

MODULE 1 SECTION

N: A, B, C, D

Topic	Portion
Introduction Chapter 1	Data Communications: Components, Representations, Data Flow, Networks: Physical Structures, Network Types: LAN, WAN, Switching, Internet
Network Models Chapter 2	Protocol Layering: Scenarios, Principles, Logical Connections.
	TCP/IP Protocol Suite: Layered Architecture, Layers in TCP/IP suite, Description of layers
	Encapsulation and Decapsulation, Addressing, Multiplexing and Demultiplexing, The OSI Model: OSI Versus TCP/IP
Data-Link Layer Chapter 11	Introduction: Nodes and Links, Services, Categories of link, Sublayers, Link Layer addressing: Types of addresses, ARP.
	Data Link Control (DLC) services: Framing, Flow and Error Control,
	Data Link Layer Protocols: Simple Protocol, Stop and Wait protocol, Piggybacking

TEXTBOOK: Data Communications and Networking, B Forouzan, 5th Ed, McGraw Hill Education, 2016, ISBN: 1-25-906475-3

MODULE 1

Data communications

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.

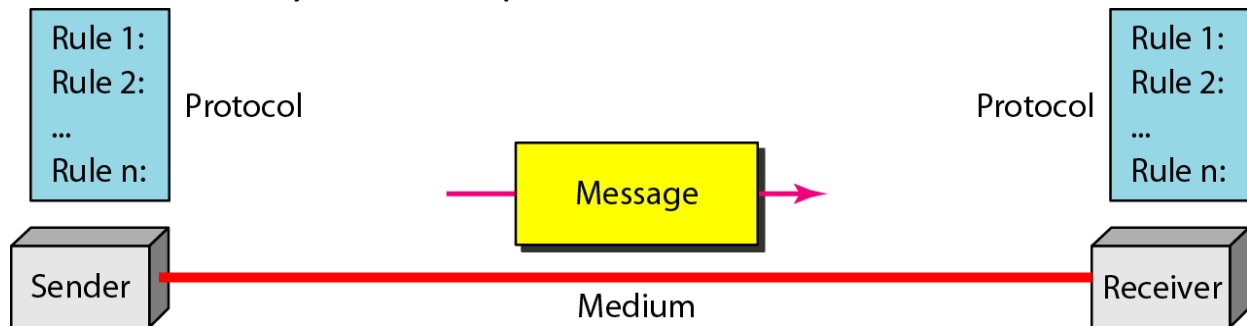
The term **telecommunication**, which includes telephony, telegraphy, and television, means communication at a distance (tele is Greek for “far”). The word data refers to information presented in whatever form is agreed upon by the parties creating and using the data.

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

1. **Delivery**- The system must deliver data to the correct destination. Data must be received by the intended device or user and only by that device or user.
2. **Accuracy**- The system must deliver the data accurately. Data that have been altered in transmission and left uncorrected are unusable.
3. **Timeliness**. The system must deliver data in a timely manner. Data delivered late are useless. In the case of video and audio, timely delivery means delivering data as they are produced, in the same order that they are produced, and without significant delay. This kind of delivery is called real-time transmission.
4. **Jitter**. Jitter refers to the variation in the packet arrival time. It is the uneven delay in the delivery of audio or video packets. For example, let us assume that video packets are sent every 30 ms. If some of the packets arrive with 30-ms delay and others with 40-ms delay, an uneven quality in the video is the result.

Components

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 soon.
3. **Receiver-** The receiver is the device that receives the message. It can be a computer, workstation, telephone handset, television, and soon.
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- 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-** 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, we can increase the size of the bit pattern to include gray scale. For example, to show four levels of gray scale, we 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.

Video- Video refers to the recording or broadcasting of a picture or movie. Video can either be reproduced 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.

Data Flow

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

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).

example- Keyboards and traditional monitors are examples of simplex devices. The keyboard can only introduce input; the monitor can only accept output.

simplex mode can use the entire capacity of the channel to send data in one direction

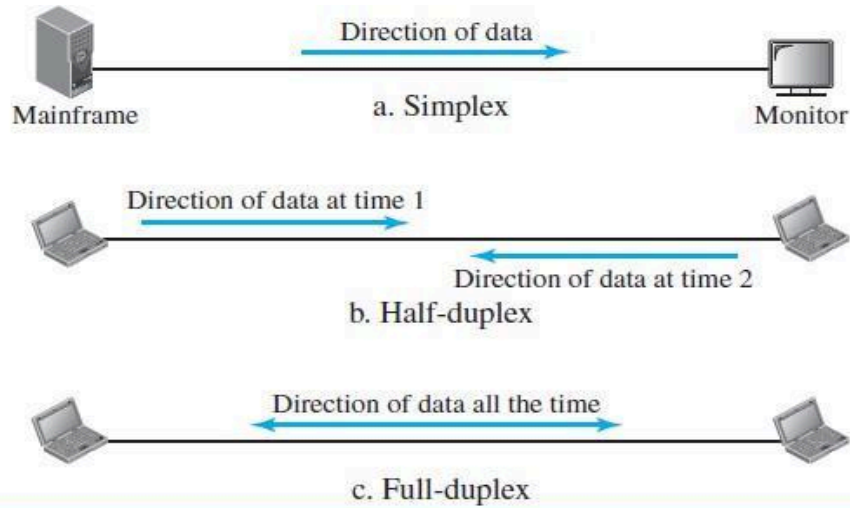


fig: dataflow

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 (see Figure b). 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.

example-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 mode (also called 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.

NETWORKS

A network is the interconnection of a set of devices capable of communication. A device can be a host (or an end system as it is sometimes called) such as a large computer, desktop, laptop, workstation, cellular phone, or security system.

A device can also be a connecting device such as a router, which connects the network to other networks, a switch, which connects devices together, a modem (modulator-demodulator), which changes the form of data, and so on.

These devices in a network are connected using wired or wireless transmission media such as cable or air. When we connect two computers at home using a plug-and-play router, we have created a network, although very small.

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.

Reliability- 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- 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.

Physical Structures

Network attributes- Type of Connection and physical topology

Type of Connection

A network is two or more devices connected through links. A link is a communications pathway that transfers data from one device to another.

There are two possible types of connections:

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 (see Figure a).

example-When we change television channels by infrared remote control, we 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 (see Figure b).

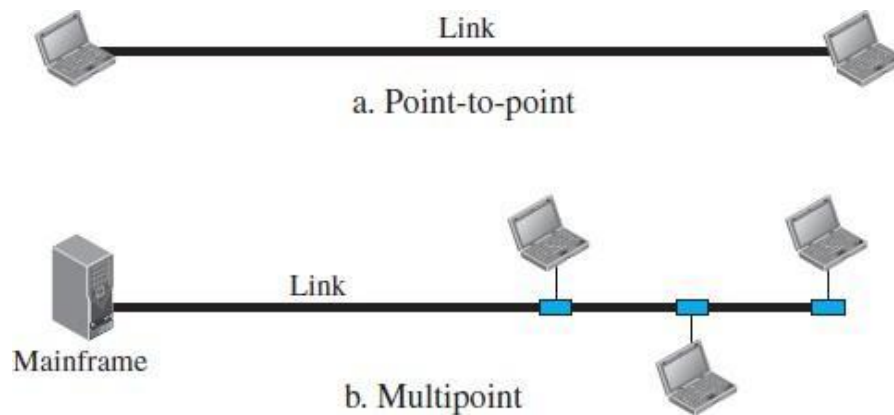


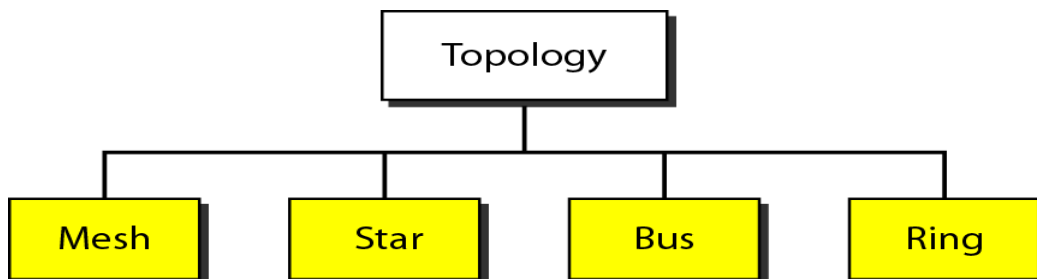
fig: Type of connection

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.

Physical Topology

The term physical topology refers to the way in which a network is laid out physically. Two 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.



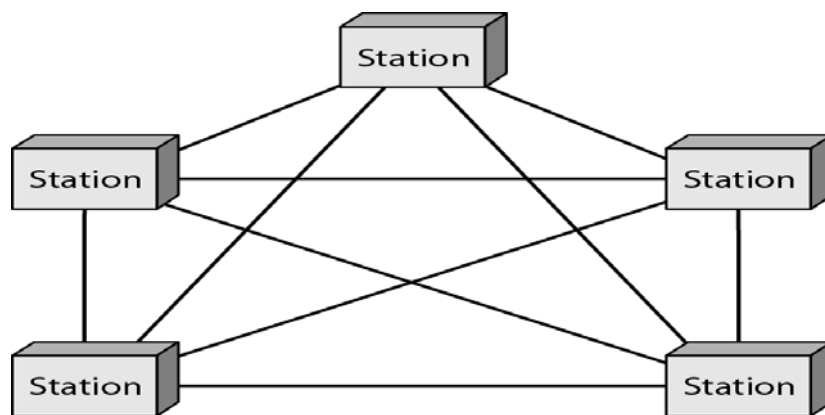
Mesh Topology- In a mesh topology, every device has a dedicated point-to-point link to every other device. The term dedicated means that the link carries traffic only between the two devices it connects.

We need $n(n - 1)$ physical links in a fully connected mesh network with n nodes. If each physical link allows communication in both directions (duplex mode), 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 (I/O) ports (see Figure) to be connected to the other $n - 1$ stations.

Advantages-

1. Use of dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problems that can occur when links must be shared by multiple devices.
2. A mesh topology is robust. If one link becomes unusable, it does not incapacitate the entire system.
3. 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.
4. 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.



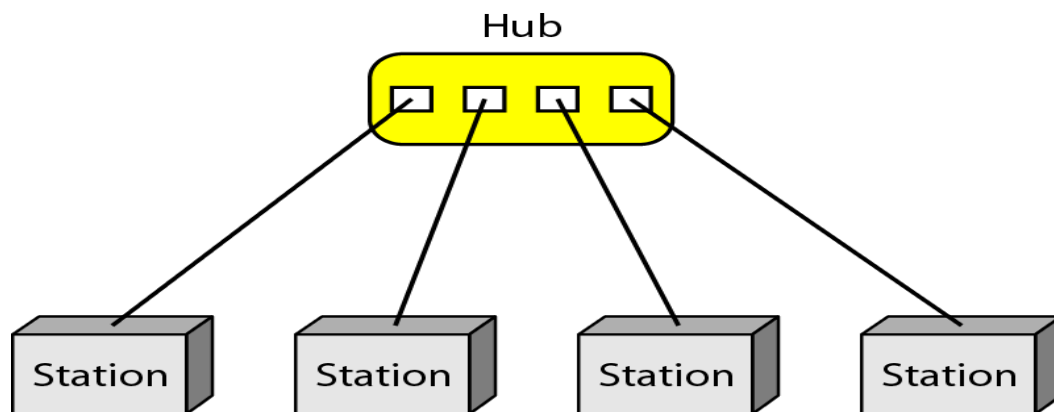
A fully connected mesh topology (five devices)

example- of a mesh topology is the connection of telephone regional offices in which each regional office needs to be connected to every other regional office.

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. 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 connecting four

stations Advantages

1. A star topology is less expensive than a mesh topology.
2. Each device needs only one link and one I/O port to connect. This factor 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.
3. robust. 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.

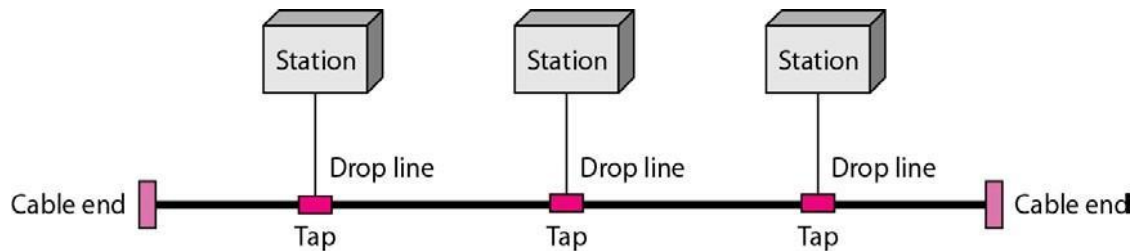
Disadvantage

1. The dependency of the whole topology on one single point, the hub. If the hub goes down, the whole system is dead.
2. more cabling is required in a star than in some other topologies (such as ring or bus).

The star topology is used in local-area networks (LANs), High-speed LANs often use a star topology with a central hub.

Bus Topology

A bus topology 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 a 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

1. Easy to install.
2. A bus uses less cabling than mesh or star topologies. 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

1. Difficult reconnection and fault isolation.
2. Difficult to add new devices.
3. 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 the given length of the cable.
4. Adding new devices requires modification or replacement of the backbone.
5. A fault or break in the bus cable stops all transmission. 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. Traditional Ethernet LANs can use a bus topology, but they are less popular now.

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.

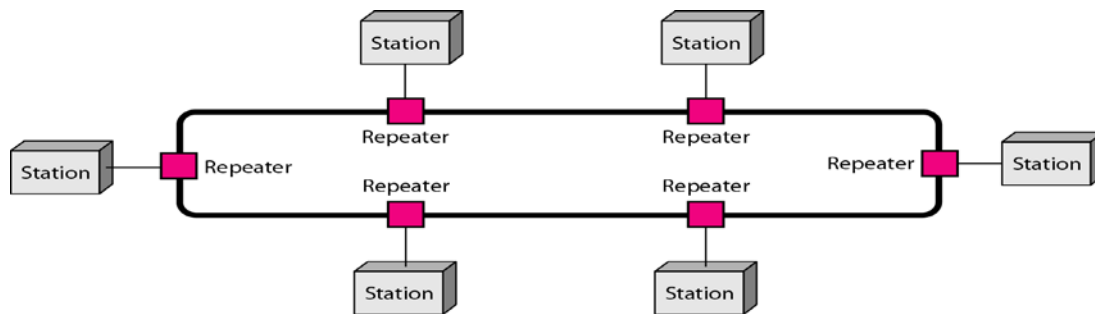


fig: A ring topology connecting six stations

Advantages

1. 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.
2. 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.

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.

NETWORK TYPES

Different types of networks

Local Area Network (LAN)-

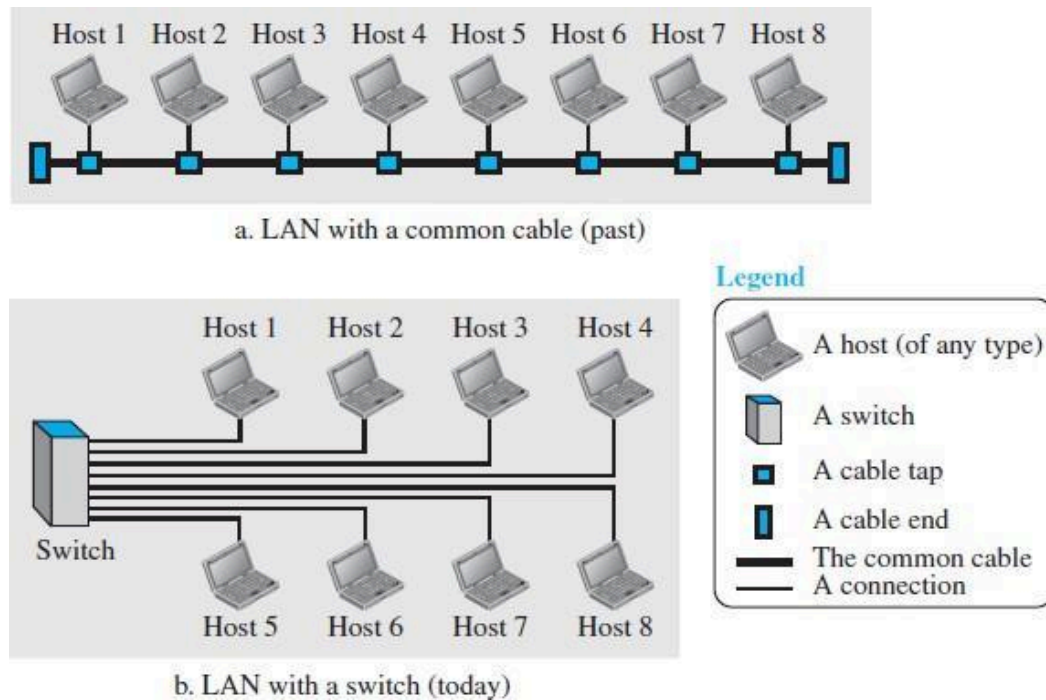


fig: An isolated LAN in the past and today

A local area network (LAN) is usually privately owned and connects some hosts in a single office, building, or campus. Depending on the needs of an organization,

A LAN can be as simple as two PCs and a printer in someone's home office, or it can extend throughout a company and include audio and video devices.

Each host in a LAN has an identifier, an address, that uniquely defines the host in the LAN. A packet sent by a host to another host carries both the source host's and the destination host's addresses.

In the past, all hosts in a network were connected through a common cable, which meant that a packet sent from one host to another was received by all hosts. The intended recipient kept the packet; the others dropped the packet.

Today, most LANs use a smart **connecting switch**, which is able to recognize the destination address of the packet and guide the packet to its destination without sending it to all other hosts. The switch alleviates the traffic in the LAN and allows more than one pair to communicate with each other at the same time if there is no common source and destination among them.

Wide Area Network

A wide area network (WAN) is also an interconnection of devices capable of communication. However, it. We see two distinct examples of WANs today: point-to-point WANs and switched WANs.

Point-to-Point WAN

A point-to-point WAN is a network that connects two communicating devices through a transmission media (cable or air).

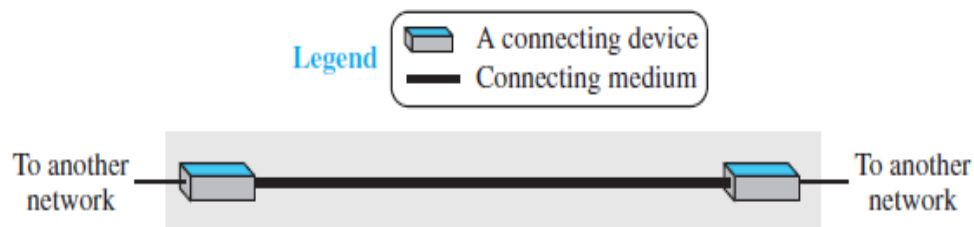


fig:Point-to-Point WAN

Switched WAN

A switched WAN is a network with more than two ends. A switched WAN is used in the backbone of global communication today. We can say that a switched WAN is a combination of several point-to-point WANs that are reconnected by switches.

Computer Communication Network

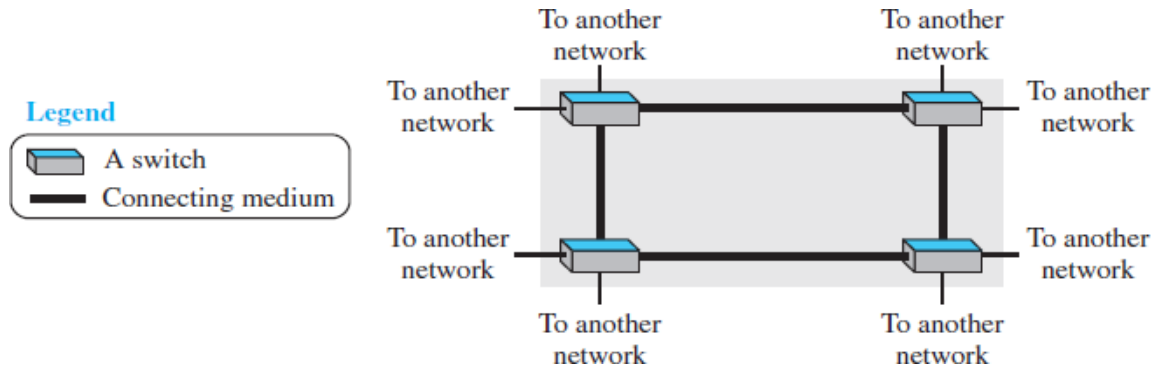


fig: A switched

WAN LAN VS WAN

<ol style="list-style-type: none">1. LAN is normally limited in size, spanning a single office, a building, or a campus.2. LAN interconnects hosts3. LAN is normally privately owned by the organization that uses it	<ol style="list-style-type: none">1. WAN has a wide geographical span, spanning a town, a state, a country, or even the world2. WAN interconnects connecting devices such as switches, routers, or modems3. a WAN is normally created and run by communication companies and leased by an organization that uses it.
---	--

Internetwork

Today, it is very rare to see a LAN or a WAN in isolation; they are connected to one another. When two or more networks are connected, they make an internetwork, or internet.

example- Assume that an organization has two offices, one on the east coast and the other on the west coast. Each office has a LAN that allows all employees in the office to communicate with each other. To make the communication between employees at different offices possible, the management leases a point-to-point dedicated WAN from a service provider, such as a telephone company, and connects the two LANs. Now the company has an internetwork, or a private internet (with lowercase i). Communication between offices is now possible.

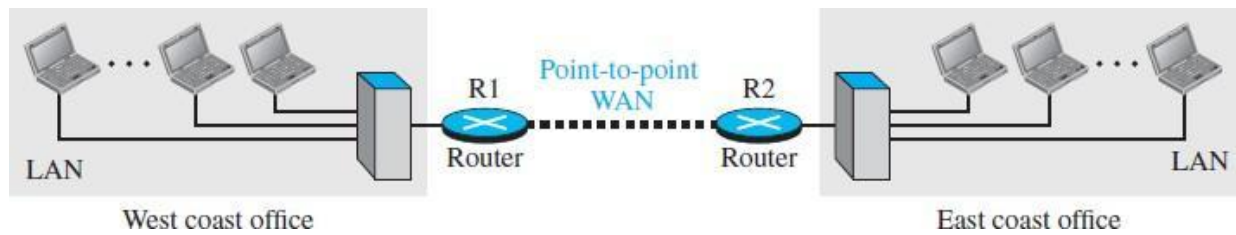


fig: A network made of two LAN and point-to-point dedicated WAN

When a host in the west coast office sends a message to another host in the same office, the router blocks the message, but the switch directs the message to the destination. On the other hand, when a host on the west coast sends a message to a host on the east coast, router R1 routes the packet to router R2, and the packet reaches the destination. Figure shows another internet with several LANs and WANs connected. One of the WANs is a switched WAN with four switches.

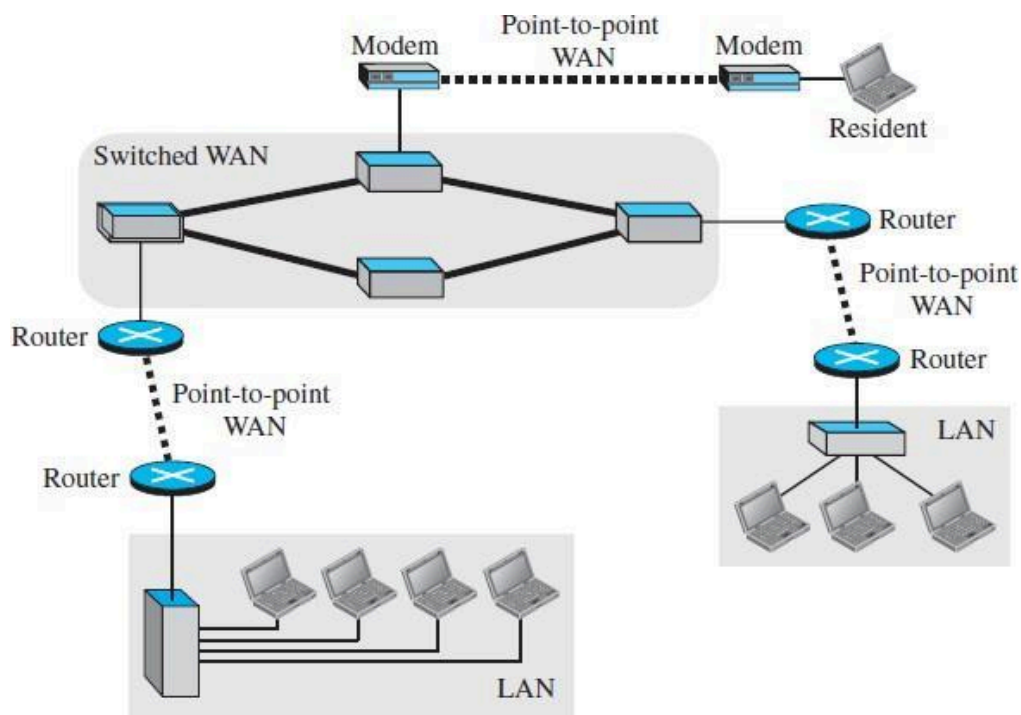


fig: A heterogeneous network made of four WANs and three LANs

Switching

An internet is a switched network in which a switch connects at least two links together. A switch needs to forward data from a network to another network when required. The two most common types of switched networks are circuit-switched and packet-switched networks.

Circuit-Switched Network

In a circuit-switched network, a dedicated connection, called a circuit, is always available between the two end systems; the switch can only make it active or inactive (continuous communication between two telephone). FIG shows a very simple switched network that connects four telephone sets at each end. We have used telephone sets instead of computers as an end system because circuit switching was very common in telephone networks in the past,

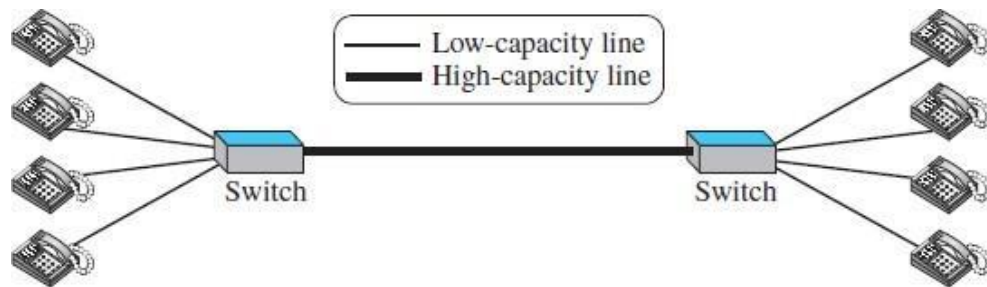


fig: Circuit-Switched Network

The thick line connecting two switches is a high-capacity communication line that can handle four voice communications at the same time; the capacity can be shared between all pairs

of telephone sets. The switches used in this example have forwarding tasks but no storing capability.

Let us look at two cases.

In the first case, all telephone sets are busy; four people at one site are talking with four people at the other site; the capacity of the thick line is fully used.

In the second case, only one telephone set at one side is connected to a telephone set at the other side; only one-fourth of the capacity of the thick line is used. This means that a circuit-switched network is efficient only when it is working at its full capacity; most of the time, it is inefficient because it is working at partial capacity.

The reason to make the capacity of the thick line four times the capacity of each voice line is that we do not want communication to fail when all telephone sets at one side want to be connected with all telephone sets at the other side.

Packet-Switched Network

In a computer network, the communication between the two computers is done in blocks of data called packets.

This allows switches to function for both storing and forwarding because a packet is an independent entity that can be stored and sent later. Fig shows a small packet-switched network that connects four computers at one site to four computers at the other site.

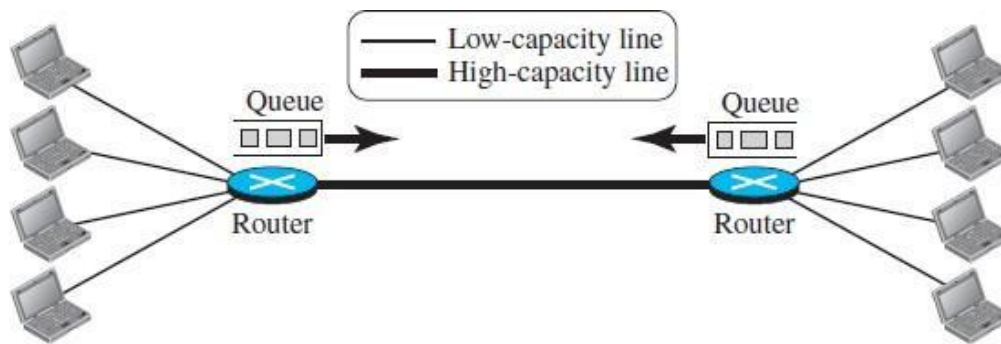


fig: Packet-Switched Network

A router in a packet-switched network has a queue that can store and forward the packet.

Example- Now assume that the capacity of the thick line is only twice the capacity of the data line connecting the computers to the routers.

If only two computers (one at each site) need to communicate with each other, there is no waiting for the packets. However, if packets arrive at one router when the thick line is already working at its full capacity, the packets should be stored and forwarded in the order they arrived. The two simple examples show that a packet-switched network is more efficient than a circuit-switched network, but the packets may encounter some delays.

The Internet

An internet (note the lowercase i) is two or more networks that can communicate with each other. The most notable internet is called the Internet (uppercase I), and is composed of thousands of interconnected networks.

Figure 1.15 shows a conceptual (not geographical) view of the Internet. The figure shows the Internet as several backbones, provider networks, and customer networks. At the top level, the backbones are large networks owned by some communication companies such as Sprint, Verizon (MCI), AT&T, and NTT. The backbone networks are connected through some complex switching systems, called peering points. At the second level, there are smaller networks, called provider networks, that use the services of the backbones for a fee. The provider networks are connected to backbones and sometimes to other provider networks.

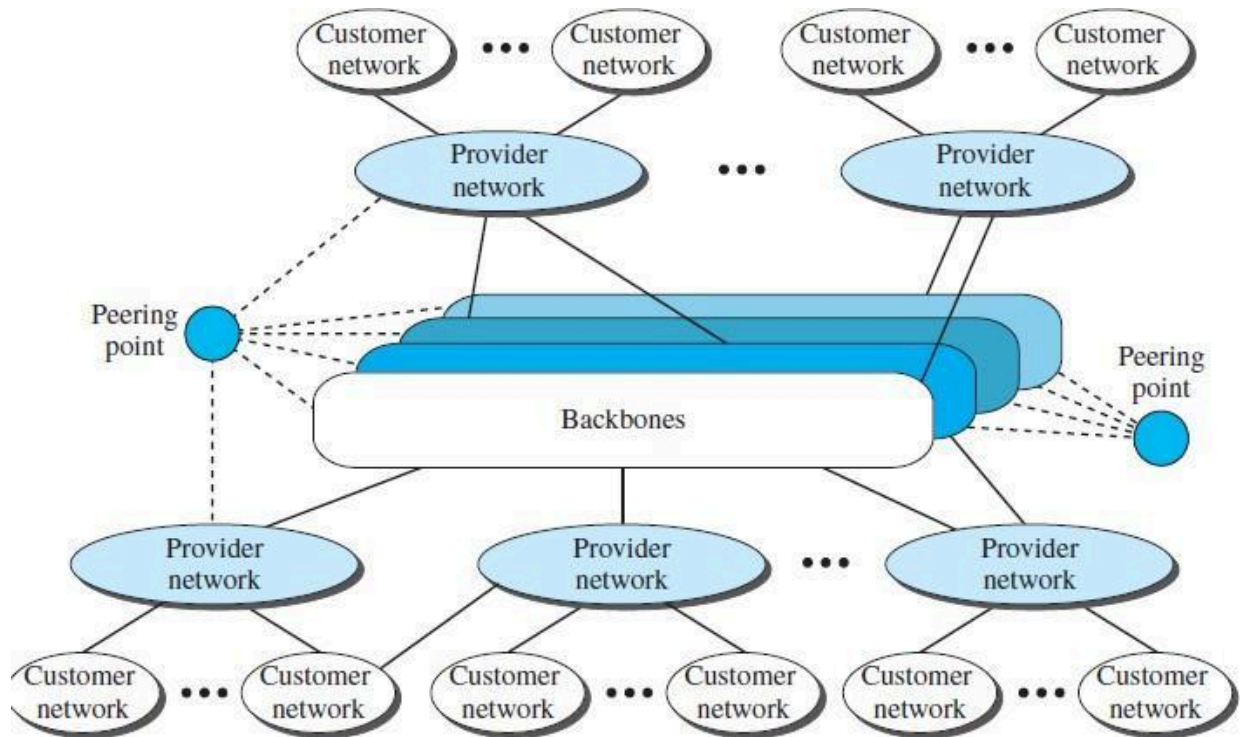


fig: The internet today

The customer networks are networks at the edge of the Internet that actually use the services provided by the Internet. They pay fees to provider networks for receiving services. Backbones and provider networks are also called Internet Service Providers (ISPs). The backbones are often referred to as international ISPs; the provider networks are often referred to as national or regional ISP.

Network Models

Protocol Layering

Protocol defines the rules that both the sender and receiver and all intermediate devices need to follow to be able to communicate effectively.

When communication is simple, we may need only one simple protocol; when the communication is complex, we may need to divide the task between different layers, in which case we need a protocol at each layer, or protocol layering.

Let us develop two simple scenarios to better understand the need for protocol layering.

Scenarios

First Scenario

In the first scenario, communication is so simple that it can occur in only one layer. Assume Maria and Ann are neighbors with a lot of common ideas. Communication between Maria and Ann takes place in one layer, face to face, in the same language, as shown in Figure

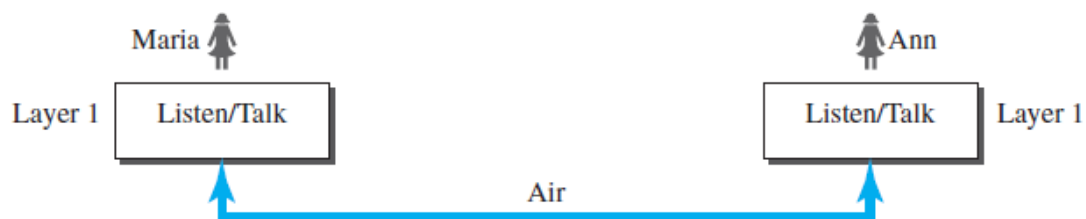


fig: single layer protocol

Second Scenario

In the second scenario, we assume that Ann is offered a higher-level position in her company, but needs to move to another branch located in a city very far from Maria. The two friends still want to continue their communication and exchange ideas because they have come up with an innovative project to start a new business when they both retire. They decide to continue their conversation using regular mail through the post office. However, they do not want their ideas to be revealed by other people if the letters are intercepted. They agree on an encryption/decryption technique. The sender of the letter encrypts it to make it unreadable by an intruder; the receiver of the letter decrypts it to get the original letter.

Now we can say that the communication between Maria and Ann takes place in three layers, as shown in Figure . We assume that Ann and Maria each have three machines (or robots) that can perform the task at each layer.

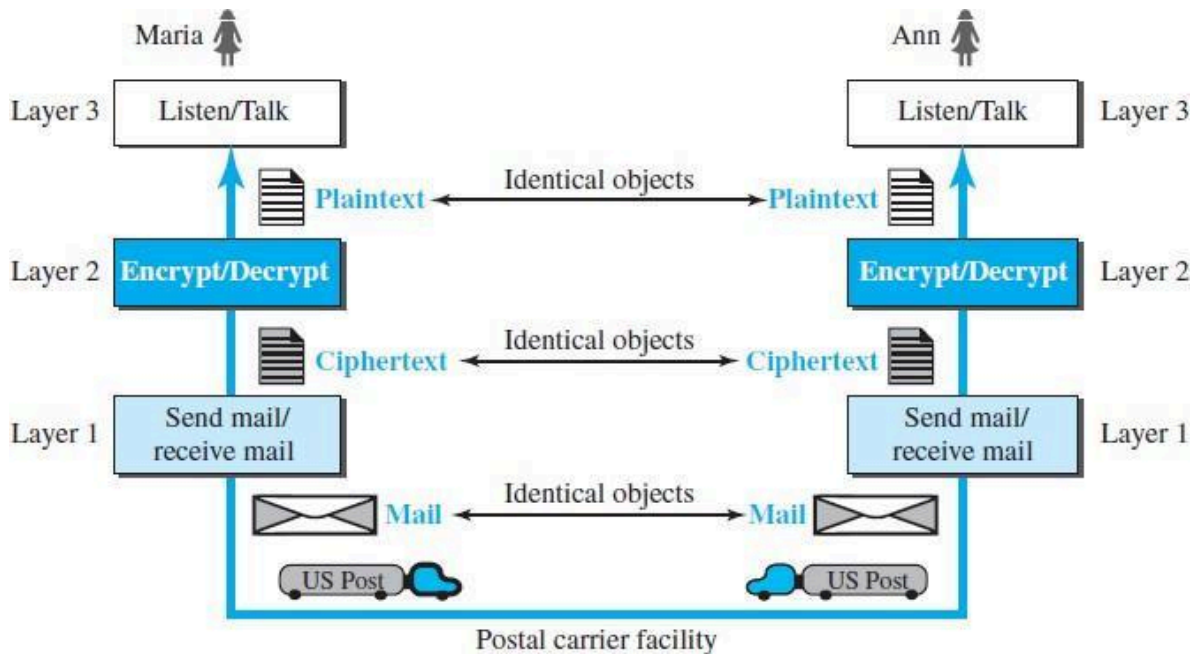


fig: A three layer protocol

Assume that Maria sends the first letter to Ann. Maria talks to the machine at the third layer although the machine is Ann and is listening to her. The third layer machine listens to what Maria says and creates the plaintext (a letter in English), which is passed to the second layer machine.

The second layer machine takes the plaintext, encrypts it, and creates the ciphertext, which is passed to the first layer machine. The first layer machine, presumably a robot, takes the ciphertext, puts it in an envelope, adds the sender and receiver addresses, and mails it.

At Ann's side, the first layer machine picks up the letter from Ann's mail box, recognizing the letter from Maria by the sender address. The machine takes out the ciphertext from the envelope and delivers it to the second layer machine. The second layer machine decrypts the message, creates the plaintext, and passes the plaintext to the third-layer machine. The third layer machine takes the plaintext and reads it as though Maria is speaking.

Need for protocol layering

1) Protocol layering enables us to divide a complex task into several smaller and simpler tasks.

For example, from fig, we could have used only one machine to do the job of all three machines. However, if the encryption/decryption done by the machine is not enough to protect their secrecy, they would have to change the whole machine. In the present situation, they need to change only the second layer machine; the other two can remain the same. This is referred to as modularity. Modularity in this case means independent layers.

2) A layer (module) can be defined as a black box with inputs and outputs, without concern about how inputs are changed to outputs. If two machines provide the same outputs when given the same inputs, they can replace each other.

For example, Ann and Maria can buy these second layer machine from two different manufacturers. As long as the two machines create the same ciphertext from the same plaintext and vice versa, they do the job.

advantages

1) Protocol layering allows to separate the services from the implementation. Lower layer gives the services to the upper layer; we don't care about how the layer is implemented.

For example, Maria may decide not to buy the machine (robot) for the first layer; she can do the job herself. As long as Maria can do the tasks provided by the first layer, in both directions, the communications system works.

2) Protocol layering in the Internet, is that communication does not always use only two end systems; there are intermediate systems that need only some layers, but not all layers. If we did not use protocol layering, we would have to make each intermediate system as complex as the end systems, which makes the whole system more expensive.

Principles of Protocol Layering

First Principle The first principle dictates that if we want bidirectional communication, we need to make each layer so that it is able to perform **two opposite tasks**, one in each direction.

For example, the third layer task is to listen (in one direction) and talk (in the other direction). The second layer needs to be able to encrypt and decrypt. The first layer needs to send and receive mail.

Second Principle The second principle that we need to follow in protocol layering is that the **two objects** under each layer at both sites should be **identical**.

For example, the object under layer 3 at both sites should be a plaintext letter. The object under layer 2 at both sites should be a cipher text letter. The object under layer 1 at both sites should be a piece of mail.

Logical Connections

After following the above two principles, we can think about logical connection between each layer as shown in Figure . This means that we have layer-to-layer communication. Maria and Ann can think that there is a logical (imaginary) connection at each layer through which they can send the object created from that layer. We will see that the concept of logical connection will help us better understand the task of layering we encounter in data communication and networking.

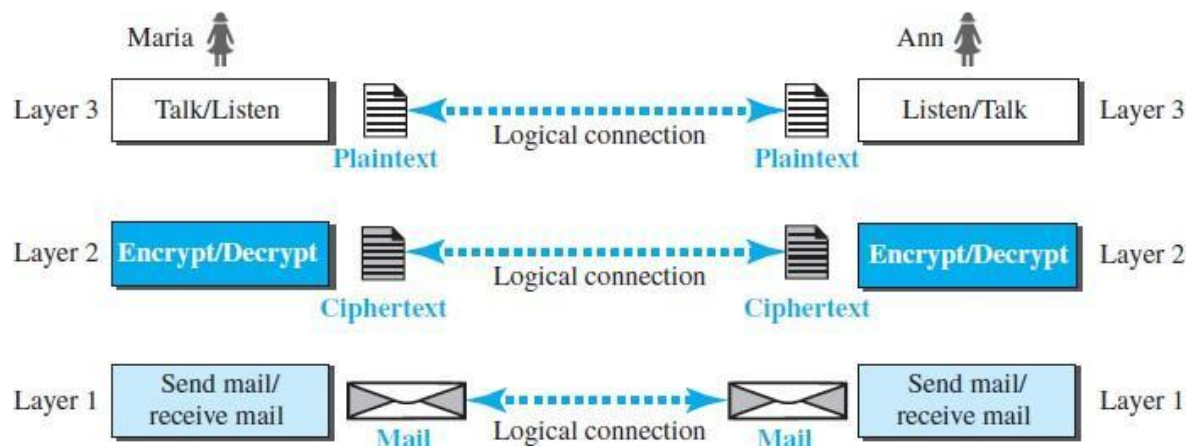


fig: Logical connection between peer layer

TCP/IP PROTOCOL SUITE

TCP/IP is a protocol suite (a set of protocols organized in different layers) used in the Internet today. It is a hierarchical protocol made up of interactive modules, each of which provides a specific functionality. The term hierarchical means that each upper level protocol is supported by the services provided by one or more lower level protocols. The original TCP/IP protocol suite was defined as four software layers built upon the hardware. Today, however, TCP/IP is thought of as a five-layer model. Figure shows both configurations.

Layered Architecture

To show how the layers in the TCP/IP protocol suite are involved in communication between two hosts, we assume that we want to use the suite in a small Internet made up of three LANs

(links), each with a link-layer switch. We also assume that the links are reconnected by one router, as shown in Figure

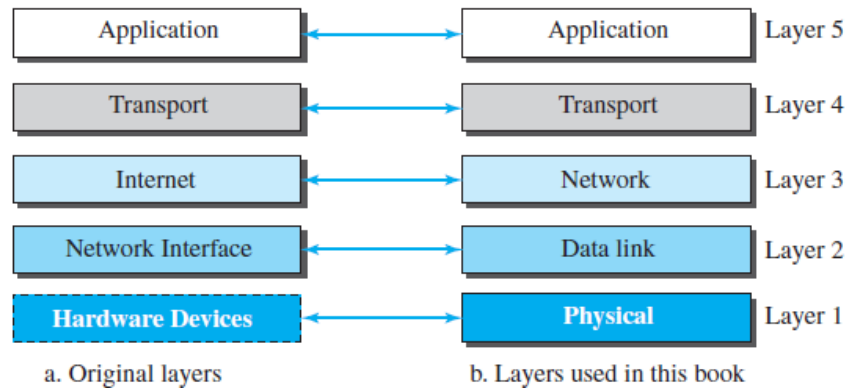


fig:layers in TCP/IP protocol suite

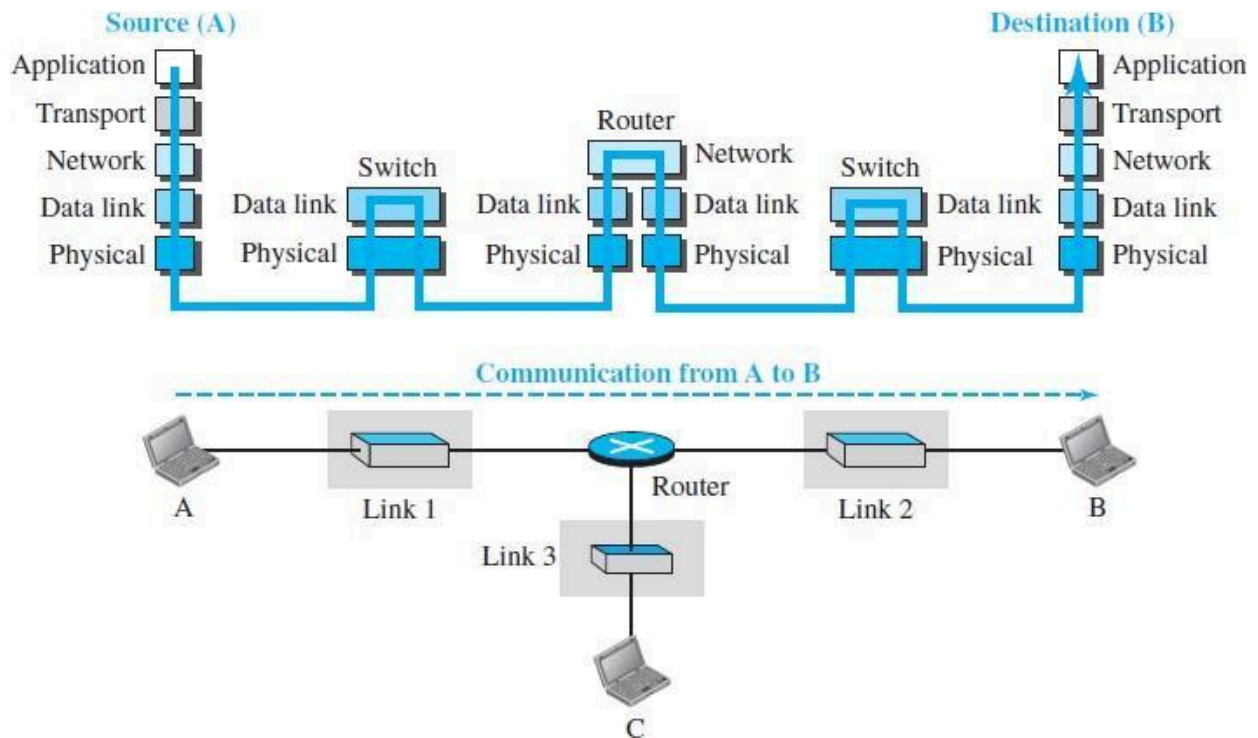


fig:Communication through an internet

Assume that computer A communicates with computer B. As the figure shows, five communicating devices in this communication: source host (computer A), the link-layer switch in link1, the router, the link-layer switch in link2, and the destination host (computer B).

The source host needs to create a message in the application layer and send it down the layers so that it is physically sent to the destination host. The destination host needs to receive the communication at the physical layer and then deliver it through the other layers to the application layer.

The router is involved in only three layers; there is no transport or application layer in a router. Although a **router is always involved in one network layer**, it is involved in n combinations of link and physical layers in which n is the number of links the router is connected to. The reason is that each link may use its own data-link or physical protocol.

For example, in the above figure, the router is involved in three links, but the message sent from source A to destination B is involved in **two links**. Each link may be using **different link-layer and physical-layer protocols**; the router needs to receive a packet from link 1 based on one pair of protocols and deliver it to link 2 based on another pair of protocols.

A link-layer switch in a link, however, is involved only in two layers, data-link and physical. Although each switch in the above figure has two different connections, the connections are in the **same link**, which uses **only one set of protocols**. This means that, unlike a router, a link-layer switch is involved only in one data-link and one physical layer.

Layers in the TCP/IP Protocol Suite

To better understand the duties of each layer, we need to think about the logical connections between layers.

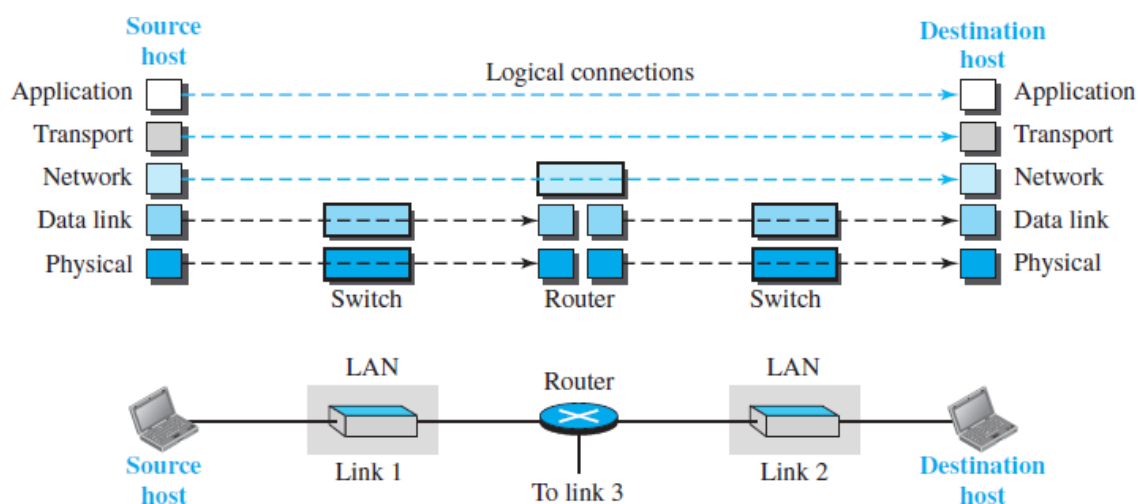


fig:Figureshowslogicalconnectionsinoourseimpleinternet.

Using logical connections makes it easier to think about the duty of each layer. As the figureshows, the duty of the application, transport, and network layers is **end-to-end**. However, the duty of the data-link and physical layers is **hop-to-hop**, in which a hop is a host or router.

In other words, the domain of duty of the top three layers is the **internet**, and the domain of duty of the two lower layers is the **link**.

Another way of thinking of the logical connections is to think about the data unit created from each layer. In the top three layers, the data unit (packets) should not be changed by any router or link-layer switch. In the bottom two layers, the packet created by the host is changed only by the routers, not by the link-layer switches.

Fig shows the second principle discussed previously for protocol layering. We show the identical objects below each layer related to each device.

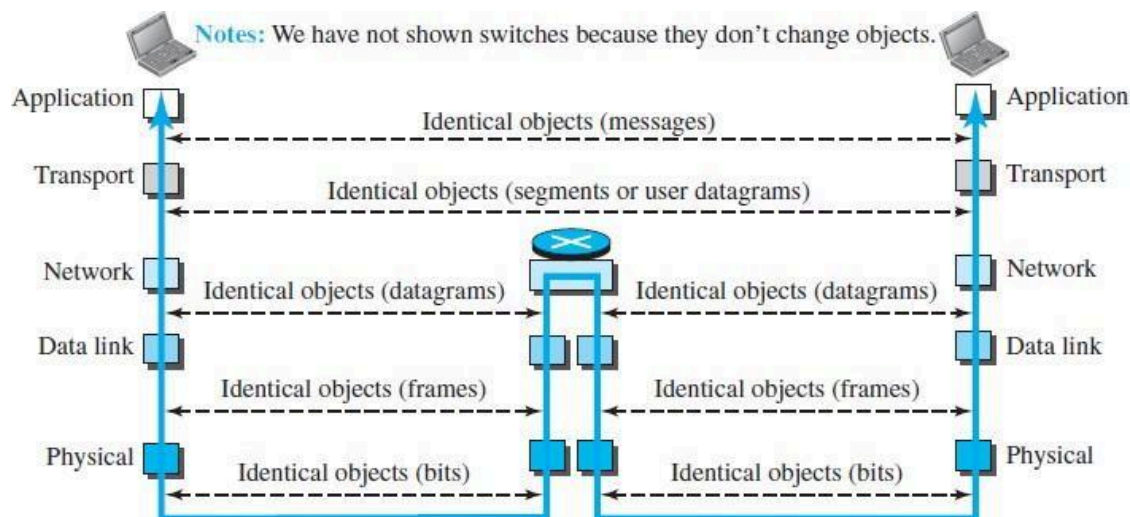


fig:identical objects in the TCP/IP protocol suite

Note that, although the logical connection at the network layer is between the two hosts, we can only say that identical objects exist between two hops in this case because a router may fragment the packet at the network layer and send more packets than received. Note that the link between two hops does not change the object.

Description of Each Layer

Physical Layer

Physical layer is responsible for **carrying individual bits** in a frame across the link. Although the physical layer is the lowest level in the TCP/IP protocol suite, the communication between two devices at the physical layer is still a logical communication because there is another, hidden layer, the transmission media, under the physical layer.

Two devices are connected by a transmission medium (cable or air). Transmission medium does not carry bits, **it carries electrical or optical signals**. So the bits received in a frame from the data-link layer are transformed and sent through the transmission media, but we can think that the logical unit between two physical layers in two devices is a bit. There are several protocols that transform a bit to a signal.

The physical layer of TCP/IP describes hardware standards such as IEEE 802.3, the specification for Ethernet network media, and RS-232, the specification for standard pin connectors.

The following are the main responsibilities of the physical layer

Definition of Hardware Specifications, Encoding and Signaling, Data Transmission and Reception, Topology and Physical Network Design

Data-link Layer

Internet is made up of several links (LANs and WANs) connected by routers. The data-link layer is responsible for taking the datagram and moving it across the link. (node to node communication)

The link can be a wired LAN with a link-layer switch, a wireless LAN, a wired WAN, or a wireless WAN. We can also have different protocols used with any link type.

In each case, the data-link layer is responsible for moving the packet through the link. TCP/IP does not define any specific protocol for the data-link layer. It supports all the standard and proprietary protocols. The data-link layer takes a datagram and encapsulates it in a packet called a **frame**.

Each link-layer protocol provides a different service like framing, Flow control, Error control and congestion control.

Network Layer

The network layer is responsible for creating a connection between the source computer and the destination computer. The communication at the network layer is **host-to-host**. However, since there can be several routers from the source to the destination, the routers in the path are responsible for choosing the **best route** for each packet.

The network layer is responsible packetizing and routing and forwarding the packet through possible routes. Other services are error and flow control, congestion control.

The network layer in the Internet includes the main protocol, Internet Protocol (IP), that defines the format of the packet, **called a datagram** at the network layer. IP also defines the format and the structure of addresses used in this layer.

IP is also responsible for routing a packet from its source to its destination, which is achieved by each router forwarding the datagram to the next router in its path.

IP is a connectionless protocol that provides no flow control, no error control, and no congestion control services. This means that if any of these services is required for an application, the application should rely only on the transport-layer protocol.

The network layer also includes unicast (one-to-one) and multicast (one-to-many) routing protocols. A routing protocol does not take part in routing (it is the responsibility of IP), but it creates forwarding tables for routers to help them in the routing process. The network layer also has some auxiliary protocols that help IP in its delivery and routing tasks.

The Internet Control Message Protocol (ICMP) helps IP to report some problems when routing a packet. The Internet Group Management Protocol (IGMP) is another protocol that helps IP in multitasking. The Dynamic Host Configuration Protocol (DHCP) helps IP to get the network-layer address for a host. The Address Resolution Protocol (ARP) is a protocol that helps IP to find the link-layer address of a host or a router when its network-layer address is given.

Transport Layer

The logical connection at the transport layer is also end-to-end. The transport layer at the source host gets the message from the application layer, encapsulates it in a transport layer packet (called a segment or a user datagram in different protocols) and sends it, through the logical (imaginary) connection, to the transport layer at the destination host.

The transport layer is responsible for giving services to the application layer: to get a message from an application program running on the source host and deliver it to the corresponding application program on the destination host. **(process to process communication)**

There are more than one protocol in the transport layer, which means that each application program can use the protocol that best matches its requirement. There are a few transport-layer protocols in the Internet, each designed for some specific task.

The main protocol, **Transmission Control Protocol (TCP)**, is a connection-oriented protocol that first establishes a logical connection between transport layers at two hosts before transferring data. It creates a logical pipe between two TCPs for transferring a stream of bytes. TCP provides flow control (matching the sending data rate of the source host with the receiving data rate of the destination host to prevent overwhelming the destination), error control (to guarantee that the segments arrive at the destination without error and resending the corrupted ones), and congestion control to reduce the loss of segments due to congestion in the network.

User Datagram Protocol (UDP), is a connectionless protocol that transmits user datagrams without first creating a logical connection. In UDP, each user datagram is an independent entity without being related to the previous or the next one (the meaning of the term connectionless). UDP is a simple protocol that does not provide flow, error, or congestion control.

Its simplicity, which means small overhead, is attractive to an application program that needs to send short messages and cannot afford the retransmission of the packets involved in TCP, when a packet is corrupted or lost.

A new protocol, Stream Control Transmission Protocol (SCTP) is designed to respond to new applications that are emerging in the multimedia.

Application Layer

As Figure shows, the logical connection between the two application layers is end-to-end. The two application layers exchange messages between each other as though there were a bridge between the two layers. However, communication is done through all the layers.

Communication at the application layer is between two processes (two programs running at this layer). To communicate, a process sends a request to the other process and receives a response. Process-to-process communication is the duty of the application layer.

The application layer in the Internet includes many predefined protocols.

- 1) The Hypertext Transfer Protocol (HTTP) is a vehicle for accessing the World Wide Web (WWW).
- 2) The Simple Mail Transfer Protocol (SMTP) is the main protocol used in electronic mail (e-mail) service.

- 3) The File Transfer Protocol (FTP) is used for transferring files from one host to another.
- 4) The Terminal Network (TELNET) and Secure Shell (SSH) are used for accessing a site remotely.
- 5) The Simple Network Management Protocol (SNMP) is used by an administrator to manage the Internet at global and local levels.
- 6) The Domain Name System (DNS) is used by other protocols to find the network-layer addresses of a computer.
- 7) The Internet Group Management Protocol (IGMP) is used to collect membership in a group.

Encapsulation and Decapsulation

One of the important concepts in protocol layering in the Internet is encapsulation/decapsulation. Figure shows this concept for the small Internet.

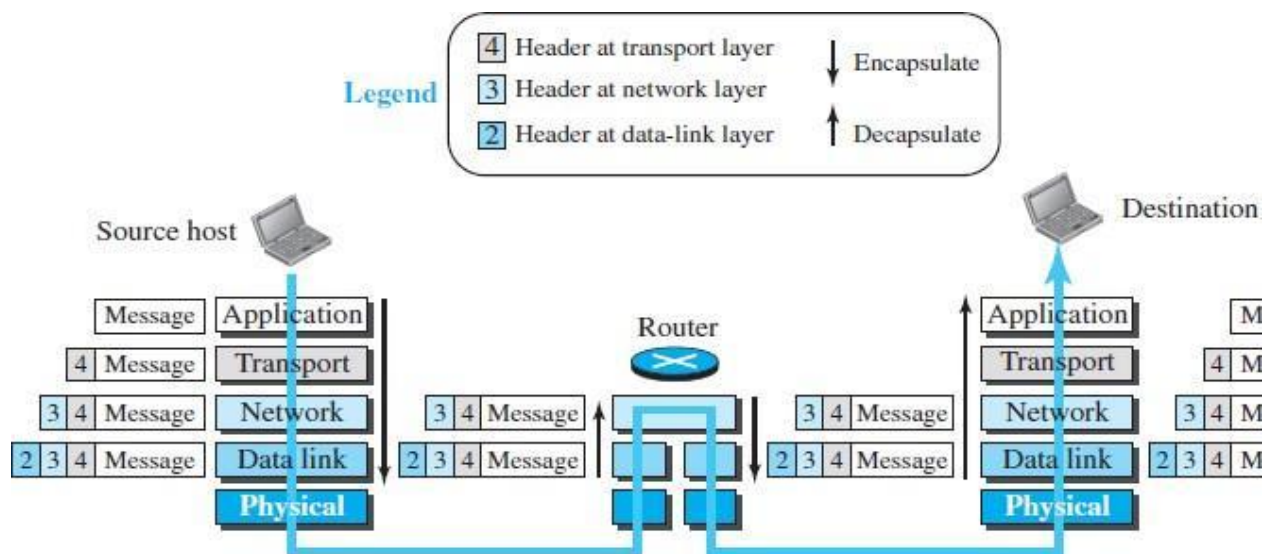


fig:encapsulation/decapsulation

We have not shown the layers for the link-layer switches because no encapsulation/decapsulation occurs in this device. Figures show the encapsulation in the source host, decapsulation in the destination host, and encapsulation and decapsulation in the router.

Encapsulation at the Source Host

At the source, we have only encapsulation.

1. At the application layer, the data to be exchanged is referred to as a message. A message normally does not contain any header or trailer, but if it does, we refer to the whole as the message. The message is passed to the transport layer.
2. The transport layer takes the message as the payload, the load that the transport layer should take care of. It adds the transport layer header to the payload, which contains the identifiers of the source and destination application programs that want to communicate plus some more information that is needed for the end-to-end delivery of the message, such as information needed for flow, error control, or congestion control. The result is the transport-layer packet, which is called the segment (in TCP) and the user datagram (in UDP). The transport layer then passes the packet to the network layer.
3. The network layer takes the transport-layer packet as data or payload and adds its own header to the payload. The header contains the addresses of the source and destination hosts and some more information used for error checking of the header, fragmentation information, and so on. The result is the network-layer packet, called a datagram. The network layer then passes the packet to the data-link layer.
4. The data-link layer takes the network-layer packet as data or payload and adds its own header, which contains the link-layer addresses of the host or the next hop (the router). The result is the link-layer packet, which is called a frame. The frame is passed to the physical layer for transmission.

Decapsulation and Encapsulation at the Router

At the router, we have both decapsulation and encapsulation because the router is connected to two or more links.

1. After the set of bits are delivered to the data-link layer, this layer decapsulates the datagram from the frame and passes it to the network layer.
2. The network layer only inspects the source and destination addresses in the datagram header and consults its forwarding table to find the next hop to which the datagram is to be delivered. The contents of the datagram should not be changed by the network layer in the router unless there is a need to fragment the datagram if it is too big to be passed through the next link. The datagram is then passed to the data-link layer of the next link.
3. The data-link layer of the next link encapsulates the datagram in a frame and passes it to the physical layer for transmission.

Decapsulation at the Destination Host

At the destination host, each layer only decapsulates the packet received, removes the payload, and delivers the payload to the next-higher layer protocol until the message reaches the application layer. It is necessary to say that decapsulation in the host involves error checking.

Addressing

We have logical communication between pairs of layers in this model. Any communication that involves two parties needs two addresses: source address and destination address. Although it looks as if we need five pairs of addresses, one pair per layer, we normally have only four because **the physical layer does not need addresses**; the unit of data exchange at the physical layer is a bit, which definitely cannot have an address.

Figure 2.9 shows the addressing at each layer. At the application layer, we normally use names to define the site that provides services, such as `someorg.com`, or the e-mail address, such as `assomebody@coldmail.com`.

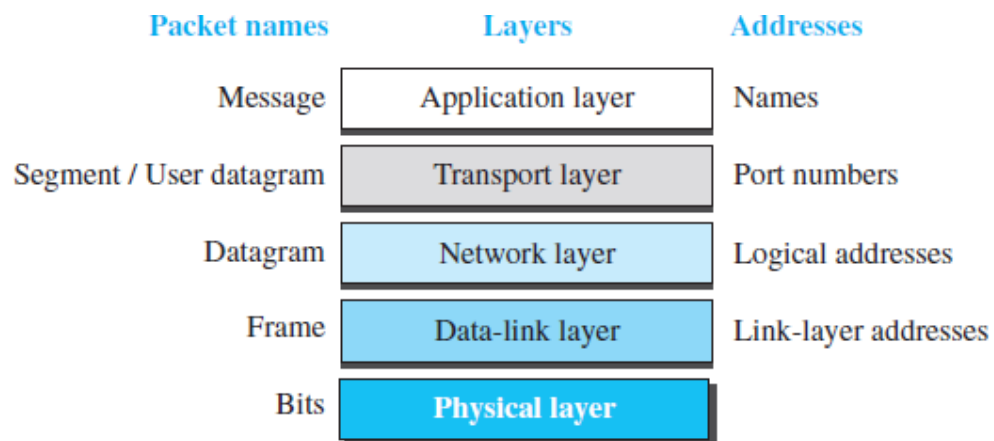


fig:Addressing in the TCP/IP Protocol suite

At the transport layer, addresses are called port numbers, and these define the application-layer programs at the source and destination. Port numbers are local addresses that distinguish between several programs running at the same time.

At the network layer, the addresses are global, with the whole Internet as the scope. A network-layer address uniquely defines the connection of a device to the Internet.

The link-layer addresses, sometimes called MAC addresses, are locally defined addresses, each of which defines a specific host or router in a network (LAN or WAN).

Multiplexing and Demultiplexing

TCP/IP protocol suite uses several protocols at some layers, we have multiplexing at the source and demultiplexing at the destination.

Multiplexing means that a protocol at a layer can encapsulate a packet from several next-higher layer protocols (one at a time); demultiplexing means that a protocol can decapsulate and deliver a packet to several next-higher layer protocols (one at a time). Figure shows the concept of multiplexing and demultiplexing at the three upper layers.

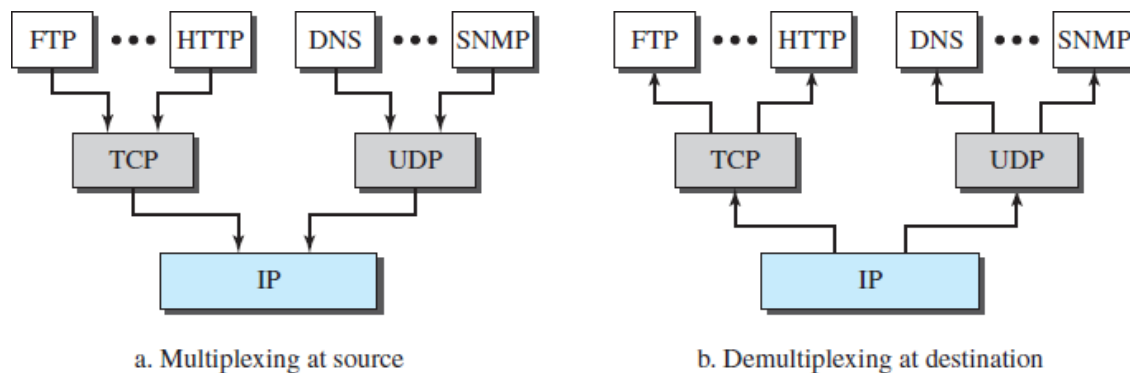


fig: multiplexing and demultiplexing

To be able to multiplex and demultiplex, a protocol needs to have a field in its header to identify to which protocol the encapsulated packets belong.

At the transport layer, either UDP or TCP can accept a message from several application-layer protocols.

At the network layer, IP can accept a segment from TCP or a user datagram from UDP. IP can also accept a packet from other protocols such as ICMP, IGMP, and so on.

At the data-link layer, a frame may carry the payload coming from IP or other protocols such as ARP.

THE OSI MODEL

An ISO standard that covers all aspects of network communications is the Open Systems Interconnection (OSI) model. It was first introduced in the late 1970s. An open system is a set of

protocols that allows any two different systems to communicate regardless of their underlying architecture.

The purpose of the OSI model is to show how to facilitate communication between different systems without requiring changes to the logic of the underlying hardware and software.

The **OSI model is not a protocol**; it is a model for understanding and designing a network architecture that is flexible, robust, and interoperable. The OSI model was intended to be the basis for the creation of the protocols in the OSI stack. The OSI model is a layered framework for the design of network systems that allows communication between all types of computer systems.

It consists of seven separate but related layers, each of which defines a part of the process of moving information across a network.

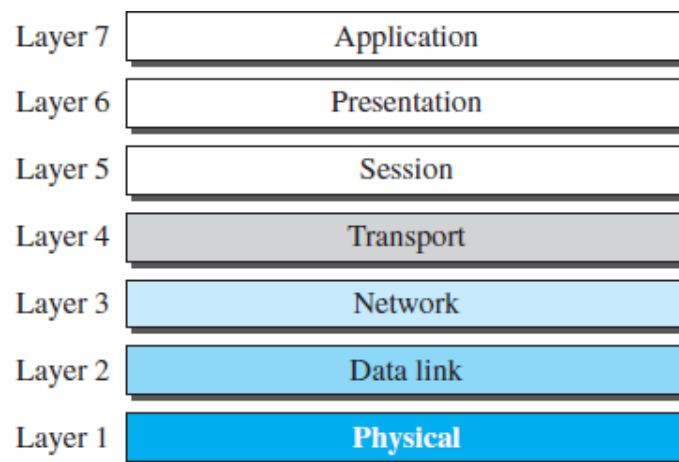


fig:OSI model OSI v

versus TCP/IP

When we compare the two models, we find that two layers, session and presentation, are missing from the TCP/IP protocol suite. These two layers were not added to the TCP/IP protocol suite after the publication of the OSI model. The application layer in the suite is usually considered to be the combination of three layers in the OSI model, as shown in Figure.

Two reasons were mentioned for this decision. First, TCP/IP has more than one transport-layer protocol. Some of the functionalities of the session layer are available in some of the transport-layer protocols.

Second, the application layer is not only one piece of software. Many applications can be developed at this layer. If some of the functionalities mentioned in the session and presentation layers are needed for a particular application, they can be included in the development of that piece of software.

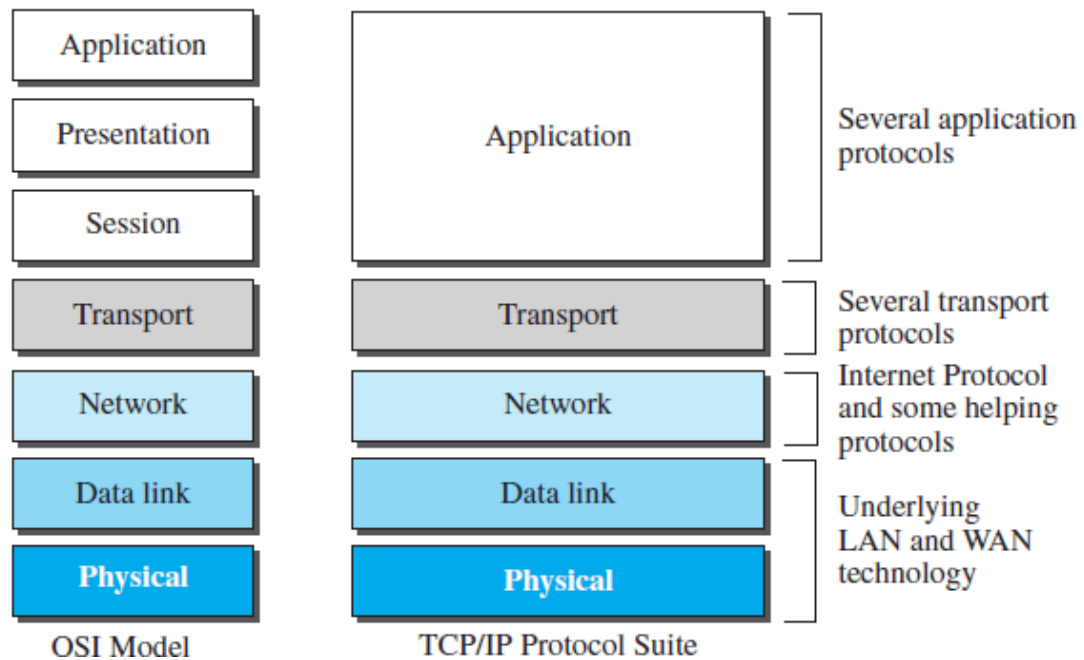


fig:TCP/IP and OSI model

