the link. Every user pair is allocated one logical channel or a frequency band.

Consider a telephone network supporting four circuits where the frequency domain is divided into four bands, each with a bandwidth of 4kHz. Fig. 2.15 illustrates FDM for this network.

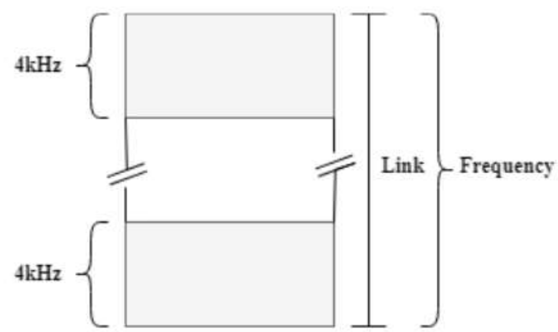

Fig. 2.15: Frequency Division Multiplexing

Time-division multiplexing (TDM): It is primarily a digital technology but can be used for analog transmission also. Here time is divided into fixed time duration frames and each frame is divided into a fixed number of time slots. When a connection is established across a link, one, time slot in every frame is dedicated to one user-pair.

The transmission rate of a circuit = Frame rate x Number of bits in a slot.

Consider a TDM network, where the time domain is segmented into frames, with each frame divided into four, time slots. Fig. 2.16 illustrates TDM for this network.

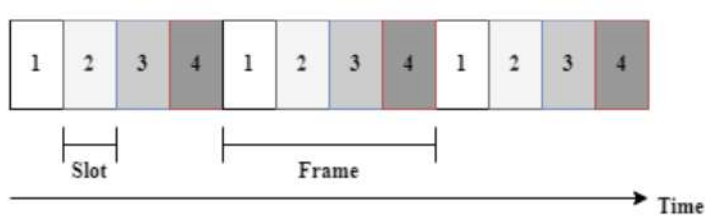

Fig. 2.16: Time Division Multiplexing

Here all the slots labeled 3 are dedicated to one connection or one user pair.

2.5.1.2 Packet-switched network (PSN)

In this type of network, the communication between the two end systems is done in blocks of data called Packets. Each packet is an independent entity and travels through communication links and packet switches. As each data packet can travel on a different route, there is no dedicated link required. So, this system is cost-effective and more efficient. There are two types of packet switches used today, Link layer switches and Routers. The switches in PSN are used for both storing and forwarding the packets. The packets are transmitted at a full transmission rate of the link. Fig. 2.17 shows a packet-switched network.

Store-and-Forward Transmission:

This is the technique used by the packet switches at the inputs of the links. Here the switches save all the bits in a data packet and then forward the packet toward its destination.

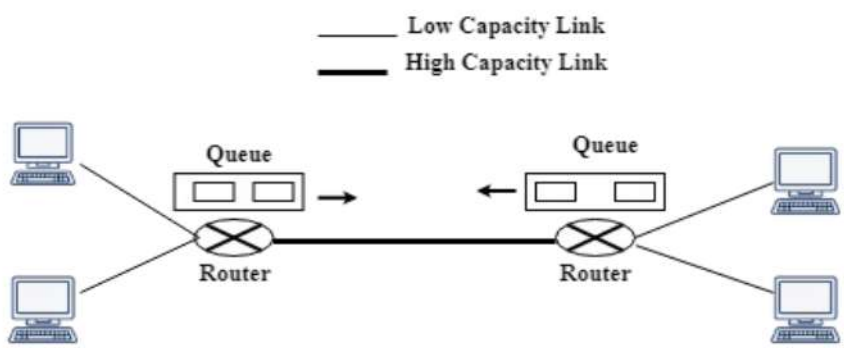

Fig. 2.17: Packet Switched Network

Consider a packet of L bits being transmitted over an R bits/sec transmission link.

The time needed to receive the packet at the second router on the right after being transmitted from a node on the left = time needed to receive all the bits by the first router on the left + time needed to transmit all the bits by it.

$= (L/R + L/R)$

$= 2L/R$ secs.

If there are N links in the path from source to destination, each with a transmission rate of R bits/sec, then the end-to-end delay for transmitting a packet of L bits = NL/R secs.

Queuing Delays and Packet Loss:

These are some of the problems faced during the transmission of packets through multiple devices and links in their path from source to destination. Every packet switch has multiple input and output ports for connecting multiple input links to output links. The packets received on an input link are stored and then forwarded. To facilitate this, storage spaces called buffers are used to store the packets. Also, every outgoing link has an output buffer to store the packets until they are ready to transmit. They remain stored here until the forward transmission link has finished the transmission of another packet and the link is free. This results in a delay and is called a **queuing delay.** The amount of queuing delay varies with the congestion level in the network.

The buffer space is limited and if the incoming packets arrive when the buffer is full, some of these packets or queued packets will be dropped and this results in **packet loss.** The number of packets dropped depends on the congestion in the network and the size of the buffer.

Forwarding Tables and Routing Protocols:

These are features added in the switch or router, in order to choose the best forward link for the packets to reach their destination.

Every end system on the Internet is identified by an IP address (Internet Protocol address). In order to reach the correct destination, every packet has the destination address in the packet header. Every router has a forwarding table consisting of a mapping of a destination address to that router's output link. So when a packet arrives at a router, it examines the destination address part, searches its forwarding table, and forwards the packet to the most appropriate output link.

The entries in the forwarding tables are set and updated regularly by the routing protocols. These protocols determine the shortest path from that router to each destination and accordingly configure the forwarding tables. These protocols are implemented at the network layer.

2.6 Performance of a Network

The performance of a network is measured in terms of the quality of service user experiences while using a network. This can be measured in many ways depending on the nature of the network and its design. The important characteristics that measure the performance of a network are Bandwidth, Throughput, Latency, Bandwidth-delay product, and Jitter.

2.6.1 Bandwidth

Based on the context used, bandwidth can be measured in Hertz or bits per second (bps). It measures how much data can be transferred over a network at a particular time.

Bandwidth in Hertz (Hz): It is the range of frequencies present in a continuous band of frequencies. And it is calculated as the difference between the highest and the lowest frequency in that band.

Bandwidth in Bits per second: It gives the number of bits that a channel, a link, or a network can transmit. So it is the maximum data rate. The data rate depends on the type of modulation used and other factors.

2.6.2 Throughput

It is a measure of the actual speed with which the data can be sent through a network in bps. Though Bandwidth and throughput measure bps, they are different measures and can be seen in the example below.

Eg: A link may have a bandwidth of B bps, but if the throughput is T bps, then we can only send T bps through this link where T is always less than B.

Example 6: Calculate the throughput of a network that can pass an average of 12,000 frames per minute, where each frame carries an average of 10,000 bits and the bandwidth of the channel is 10 Mbps.

Solution: Throughput T = (No. of Frames/second × Bits/frame) bps

$= 12000/60 \times 10000$
$= 200 \times 10000$
$= 2 \times 10^6$ bps
$= 2$ Mbps

2.6.3 Latency (Delay)

This term defines the time taken for a complete message to arrive at the destination from the first bit sent out from the source.

Latency consists of **four** components: Propagation delay, Transmission delay, Queuing delay, and Processing delay

Propagation delay: It is the time required for a single bit to travel from the source to the destination. It is expressed as,

The propagation delay = Distance/Propagation Speed, where the propagation speed of electromagnetic signals depends on their frequency and the medium in which they travel.

Transmission delay: It is the time required for transmitting the complete message. This time depends on the size of the message and the bandwidth of the channel and is expressed as,

Transmission delay = Message size/Bandwidth

Queuing delay: It is the time taken by the intermediate or end devices to store the message before it is processed. It is not a fixed factor but depends on the load imposed on the network. So when the traffic is heavy, the intermediate devices need to queue the messages and process them one by one.

Processing Delay: It is the time that the router takes to find out where to send the packet and queue the packets on that link for transmission. This delay depends on the complexity of the routing protocols.

The latency = Propagation delay + Transmission delay + Queuing delay + Processing Delay

2.6.4 Bandwidth – Delay Product:

This term is very significant in data communication.

Let us understand this term with an analogy as shown as given below:

The link between two points on a network can be thought of as a pipe. If the cross-section of the pipe represents the bandwidth, and the length of the pipe represents the delay, then the volume of the pipe defines the bandwidth-delay product, as shown in Fig. 2.18

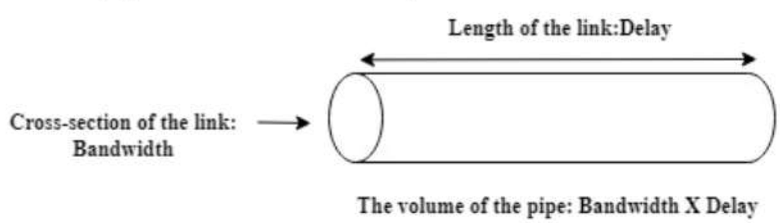

Fig.2.18: Concept of Bandwidth-Delay product

Let us apply the above concept to a communication link by considering two cases of transmission for filling the link with bits.

Case 1: Let the link bandwidth be 1bps and a delay of 5secs. Bandwidth-delay product will be 5 bits .Fig. 2.19 shows how 5 bits are filled on the link in 5 seconds time.

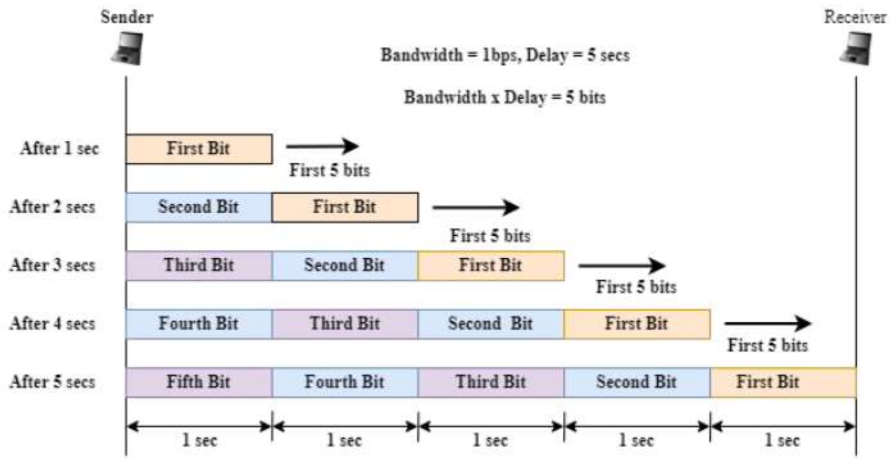

Fig. 2.19: Filling of the link by bits in Case 1

Case 2: Let the link bandwidth be 5 bps and a delay of 5 secs. Fig. 2.20 shows how 25 bits are filled on the link in 5 seconds time.

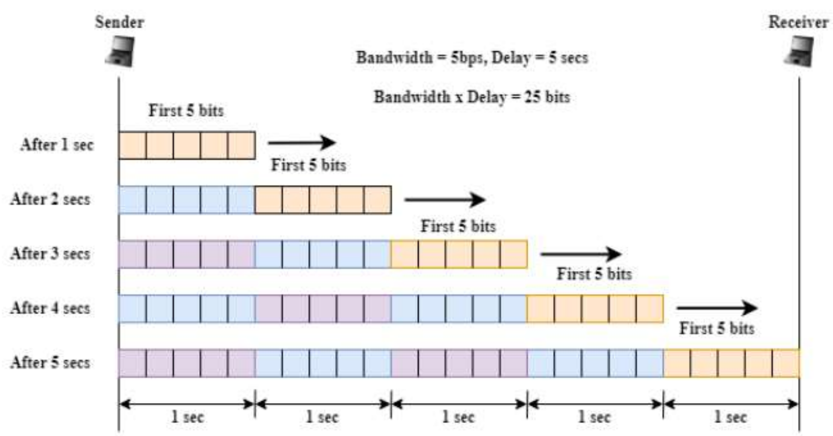

Fig. 2.20: Filling of link by bits in Case 2

2.6.5 Jitter

Different packets of a message may take different paths from their source to their destination. This causes different delays when they arrive at their destination. In a time-sensitive application like that of video or audio data, this is experienced as Jitter.

So, jitter is the variation in the time of arrival of the different packets at the receiver due to network congestion or paths taken by them.

We can say that a real-time application using the packets experiences a jitter, if the delays for the three packets first, second, and third are 20 ms, 45 ms, and 40 ms respectively.

2.7 Self-Assessment Questions

Q1. What is the Internet? List some of its services. (5 marks, L2)

Q2. Define a Protocol. How does protocol layering benefit communication? (6 marks, L4)

Q3. What are the two principles on which protocol layering is based? (2 marks, L2)

Q4. List the different access technologies. Describe them briefly. (10 marks, L3)

Q5. What are the different media used for data transmission? Describe them (8 marks, L3)

Q6. How is switching useful? What are the two types of switched networks? Explain. (7 marks, L3)

Q7. Discuss the important performance parameters. (10 marks. L3)

2.8 Self-Assessment Activities

A1. Why are open standards important for protocols? Discuss

A2. Explain why has packet switching been taken over by circuit switching.

A3. Find out the different residential internet access technologies in your city along with the advertised data rate and monthly price.

A4. Build a network topology of 5 devices and simulate data transmission between them using any software tool (Hint: Wireshark or NS3)

A5. Interact with a network administrator and find out the topology and protocols used for communication in the LAN where you are connected in your workplace.

2.9 Multiple-Choice Questions

Q1. Application layer data unit is ______ [1 mark, L1]

A. Bits
B. Frames
C. Datagram
D. Message

Q2. Transport Layer data unit is _________[1 mark, L1]

A. Bits
B. Frames
C. Segment / User Datagram
D. Messages

Q3. Physical Layer data unit is _______[1 mark, L1]

A. Bits
B. Frames
C. Datagram
D. Messages

Q4. Find out the OSI layer, which performs error control. [1 mark, L1]

A. Data Link Layer
B. Transport Layer
C. Session Layer
D. Presentation Layer

Q5. Which of the layers in the TCP/IP protocol stack is used for transferring files from one machine to another? [1 mark, L1]

A. Application Layer
B. Transport Layer
C. Network Layer
D. Physical Layer

Q6. The Data Link layer is ——— and the session layer is ———. [1 mark, L1]

A. End-to-End, Hop-to-Hop
B. Hop-to-Hop, End-to-End
C. Hop-to-Hop, Hop-to-Hop
D. End-to-End, End-to-End

Q7. The switching method used in the traditional telephone network is,

A. Circuit Switching
B. Packet Switching
C. Both A and B
D. None of the above

Q8. A dedicated path is established between the transmitter and the receiver after the connection establishment in,

A. Circuit Switching
B. Packet Switching
C. Both A and B
D. None of the above

Q9. The switching method which is more efficient in using network resources is,

A. Circuit Switching
B. Packet Switching
C. Both A and B
D. Depends on the size of the network

Q10. _______ defines the time taken for a complete message to arrive at the destination from the first bit sent out from the source

A. Latency
B. Propagation delay
C. Queuing delay
D. None of the above

Q11. The delay which depends on the size of the message and the bandwidth of the link is _______

A. Transmission Delay
B. Propagation delay
C. Queuing delay
D. None of the above

Q12. The delay which depends on the load on the network is _______

A. Transmission Delay
B. Propagation delay
C. Queuing delay
D. None of the above

2.10 Keys to Multiple-Choice Questions

Q1. Messages (D).
Q2. Segment or User Datagram (C).
Q3. Bits (A)
Q4. Data Link Layer (A)
Q5. Application Layer (A)
Q6. Hop-to-hop, End-to-end (B)
Q7. Circuit Switching (A)
Q8. Circuit Switching (A)
Q9. Packet Switching (B)
Q10. Latency (A)
Q11. Transmission Delay (A)
Q12. Queuing Delay (C)

2.11 Summary of The Unit

This unit covers three important topics: Reference models for building a data communication network, its edge, and its core or the internal part of it. An introduction to the Internet and the reference models on which the different functional components of the Internet are designed are discussed in the first unit. In order to provide services, all the components in the network need to communicate and follow certain rules to avoid data loss. These rules are implemented in the form of protocols at every layer in the TCP/IP protocol suite. In the second section, the layered architecture and protocol layering is discussed. In the third section, the different technologies used to provide access to the Internet are covered. Communication network technology has grown enormously and entered all facets of our daily lives and this has been possible because of switching technology, which has been discussed in the next section. The performance of a network can be measured by various characteristics such as Bandwidth, Throughput, Latency, Bandwidth-delay product, and Jitter.

2.12 Glossary

1. Request for Comments (RFC) – It is a publication from the Internet Engineering Task Force (IETF), a standard-setting body of the Internet. An RFC is authored by computer scientists, individuals, or groups of engineers in the form of a memorandum that describes research, behaviors, methods, or innovations related to the working of the Internet and Internet-connected systems.
2. IEEE 802.11 – These are a set of communication standards for wireless LANs given by the Institute of Electrical and Electronic Engineers (IEEE)

2.13 Recommended Learning Resources

[1] James F Kurose and Keith W Ross, Computer Networking, A Top-Down Approach, Sixth Edition, Pearson,2017.

[2] Behrouz A Forouzan, Data and Communications and Networking, Fifth Edition, McGraw Hill, Indian Edition

2.14 References

[1] https://isaaccomputerscience.org/concepts/net_internet_tcp_ip_stack_g?examBoard=all&stage=all

[2] https://docs.oracle.com/cd/E19620-01/805-4041/intro-78284/index.html

[3] https://documentation.softwareag.com/adabas/wcp652mfr/wtc/wtc_prot.htm

[4] https://www.w3.org/People/Frystyk/thesis/TcpIp.html

[5] https://publicdomainvectors.org/en/free-clipart/Vector-illustration-of-simple-man-or-person-silhouette-icon/21350.html (Fig. 2.4 and 2.5)

[6] https://vectorportal.com/vector/mail-envelope-vector-symbol/9100 (Fig 2.5)

[7] https://creazilla.com/nodes/5627647-002-text-document-icon (Fig 2.5)

[8] https://openclipart.org/detail/97543/text-file-icon (Fig 2.5)

[9] https://www.propatel.com/encapsulation-and-decapsulation-difference

[10] https://www.omnisecu.com/tcpip/tcpip-encapsulation-decapsulation.php (Fig. 2.8 and Fig. 2.9)

[11] https://www.needpix.com/photo/download/854666/telephone-phone-communication-technology-connection-free-vector-graphics-free-pictures-free-photos-free-images (Fig. 2.11)

[12]https://creazilla.com/nodes/3432054-house-icon (Fig. 2.13)