University of Basrah
College of Engineering   Computer Engineering Dept.

Computer Networks

4th Year

Computer Engineering

Title: Computer Networks
Code: CoE435
Stage: 4th Year
Lecturer: Dr. Abbas A. Jasim

Based on References:

Data Communication and Networks (2007) by B. Forouzan
Local Area Networks (2003) by B. Forouzan

Introduction to Computer Networks

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

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

Network Criteria

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

Performance

Performance can be measured in many ways, including transit time and response time. **Transit time** is the amount of time required for a message to travel from one device to another. **Response time** is the elapsed time between an inquiry and a response. The performance of a network depends on a number of factors, including the number of users, the type of transmission medium, the capabilities of the connected hardware, and the efficiency of the software.

Performance is often evaluated by two networking metrics: **throughput** and **delay**. We often need more throughput and less delay. However, these two criteria are often contradictory. If we try to send more data to the network, we may increase throughput but we increase the delay because of traffic congestion in the network.

Reliability

In addition to accuracy of delivery, network reliability is measured by the frequency of failure, the time it takes a link to recover from a failure, and the network's robustness in a catastrophe.

Security

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

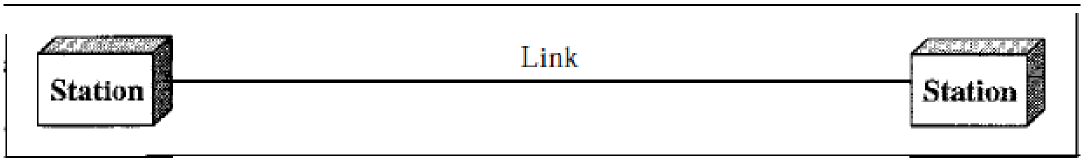
Physical Structures

Before discussing networks, we need to define some network attributes.

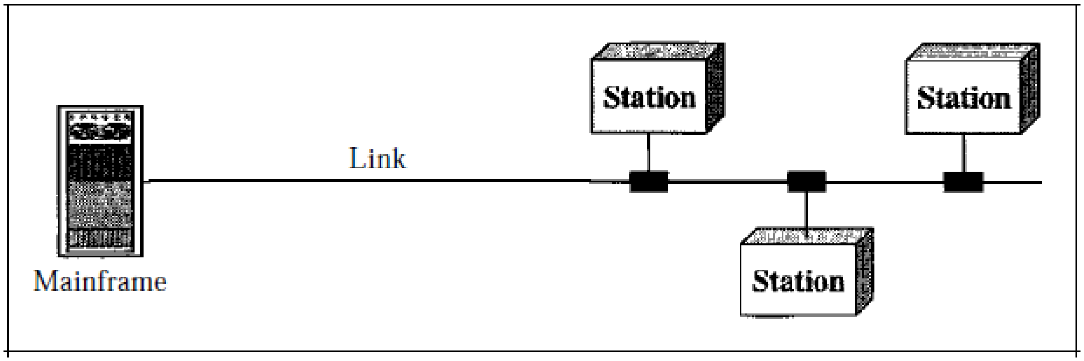
Type of Connection

A **network** is two or more devices connected through links. A **link** is a communications pathway that transfers data from one device to another. For communication to occur, two devices must be connected in some way to the same link at the same time.

There are two possible types of connections: point-to-point and multipoint.



a. Point-to-point



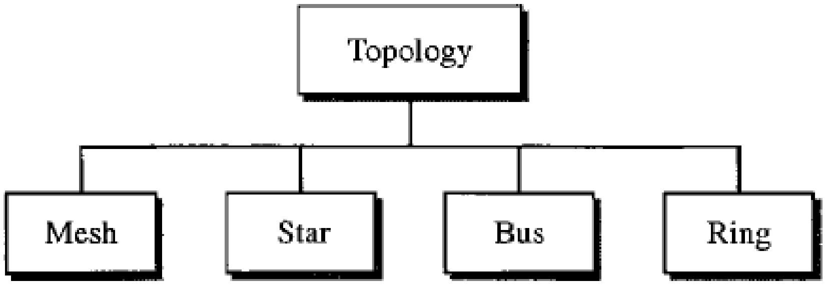
b. Multipoint

A **point-to-point** connection provides a dedicated link between two devices. The entire capacity of the link is reserved for transmission between those two devices. Most point-to-point connections use an actual length of wire or cable to connect the two ends, but other options, such as microwave or satellite links, are also possible. When you change television channels by infrared remote control,

A **multipoint** (also called multi drop) connection is one in which more than two specific devices share a single link. In a multipoint environment, the capacity of the channel is shared. If several devices can use the link simultaneously, it is a *spatially shared* connection. If users must take turns, it is a *timeshared* connection.

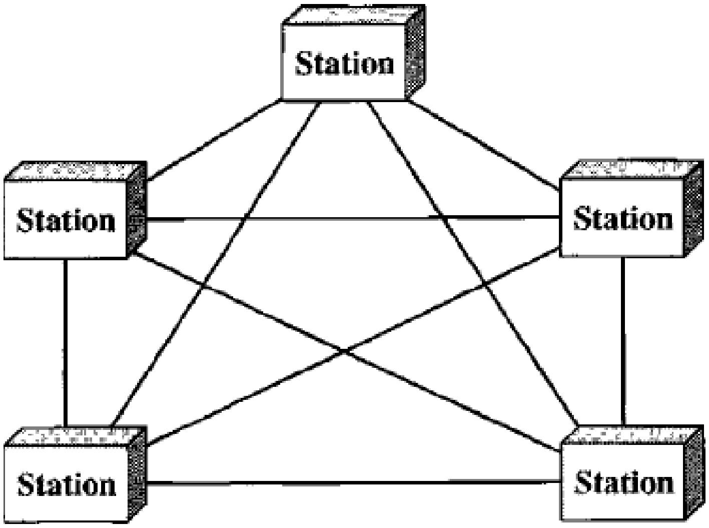
Physical Topology

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

In a mesh topology, every device has a dedicated point-to-point link to every other device. The term *dedicated* means that the link carries traffic only between the two devices it connects. To find the number of physical links in a fully connected mesh network with *n* nodes, we first consider that each node must be connected to every other node. Node 1 must be connected to *n* - 1 nodes, node 2 must be connected to *n* – 1 nodes, and finally node *n* must be connected to *n* - 1 nodes. We need *n(n* - 1) physical links. However, if each physical link allows communication in both directions (duplex mode), we can divide the number of links by 2. In other words, we can say that in a mesh topology, we need: *n(n* -1) /2 duplex-mode links.



A**dvantages** over other network topologies.

1- The use of dedicated links guarantees that each connection can carry its own data load, thus eliminating the traffic problems that can occur when links must be shared by multiple devices.

2- A mesh topology is robust. If one link becomes unusable, it does not incapacitate the entire system.

3- There is the advantage of privacy or security. When every message travels along a dedicated line, only the intended recipient sees it. Physical boundaries prevent other users from gaining access to messages.

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.

The main **disadvantages** of a mesh are related to the amount of cabling and the number of I/O ports required:

1- Because every device must be connected to every other device, installation and reconnection are difficult.

2- The sheer bulk of the wiring can be greater than the available space (in walls, ceilings, or floors) can accommodate.

3- The hardware required to connect each link (I/O ports and cable) can be prohibitively expensive.

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

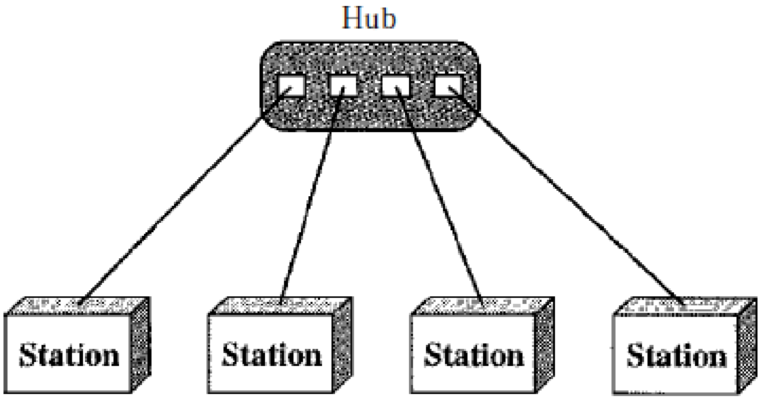
Star Topology

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. Unlike a mesh topology, a star topology does not allow direct traffic between devices. The controller acts as an exchange: If one device wants to send data to another, it sends the data to the controller, which then relays the data to the other connected device.

A star topology is less expensive than a mesh topology. In a star, each device needs only one link and one I/O port to connect it to any number of others. This factor also makes it easy to install and reconfigure. Far less cabling needs to be housed, and additions, moves, and deletions involve only one connection: between that device and the hub. Other advantages include robustness. If one link fails, only that link is affected. All other links remain active. This factor also lends itself to easy fault identification and fault isolation. As long as the hub is working, it can be used to monitor link problems and bypass defective links.

One big **disadvantage** of a star topology is the dependency of the whole topology on one single point, the hub. If the hub goes down, the whole system is dead.
Although a star requires far less cable than a mesh, each node must be linked to a central hub. For this reason, often more cabling is required in a star than in some

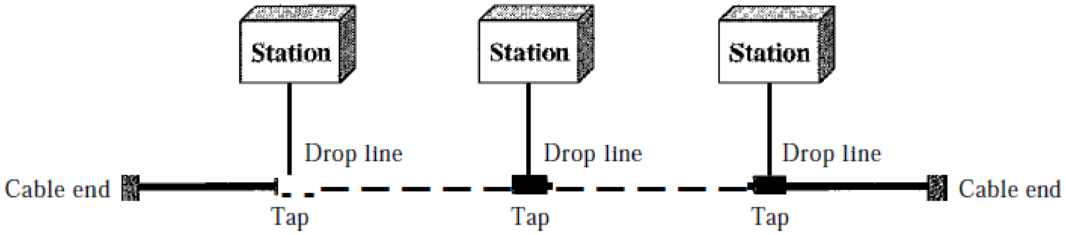
other topologies (such as ring or bus). The star topology is used in local-area networks (LANs).High-speed LANs often use a star topology with a central hub.



Then, Star LAN uses point to point unidirectional connection between stations and hub carrying base band or signals carried in guided or unguided medium respectively.

Bus Topology

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



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

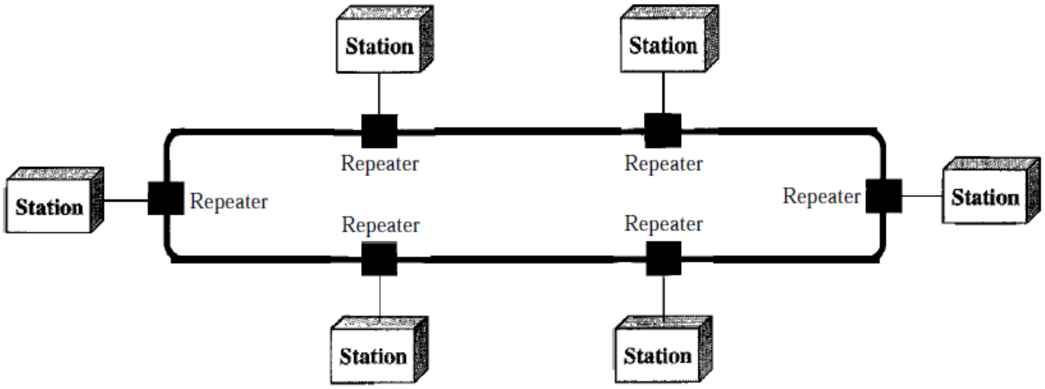
Advantages of a bus topology include ease of installation. Backbone cable can be laid along the most efficient path, and then connected to the nodes by drop lines of various lengths. In this way, a bus uses less cabling than mesh or star topologies.

Disadvantages include difficult reconnection and fault isolation. A bus is usually designed to be optimally efficient at installation. It can therefore be difficult to add new devices. Signal reflection at the taps can cause degradation in quality. This degradation can be controlled by limiting the number and spacing of devices connected to a given length of cable. Adding new devices may therefore require modification or replacement of the backbone. In addition, a fault or break in the bus cable stops all transmission, even between devices on the same side of the problem. The damaged area reflects signals back in the direction of origin, creating noise in both directions. Bus topology was the one of the first topologies used in the design of early local area networks.

Thus, **Bus topology** uses internal or external passive interface , coaxial cables with terminators at the end to remove signals, broadband two dimension transmission , and multipoint connection.

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.



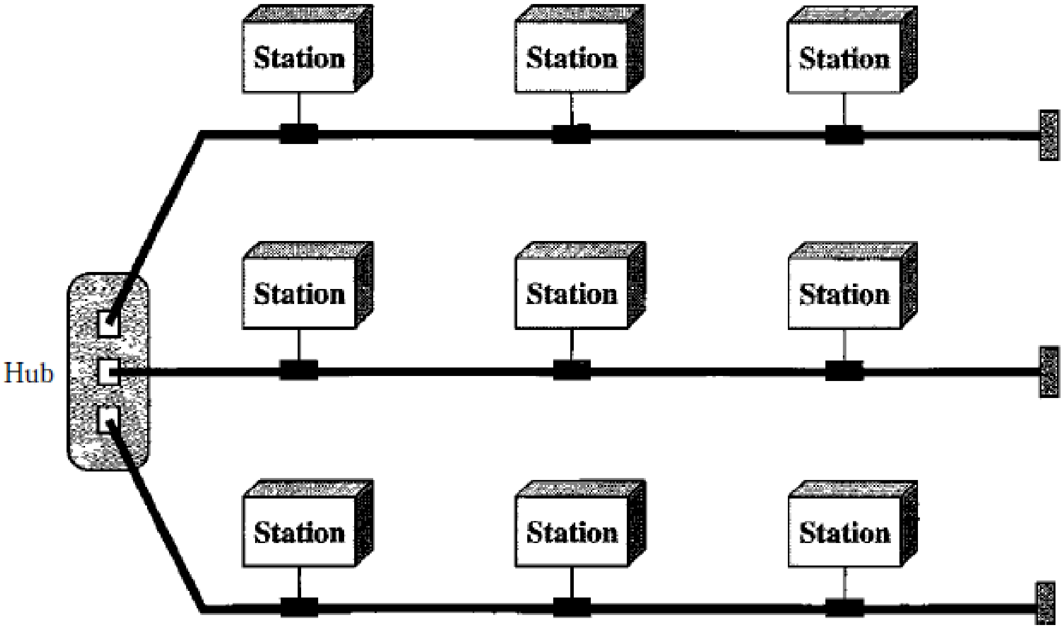
Each device is linked to only its immediate neighbors (either physically or logically). To add or delete a device requires changing only two connections. The only <u>constraints</u> are media and traffic considerations (maximum ring length and number of devices). In addition, fault isolation is simplified.

Generally, in a ring, a signal is circulating at all times. If one device does not receive a signal within a specified period, it can issue an alarm. However, <u>unidirectional</u> traffic can be a disadvantage. In a simple ring, a break in the ring (such as a disabled station) can disable the entire network. This weakness can be solved by using a <u>dual ring</u> 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.

<u>In general</u>, Ring topology uses guided media with active interface to transmit baseband signals in unidirectional links. Two states of links are operational (listen or transmit) and bypass. Two strategies to remove frames: <u>remove by source</u> and <u>remove by destination</u> (not support multicast or broadcast).

Hybrid Topology

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



Network Models

Computer networks are created by different entities. Standards are needed so that these heterogeneous networks can communicate with one another. The two best-known standards are the OSI model and the Internet model. The OSI (Open Systems Interconnection) model defines a seven-layer network; the Internet model defines a five-layer network.

Categories of Networks

Today when we speak of networks, we are generally referring to two primary categories: local-area networks and wide-area networks. The category into which a network falls is determined by its size. A LAN normally covers limited area; a WAN can be worldwide. Networks of a size in between are normally referred to as metropolitan area networks and span tens of miles.

Local Area Network

A local area network (LAN) is usually privately owned and links the devices in a single office, building, or campus. Depending on the needs of an organization and the type of technology used, 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 peripherals. Currently, LAN size is limited to a few kilometers. LANs are designed to allow resources to be shared between personal computers or workstations. The resources to be shared can include *hardware* (e.g., a printer), *software* (e.g., an application program), or *data*. In addition to size, LANs are distinguished from other types of networks by their transmission media and topology. In general, a given LAN uses only one type of transmission medium. The most common LAN topologies are bus, ring, and star. Early LANs had data rates in the 4 to 16 megabits per second (Mbps) range. Today, however, speeds are normally 100 or 1000 Mbps. Wireless LANs are the newest evolution in LAN technology.

Wide Area Network

A wide area network (WAN) provides long-distance transmission of data, image, audio, and video information over large geographic areas that may comprise a country, a continent, or even the whole world. A WAN can be as complex as the backbones that connect the Internet or as simple as a dial-up line that connects a home computer to the Internet. We normally refer to the first as a switched WAN and to the second as a point-to-point WAN.

The switched WAN connects the end systems, which usually comprise a router (internetworking connecting device) that connects to another LAN or WAN. The point-to-point WAN is normally a line leased from a telephone or cable TV provider that connects a home computer or a small LAN to an Internet service provider (lSP). This type of WAN is often used to provide Internet access.

Metropolitan Area Networks

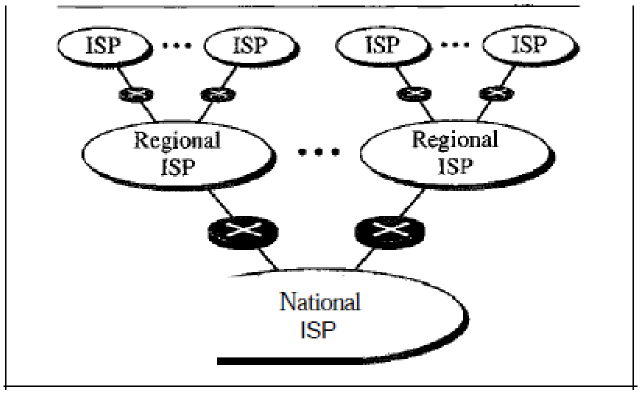
A metropolitan area network (MAN) is a network with a size between a LAN and a WAN. It normally covers the area inside a town or a city. It is designed for customers who need a high-speed connectivity, normally to the Internet, and have endpoints spread over a city or part of city. A good example of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

Interconnection of Networks: Internetwork

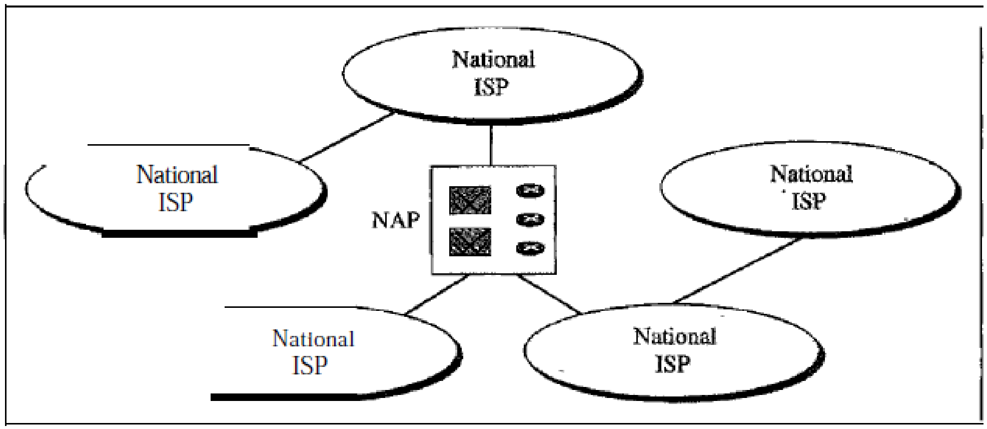
Today, it is very rare to see a LAN, a MAN, or a WAN in isolation; they are connected to one another. When two or more networks are connected, they become an internetwork, or internet. As an example, assume that an organization has two offices, one on the east coast and the other on the west coast. The established office on the west coast has a bus topology LAN; the newly opened office on the east coast has a star topology LAN. The president of the company lives somewhere in the middle and needs to have control over the company from her home. To create a backbone WAN for connecting these three entities (two LANs and the president's computer), a switched WAN (operated by a service provider such as a telecom company) has been leased. To connect the LANs to this switched WAN, however, three point-to-point WANs are required. These point-to-point WANs can be a high-speed DSL line offered by a telephone company or a cable modern line offered by a cable TV provider

THE INTERNET

The Internet has revolutionized many aspects of our daily lives. It has affected the way we do business as well as the way we spend our leisure time The Internet is a communication system that has brought a wealth of information to our fingertips and organized it for our use. The Internet today is run by private companies, not the government. Figure below shows a conceptual (not geographic) view of the Internet.



a. Structure of a national ISP



b. Interconnection of national ISPs

International Internet Service Providers

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

National Internet Service Providers

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

Regional Internet Service Providers

Regional internet service providers or regional ISPs are smaller ISPs that are connected to one or more national ISPs. They are at the third level of the hierarchy with a smaller data rate.

Local Internet Service Providers

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

LAN Network Models:

LAN can be configured as either client server or peer to peer model.

Client server Model

The **client/server model** is a computing model that acts as a distributed application which partitions tasks or workloads between the providers of a resource or service, called servers, and service requesters, called clients. Often clients and servers communicate over a computer network on separate hardware, but both client and server may reside in the same system. A server machine is a host that is running one or more server programs which share their resources with clients. A client does not share any of its resources, but requests a server's content or service function. Clients therefore initiate communication sessions with servers which await incoming requests.

Dedicated Architecture can improve the efficiency of a client server systems by using one server for each application that exists within an organization. While this can be somewhat cost prohibitive depending on the size of the organization, it will help in maintaining and troubleshooting some of the issues that can arise when using any technology.

A **peer-to-peer** (abbreviated to **P2P**) computer network is one in which each computer in the network can act as a client or server for the other computers in the network, allowing shared access to various resources such as files, peripherals, and sensors without the need for a central server. P2P networks can be set up within the home, a business, or over the Internet. Each network type requires all computers in the network to use the same or a compatible program to connect to each other and access files and other resources found on the other computer. P2P networks can be used for sharing content such as audio, video, data, or anything in digital format.

P2P is a distributed application architecture that partitions tasks or workloads among peers. Peers are equally privileged participants in the application. Each computer in the network is referred to as a node. The owner of each computer on a P2P network would set aside a portion of its resources - such as processing power, disk storage, or network bandwidth - to be made directly available to other network participant, without the need for central coordination by servers or stable hosts. With this model, peers are both suppliers and consumers of resources, in contrast to the traditional client–server model where only the server supply (send), and clients consume (receive). Emerging collaborative P2P systems are going beyond the era of peers doing similar things while sharing resources, and are looking for diverse peers that can bring in unique resources and capabilities to a virtual community thereby empowering it to engage in greater tasks beyond that can be accomplished by individual peers, yet are beneficial to all the peers.

Network Models

A network is a combination of hardware and software that sends data from one location to another. The hardware consists of the physical equipment that carries signals from one point of the network to another.

Data Communication Model is complex due to:

i- Network need to communicate end systems (ex: computers) and intermediate systems (ex: Routers) manufactured by a variety of venders.

ii- Different application program need to communicate with one another.

iii- An end system cannot be responsible for carrying information process as whole.

LAYERED TASKS

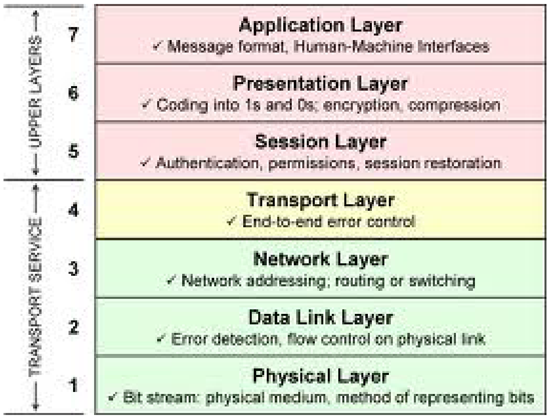
Layered architecture is used to handle computer network complexity such as complex task is broken into smaller subtasks and each subtask is assigned to layer.

THE OSI MODEL

Established in 1947, the International Standards Organization (ISO) is a multinational body dedicated to worldwide agreement on international standards. An ISO standard that covers all aspects of network communications is the Open Systems Interconnection 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 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. An understanding of the fundamentals of the OSI model provides a solid basis for exploring data communications.



7	**Application Layer** ✓ Message format, Human-Machine Interfaces
6	**Presentation Layer** ✓ Coding into 1s and 0s; encryption, compression
5	**Session Layer** ✓ Authentication, permissions, session restoration
4	**Transport Layer** ✓ End-to-end error control
3	**Network Layer** ✓ Network addressing; routing or switching
2	**Data Link Layer** ✓ Error detection, flow control on physical link
1	**Physical Layer** ✓ Bit stream: physical medium, method of representing bits

OSI Layered Architecture

The OSI model is composed of seven ordered layers: physical (layer 1), data link (layer 2), network (layer 3), transport (layer 4), session (layer 5), presentation (layer 6), and application (layer 7). In *developing* the model, the designers distilled the process of transmitting data to its most fundamental elements. They identified which networking functions had related uses and collected those functions into discrete groups that became the layers. Each layer defines a family of functions distinct from those of the other layers. By defining and localizing functionality in this fashion, the designers created an architecture that is both comprehensive and flexible. Most importantly, the OSI model allows complete interoperability between otherwise incompatible systems. Within a single machine, each layer calls upon the services of the layer just below it. Layer 3, for example, uses the services provided by layer 2 and provides services for layer 4. Between machines, layer *x* on one machine communicates with layer *x* on another machine. This communication is governed by an agreed-upon series of rules and conventions called protocols. The processes on each machine that communicate at a given layer are called peer-to-peer processes. Communication between machines is therefore a peer-to-peer process using the protocols appropriate to a given layer.

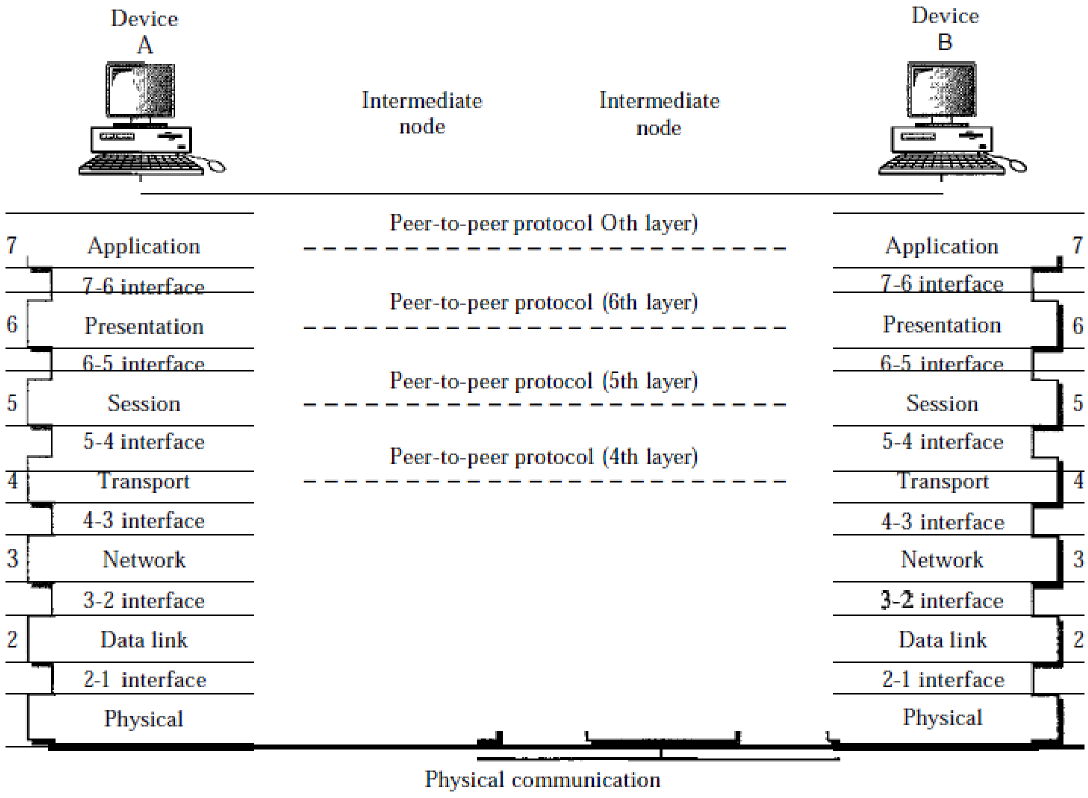
Peer-to-Peer Processes

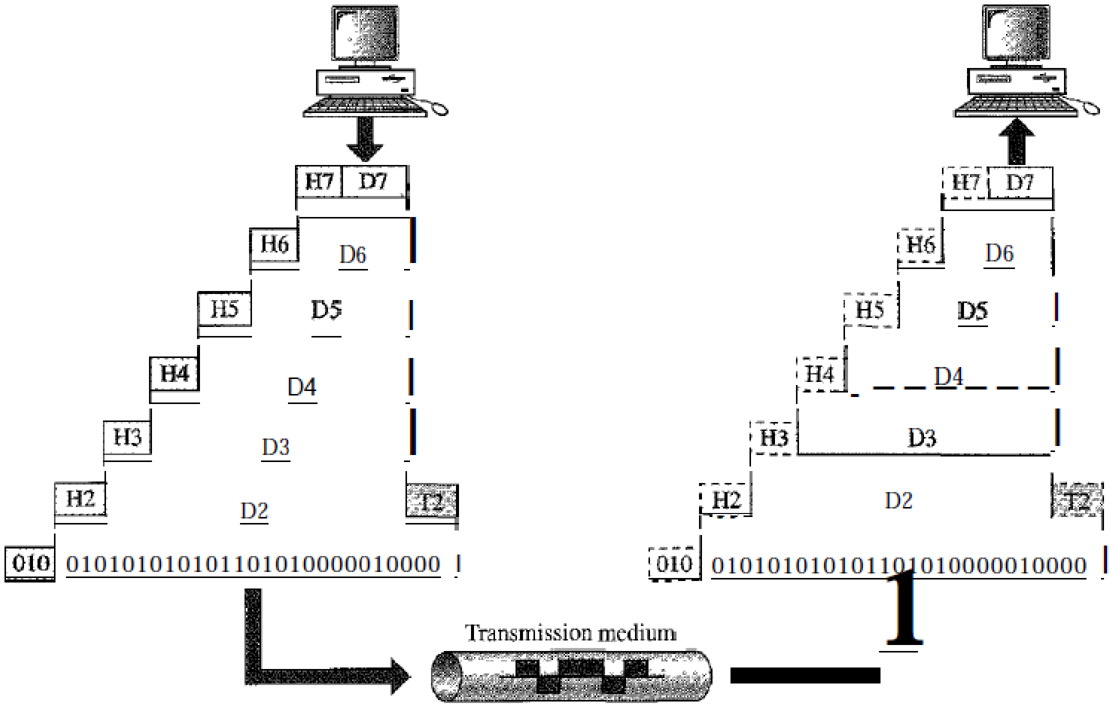
At the physical layer, communication is direct: device A sends a stream of bits to device B (through intermediate nodes). At the higher layers, however, communication must move down through the layers on device A, over to device B, and then back up through the layers. Each layer in the sending device adds its own information to the message it receives from the layer just above it and passes the whole package to the layer just below it.

At layer 1 the entire package is converted to a form that can be transmitted to the receiving device. At the receiving machine, the message is unwrapped layer by layer, with each process receiving and removing the data meant for it. For example, layer 2 removes the data meant for it, then passes the rest to layer 3. Layer 3 then removes the data meant for it and passes the rest to layer 4, and so on.

Interfaces between Layers

The passing of the data and network information down through the layers of the sending device and back up through the layers of the receiving device is made possible by an interface between each pair of adjacent layers. Each interface defines the information and services a layer must provide for the layer above it. Well-defined interfaces and layer functions provide modularity to a network. As long as a layer provides the expected services to the layer above it, the specific implementation of its functions can be modified or replaced without requiring changes to the surrounding layers.


Device
A
Intermediate
node
Intermediate
node
Device
B
Peer-to-peer protocol Oth layer)
Peer-to-peer protocol (6th layer)
Peer-to-peer protocol (5th layer)
Peer-to-peer protocol (4th layer)
7 Application
7-6 interface
6 Presentation
6-5 interface
5 Session
5-4 interface
4 Transport
4-3 interface
3 Network
3-2 interface
2 Data link
2-1 interface
Physical
Application 7
7-6 interface
Presentation 6
6-5 interface
Session 5
5-4 interface
Transport 4
4-3 interface
Network 3
3-2 interface
Data link 2
2-1 interface
Physical
Physical communication


H7
D7
H6
D6
H5
D5
H4
D4
H3
D3
H2
D2
T2
010
0101010101011010100000010000
Transmission medium
H7
D7
H6
D6
H5
D5
H4
D4
H3
D3
H2
D2
T2
010
0101010101011010100000010000

Organization of the Layers

The seven layers can be thought of as belonging to three subgroups. Layers 1, 2, and 3-physical, data link, and network-are the network support layers; they deal with the physical aspects of moving data from one device to another (such as electrical specifications, physical connections, physical addressing, and transport timing and reliability). Layers 5, 6, and 7-session, presentation, and application-can be thought of as the user support layers; they allow interoperability among unrelated software systems. Layer 4, the transport layer, links the two subgroups and ensures that what the lower layers have transmitted is in a form that the upper layers can use. implementation of its functions can be modified or replaced without requiring changes to the surrounding layers.

The upper OSI layers are almost always implemented in software; lower layers are a combination of hardware and software, except for the physical layer, which is mostly hardware

At each layer, a **header,** or possibly a **trailer,** can be added to the data unit. Commonly, the trailer is added only at layer 2. When the formatted data unit passes through the physical layer (layer 1), it is changed into an electromagnetic signal and transported along a physical link. Upon reaching its destination, the signal passes into layer 1 and is transformed back into digital form. The data units then move back up through the OSI layers. As each block of data reaches the next higher layer, the headers and trailers attached to it at the corresponding sending layer are removed, and actions appropriate to that layer are taken. By the time it reaches layer 7, the message is again in a form appropriate to the application and is made available to the recipient.

Encapsulation

A packet (header and data) at level 7 is encapsulated in a packet at level 6. The whole packet at level 6 is encapsulated in a packet at level 5, and so on.

In other words, the data portion of a packet at level *N* - 1 carries the whole packet (data and header and maybe trailer) from level *N*. The concept is called *encapsulation;* level *N* - 1 is not aware of which part of the encapsulated packet is data and which part is the header or trailer. For level *N* - 1, the whole packet coming from level *N* is treated as one integral unit.

LAYERS IN THE OSI MODEL

In this section we briefly describe the functions of each layer in the OSI model.

Physical Layer

The physical layer coordinates the functions required to carry a bit stream over a physical medium. It deals with the mechanical and electrical specifications of the interface and transmission medium. It also defines the procedures and functions that physical devices and interfaces have to perform for transmission to Occur. The physical layer is also concerned with the following:

# ***Physical characteristics of interfaces and medium***. The physical layer defines the characteristics of the interface between the devices and the transmission medium. It also defines the type of transmission medium.

# ***Representation of bits.*** The physical layer data consists of a stream of bits (sequence of Os or 1s) with no interpretation. To be transmitted, bits must be encoded into signals--electrical or optical. The physical layer defines the type of encoding (how Os and I s are changed to signals).

# ***Data rate***. The transmission rate-the number of bits sent each second-is also defined by the physical layer. In other words, the physical layer defines the duration of a bit, which is how long it lasts.

# ***Synchronization of bits***. The sender and receiver not only must use the same bit rate but also must be synchronized at the bit level. In other words, the sender and the receiver clocks must be synchronized.

# ***Line configuration***. The physical layer is concerned with the connection of devices to the media. In a point-to-point configuration, two devices are connected

through a dedicated link. In a multipoint configuration, a link is shared among several devices.

# ***Physical topology***. The physical topology defines how devices are connected to make a network. Devices can be connected by using a mesh topology (every device is connected to every other device), a star topology (devices are connected through a central device), a ring topology (each device is connected to the next, forming a ring), a bus topology (every device is on a common link), or a hybrid topology (this is a combination of two or more topologies).

#**Transmission mode**. The physical layer also defines the direction of transmission between two devices: simplex, half-duplex, or full-duplex. In simplex mode, only one device can send; the other can only receive. The simplex mode is a one-way communication. In the half-duplex mode, two devices can send and receive, but not at the same time. In a full-duplex (or simply duplex) mode, two devices can send and receive at the same time.

Data Link Layer

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

Other responsibilities of the data link layer include the following:

# Framing: The data link layer divides the stream of bits received from the network layer into manageable data units called frames.

#***addressing***: If frames are to be distributed to different systems on the network, the data link layer adds a header to the frame to define the sender and/or receiver of the frame. If the frame is intended for a system outside the sender's network, the receiver address is the address of the device that connects the network to the next one.

# ***Flow control***. If the rate at which the data are absorbed by the receiver is less than the rate at which data are produced in the sender, the data link layer imposes a flow control mechanism to avoid overwhelming the receiver.

# ***Error control***. The data link layer adds reliability to the physical layer by adding mechanisms to detect and retransmit damaged or lost frames. It also uses a mechanism to recognize duplicate frames. Error control is normally achieved through a trailer added to the end of the frame.

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

Network Layer

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

If two systems are connected to the same link, there is usually no need for a network layer. However, if the two systems are attached to different networks (links) with connecting devices between the networks (links), there is often a need for the network layer to accomplish source-to-destination delivery.

Other responsibilities of the network layer include the following:

# ***Creating Logical end to end connection***. The end systems should see logical connection between them without worrying about links and connecting devices.

# ***Hiding the details of the lower layer***. Hiding the details of data link and physical layers from upper layers.

# ***Logical addressing***. The physical addressing implemented by the data link layer handles the addressing problem locally. If a packet passes the network boundary, we need another addressing system to help distinguish the source and destination

systems. The network layer adds a header to the packet coming from the upper layer that, among other things, includes the logical addresses of the sender and receiver. We discuss logical addresses later in this chapter.

# ***Routing***. When independent networks or links are connected to create *intemetworks* (network of networks) or a large network, the connecting devices (called *routers* or *switches)* route or switch the packets to their final destination. One of the functions of the network layer is to provide this mechanism.

As the figure shows, now we need a source-to-destination delivery. The network layer at A sends the packet to the network layer at B. When the packet arrives at router B, the router makes a decision based on the final destination (F) of the packet. As we will see in later chapters, router B uses its routing table to find that the next hop is router E. The network layer at B, therefore, sends the packet to the network layer at E. The network layer at E, in term, sends the packet to the network layer at F.

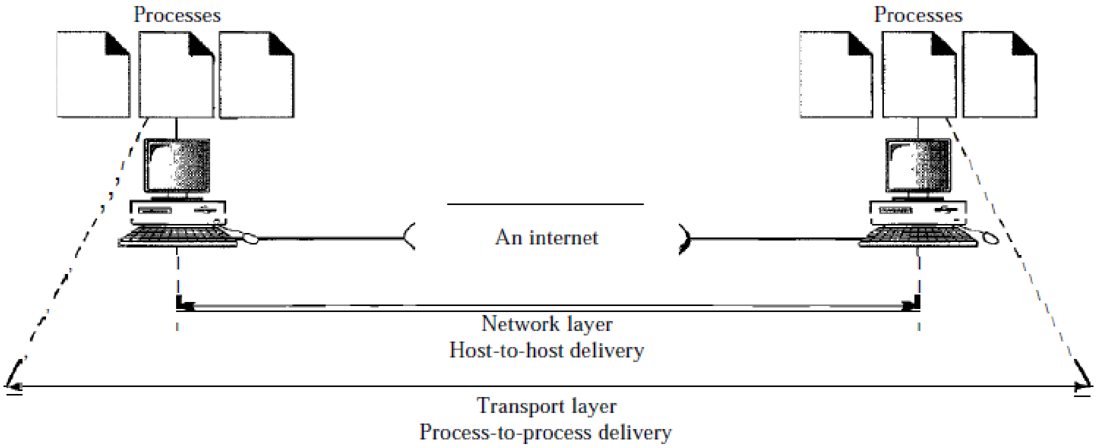
Transport Layer

The transport layer is responsible for process-to-process delivery of the entire message. A process is an application program running on a host. Whereas the network layer oversees source-to-destination delivery of individual packets, it does not recognize any relationship between those packets. It treats each one independently, as though each piece belonged to a separate message, whether or not it does. The transport layer, on the other hand, ensures that the whole message arrives intact and in order, overseeing both error control and flow control at the source-to-destination level.

Other responsibilities of the transport layer include the following:

# Service-point addressing. Computers often run several programs at the same time. For this reason, source-to-destination delivery means delivery not only from one computer to the next but also from a specific process (running program) on one computer to a specific process (running program) on the other. The transport layer header must therefore include a type of address called a *service-point address* (or port address). The network layer gets each packet to the correct computer; the transport layer gets the entire message to the correct process on that computer.

# ***Segmentation and reassembly***. A message is divided into transmittable segments, with each segment containing a sequence number. These numbers enable the transport layer to reassemble the message correctly upon arriving at the destination and to identify and replace packets that were lost in transmission.



# ***Connection control.*** The transport layer can be either connectionless or connectionoriented.

A connectionless transport layer treats each segment as an independent packet and delivers it to the transport layer at the destination machine. A connectionoriented transport layer makes a connection with the transport layer at the destination machine first before delivering the packets. After all the data are transferred, the connection is terminated.

# ***Flow control.*** Like the data link layer, the transport layer is responsible for flow control. However, flow control at this layer is performed end to end rather than across a single link.

# ***Error control***. Like the data link layer, the transport layer is responsible for error control. However, error control at this layer is performed process-toprocess rather than across a single link. The sending transport layer makes sure that the entire

message arrives at the receiving transport layer without error (damage, loss, or duplication). Error correction is usually achieved through retransmission.

Session Layer

The services provided by the first three layers (physical, data link, and network) are not sufficient for some processes. The session layer is the network *dialog controller.* It establishes, maintains, and synchronizes the interaction among communicating systems.

Specific responsibilities of the session layer include the following:

# **Dialog control.** The session layer allows two systems to enter into a dialog. It allows the communication between two processes to take place in either halfduplex (one way at a time) or full-duplex (two ways at a time) mode.

# **Synchronization**. The session layer allows a process to add checkpoints, or synchronization points, to a stream of data. For example, if a system is sending a file of 2000 pages, it is advisable to insert checkpoints after every 100 pages to ensure that each 100-page unit is received and acknowledged independently. In this case, if a crash happens during the transmission of page 523, the only pages that need to be resent after system recovery are pages 501 to 523. Pages previous to 501 need not be resent.

# **Atomization.** Some times an application need several transport connections. Session layer can combine these several connections into one task.

Presentation Layer

The presentation layer is concerned with the syntax and semantics of the information exchanged between two systems. Figure 2.13 shows the relationship between the presentation layer and the application and session layers.

Specific responsibilities of the presentation layer include the following:

# ***Translation***. The processes (running programs) in two systems are usually exchanging information in the form of character strings, numbers, and so on. The infonnation must be changed to bit streams before being transmitted. Because different computers use different encoding systems, the presentation layer is responsible for interoperability between these different encoding methods. The presentation layer at the sender changes the information from its sender-dependent format into a common format. The presentation layer at the receiving machine changes the common format into its receiver-dependent format.

# ***Encryption***. To carry sensitive information, a system must be able to ensure privacy. Encryption means that the sender transforms the original information to another form and sends the resulting message out over the network. Decryption reverses the original process to transform the message back to its original form.

# ***Compression***. Data compression reduces the number of bits contained in the information. Data compression becomes particularly important in the transmission of multimedia such as text, audio, and video.

Application Layer

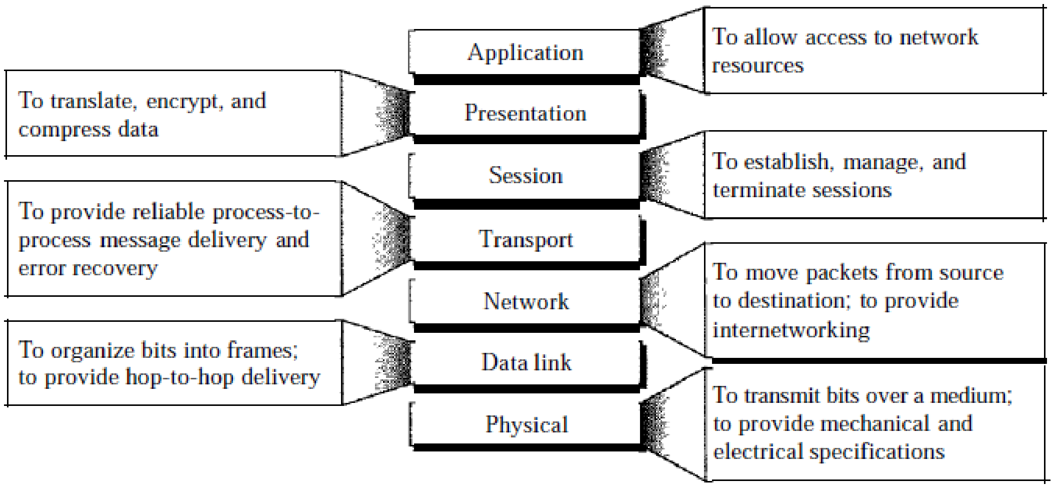
The application layer enables the user, whether human or software, to access the network. It provides user interfaces and support for services such as electronic mail, remote file access and transfer, shared database management, and other types of distributed information services. Specific services provided by the application layer include the following:

# **Network virtual terminal**. A network virtual terminal is a software version of a physical terminal, and it allows a user to log on to a remote host. To do so, the application creates a software emulation of a terminal at the remote host. The user's computer talks to the software terminal which, in turn, talks to the host, and vice versa. The remote host believes it is communicating with one of its own terminals and allows the user to log on.

# File access, transfer,and management. This application allows a user to access files in a remote host (to make changes or read data), to retrieve files from a remote computer for use in the local computer, and to manage or control files in a remote computer locally.

# **Mail services.** This application provides the basis for e-mail forwarding and storage.

# **Directory services**. This application provides distributed database sources and access for global information about various objects and services.


Application
To allow access to network resources
Presentation
To translate, encrypt, and compress data
Session
To establish, manage, and terminate sessions
Transport
To provide reliable process-to-process message delivery and error recovery
Network
To move packets from source to destination; to provide internetworking
Data link
To organize bits into frames; to provide hop-to-hop delivery
Physical
To transmit bits over a medium; to provide mechanical and electrical specifications

TCP/IP PROTOCOL SUITE

The TCP/IP protocol suite was developed prior to the OSI model. Therefore, the layers in the TCP/IP protocol suite do not exactly match those in the OSI model. The original TCP/IP protocol suite was defined as having four layers: host-to-network, internet, transport, and application. However, when TCP/IP is compared to OSI, we can say that the host-to-network layer is equivalent to the combination of the physical and data link layers. The internet layer is equivalent to the network layer, and the application layer is roughly doing the job of the session, presentation, and application layers with the transport layer in TCP/IP taking care of part of the duties of the session layer. So we assume that the TCP/IP protocol suite is made of five layers: physical, data link, network, transport, and application. The first four layers provide physical standards, network interfaces, internetworking, and transport functions that correspond to the first four layers of the OSI model.

TCP/IP is a hierarchical protocol made up of interactive modules, each of which provides a specific functionality; however, the modules are not necessarily interdependent.

Whereas the OSI model specifies which functions belong to each of its layers, the layers of the *TCP/IP* protocol suite contain relatively independent protocols that can be mixed and matched depending on the needs of the system. The term *hierarchical* means that each upper-level protocol is supported by one or more lower-level protocols.

At the transport layer, *TCP/IP* defines three protocols: Transmission Control Protocol (TCP), User Datagram Protocol (UDP), and Stream Control Transmission Protocol (SCTP). At the network layer, the main protocol defined by TCP/IP is the Internetworking Protocol (IP); there are also some other protocols that support data movement in this layer.

OSI #	**OSI Layer Name**	**TCP/IP #**	**TCP/IP Layer Name**	**Encapsulation Units**	**TCP/IP Protocols**
7	Application	4	Application	data	FTP, HTTP, POP3, IMAP, telnet, SMTP, DNS, TFTP
6	Presentation			data	
5	Session			data	
4	Transport	3	Transport	segments	TCP, UDP
3	Network	2	Internet	packets	IP
2	Data Link	1	Network Access	frames	
1	Physical			bits	

Physical and Data Link Layers

At the physical and data link layers, *TCP/IP* does not define any specific protocol. It supports all the standard and proprietary protocols. A network in a *TCP/IP* internetwork can be a local-area network or a wide-area network.

Network Layer

At the network layer (or, more accurately, the internetwork layer), *TCP/IP* supports the Internetworking Protocol. IP, in turn, uses four supporting protocols: ARP, RARP, ICMP, and IGMP.

Internetworking Protocol (IP)

The Internetworking Protocol (IP) is the transmission mechanism used by the TCP/IP protocols. It is an unreliable and connectionless protocol-a best-effort delivery service. The term *best effort* means that IP provides no error checking or tracking. IP assumes the unreliability of the underlying layers and does its best to get a transmission through to its destination, but with no guarantees.

IP transports data in packets called *datagrams*, each of which is transported separately. Datagrams can travel along different routes and can arrive out of sequence or be duplicated. IP does not keep track of the routes and has no facility for reordering datagrams once they arrive at their destination. The limited functionality of IP should not be considered a weakness, however. IP provides bare-bones transmission functions that free the user to add only those facilities necessary for a given application and thereby allows for maximum efficiency.

Address Resolution Protocol

The Address Resolution Protocol (ARP) is used to associate a logical address with a physical address. On a typical physical network, such as a LAN, each device on a link is identified by a physical or station address, usually imprinted on the network interface card (NIC). ARP is used to find the physical address of the node when its Internet address is known.

Reverse Address Resolution Protocol

The Reverse Address Resolution Protocol (RARP) allows a host to discover its Internet address when it knows only its physical address. It is used when a computer is connected to a network for the first time or when a diskless computer is booted.

Internet Control Message Protocol

The Internet Control Message Protocol (ICMP) is a mechanism used by hosts and gateways to send notification of datagram problems back to the sender. ICMP sends query and error reporting messages.

Internet Group Message Protocol

The Internet Group Message Protocol (IGMP) is used to facilitate the simultaneous transmission of a message to a group of recipients.

Transport Layer

Traditionally the transport layer was represented in *TCP/IP* by two protocols: TCP and UDP. IP is a host-to-host protocol, meaning that it can deliver a packet from one physical device to another. UDP and TCP are transport level protocols responsible for delivery of a message from a process (running program) to another process. A new transport layer protocol, SCTP, has been devised to meet the needs of some newer applications.

User Datagram Protocol

The User Datagram Protocol (UDP) is the simpler of the two standard TCPIIP transport protocols. It is a process-to-process protocol that adds only port addresses, checksum error control, and length information to the data from the upper layer.

Transmission Control Protocol

The Transmission Control Protocol (TCP) provides full transport-layer services to applications. TCP is a reliable stream transport protocol. The term *stream,* in this context, means connection-oriented: A connection must be established between both ends of a transmission before either can transmit data.

At the sending end of each transmission, TCP divides a stream of data into smaller units called *segments*. Each segment includes a sequence number for reordering after receipt, together with an acknowledgment number for the segments received. Segments are carried across the internet inside of IP datagrams. At the receiving end, TCP collects each datagram as it comes in and reorders the transmission based on sequence numbers.

Stream Control Transmission Protocol

The Stream Control Transmission Protocol (SCTP) provides support for newer applications such as voice over the Internet. It is a transport layer protocol that combines the best features of UDP and TCP.

Application Layer

The *application layer* in TCP/IP is equivalent to the combined session, presentation, and application layers in the OSI model. Many protocols are defined at this layer.