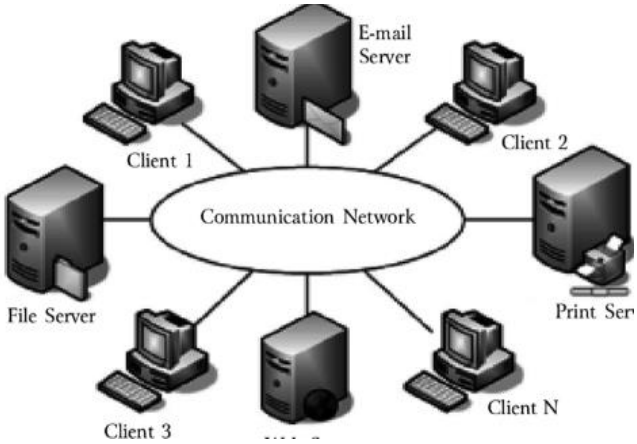
Unit-1

What is a Computer Network?

Computer Network is a group of computers connected with each other through wires, optical fibers so that various devices can interact with each other through a network. The aim of the computer network is the sharing of resources among various devices.



Features of Computer network

Communication speed

Network provides us to communicate over the network in a fast and efficient manner. For example, we can do video conferencing, email messaging, etc.

File sharing

File sharing is one of the major advantage of the computer network.

Security

Network allows the security by ensuring that the user has the right to access the certain files and applications.

Reliability

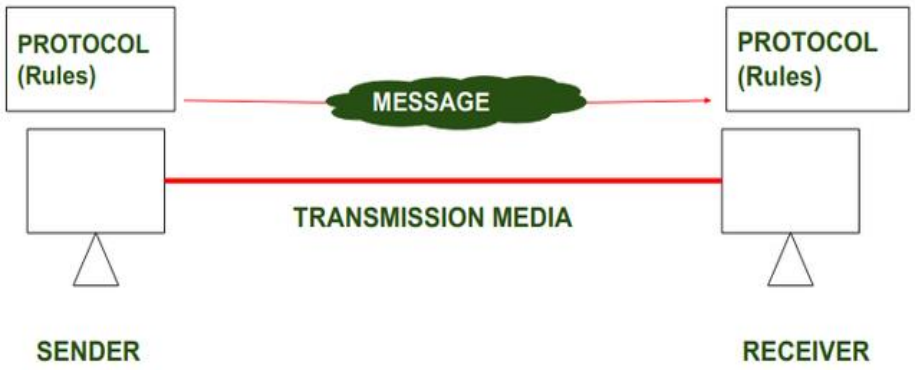
Computer network can use the alternative source for the data communication in case of any hardware failure.

Scalability

The measure of a system's ability to increase or decrease in performance and cost in response to changes in application and system processing demands

Protocol

- A set of rules governing data communication is called protocol. These rules help in data communication, with connected devices and govern all aspects of information communication.
- Ex: A product works and is widely used regardless of individual manufacturer. Standards provide guidelines for vendors, and manufactures, for designing a product. This leads to open and competitive market for manufacturers.



PROTOCOL LAYERING

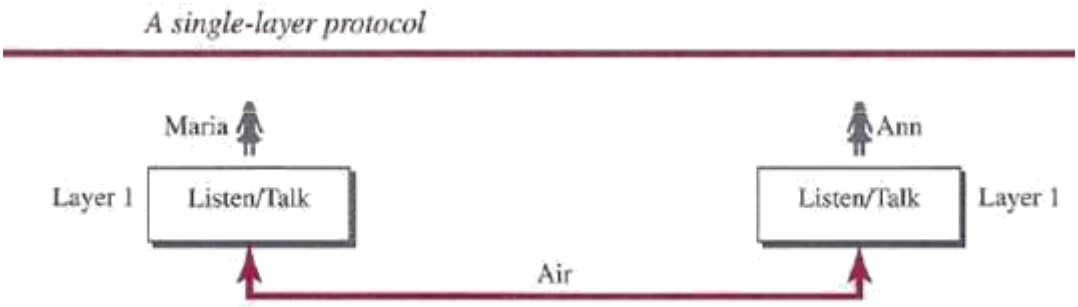
We defined the term *protocol* in Chapter 1. In data communication and networking, a protocol defines the rules that both the sender and receiver and all intermediate devices need to follow to be able to communicate effectively. When communication is simple, we may need only one simple protocol; when 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.

Scenarios

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

First Scenario

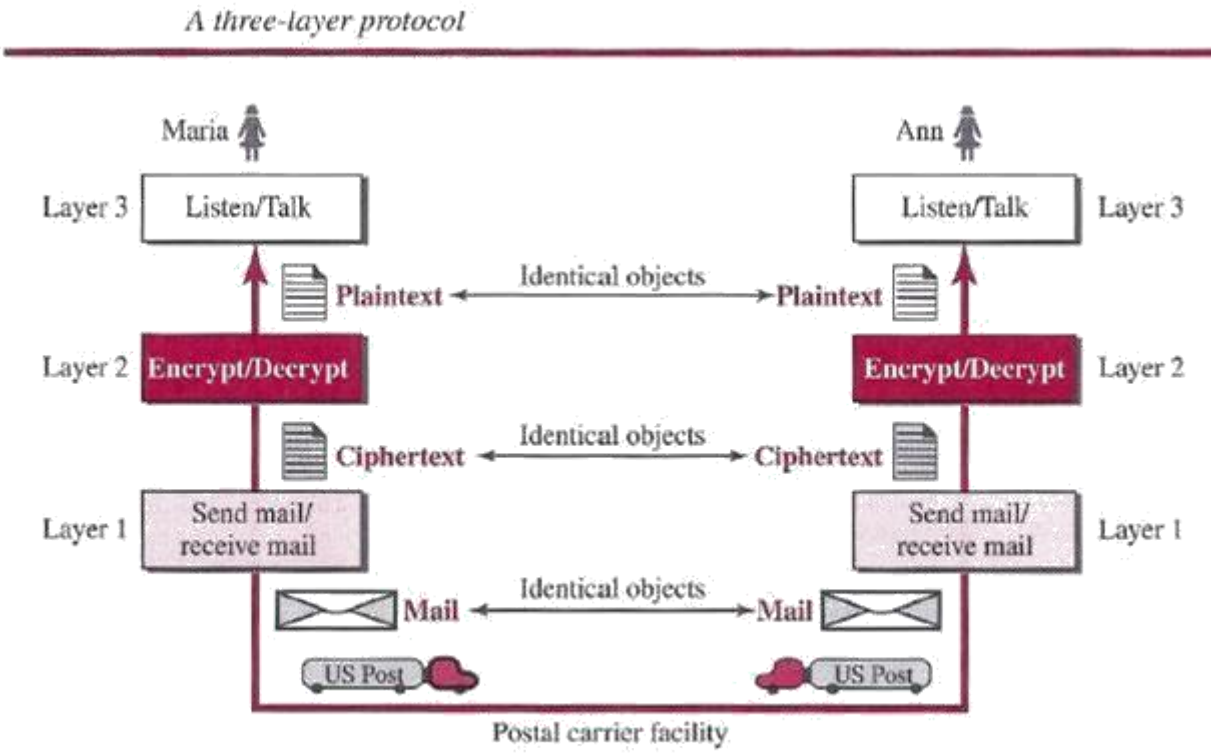
In the first scenario, communication is so simple that it can occur in only one layer. Assume Maria and Ann are neighbors with a lot of common ideas. Communication between Maria and Ann takes place in one layer, face to face, in the same language, as shown in Figure.



Even in this simple scenario, we can see that a set of rules needs to be followed. First, Maria and Ann know that they should greet each other when they meet. Second, they know that they should confine their vocabulary to the level of their friendship. Third, each party knows that she should refrain from speaking when the other party is speaking. Fourth, each party knows that the conversation should be a dialog, not a monolog: both should have the opportunity to talk about the issue. Fifth, they should exchange some nice words when they leave. We can see that the protocol used by Maria and Ann is different from the communication between a professor and the students in a lecture hall. The communication in the second case is mostly monolog; the professor talks most of the time unless a student has a question, a situation in which the protocol dictates that she should raise her hand and wait for permission to speak. In this case, the communication is normally very formal and limited to the subject being taught.

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



Principles of Protocol Layering

Let us discuss two 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

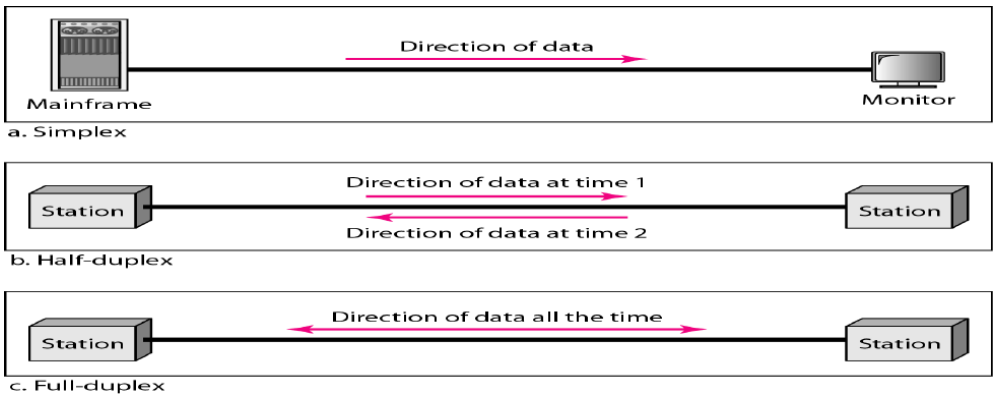
The second principle that we need to follow in protocol layering is that the two objects under each layer at both sites should be identical. For example, the object under layer 3 at both sites should be a plaintext letter. both sites should be a cipher text letter. The object under layer 1 at both sites 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 below 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.

Data Flow

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



Simplex In simplex mode, the communication is unidirectional, as on a one-way s-treet. Only one of the two devices on a link can transmit; the other canonly receive (Figure a). Keyboards and traditional monitors are examples of simplex devices.

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

Full-Duplex

In full-duplex, both stations can transmit and receive simultaneously (Figure c). One common example of full-duplex communication is the telephone network.When two people are communicating by a telephone line, both can talk andlisten at the same time. The full-duplex mode is used when communication inboth directions is required all the time.

Components in Computer Networks

1. Network Interface Cards

NIC represents Network Interface Cards. A personal computer development board linked to a PC or server and works with the network control structure to force the network's data stream. There are many network documentations that call NIC a network board. In this manner, all NICs are connected to a network

2. Hubs

A Hub is a hardware device that divides the network connection among multiple devices. When computer requests for some information from a network, it first sends the request to the Hub through cable. Hub will broadcast this request to the entire network.

3. Switches

Switch is a network device that connects other devices to Ethernet networks through twisted pair cables. It uses packet switching technique to receive, store and forward data packets on the network. The switch maintains a list of network addresses of all the devices connected to it

4. Router

The router is a physical internetworking device that is designed to receive, analyze, and forward data packets between computer networks.

5. Modem

A modem is a hardware device that allows the computer to connect to the internet over the existing telephone line. A modem is not integrated with the motherboard rather than it is installed on the PCI slot found on the motherboard.

Network Topology

It refers to the physical arrangement and representation of all the nodes and components of the network. In general terms, Topology defines the structure of the entire network. The network topology is divided into four types.

- 1. Bus Topology
- In this arrangement, the nodes (computers) are connected through interface connectors to a single communication line (central cable) that carries the message in both the directions. The central cable to which all the nodes are connected is the backbone of the network. It is called a bus.
- The signal in this arrangement travels in both directions to all the machines until it finds the recipient machine.
- 2. Ring Topology
- As the name suggests, in a ring topology, the computers are connected in a circular and closed loop. The message in this topology moves only in one direction around the ring from one node to another node and is checked by each node for a matching destination address. So, the data keeps moving until it reaches its destination.
- All nodes are equal; a client-server relationship does not exist between them.
- 3. Star Topology
- Star topology is a network layout in which all devices are connected to a central hub or switch. This central hub acts as a central point of communication and controls the flow of data between devices. This topology is often used in small to medium-sized networks, such as home networks or small office networks
- 4. Mesh Topology
- In a mesh topology, every device is connected to another device in a network using a point-to-point connection. The connection is generally known as a dedicated connection, as the link transports data between two devices. The number of links in a mesh topology is calculated using the formula below.
- Military organizations use mesh topology to avoid breaks down in communications.
- Many connections = n * (n - 1) /2. Here, "n" represents the number of nodes in a network.

Computer Network Types

1. Local Area Network

A computer network spanned inside a building and operated under single administrative system is generally termed as Local Area Network (LAN). Usually,LAN covers an organization' offices, schools, colleges or universities LAN provides a useful way of sharing the resources between end users. The resources such as printers, file servers, scanners, and internet are easily sharable among computers.

2. Personal Area Network

A Personal Area Network (PAN) is smallest network which is very personal to a user. This may include Bluetooth enabled devices or infra-red enabled devices. PAN has connectivity range up to 10 meters. PAN may include wireless computer keyboard and mouse, Bluetooth

enabled headphones, wireless printers and TV remotes.

3. Metropolitan Area Network

Metropolitan Area Network (MAN) is an extensive network that connects numerous corporate LANs together. Their communication devices and equipment are maintained by a group or single network provider that sells its networking services to corporate customers. MANs often take the role of high-speed network that allows sharing of regional resources.

4. Wide Area Network

A WAN, also called the Wide Area Network, is defined as a telecommunications network that extends over a large area. Generally, telecommunication networks are Wide Area Network. These networks provide connectivity to MANs and LANs. Since they are equipped with very high speed backbone, WANs use very expensive network equipment

INTERNET HISTORY

Now that we have given an overview of the Internet, let us give a brief history of the internet. This brief history makes it clear how the Internet has evolved from a private network to a global one in less than 40 years.

Early History

There were some communication networks, such as telegraph and telephone networks, before 1960. These networks were suitable for constant-rate communication at that time, which means that after a connection was made between two users, the encoded message (telegraphy) or voice (telephony) could be exchanged.

ARPANET

In the mid-1960s, mainframe computers in research organizations were stand-alone devices. Computers from different manufacturers were unable to communicate with one another. The Advanced Research Projects Agency (ARPA) in the Department of Defense (DOD) was interested in finding a way to connect computers so that the researchers they funded could share their findings, thereby reducing costs and eliminating duplication of effort. In 1967, at an Association for Computing Machinery (ACM) meeting, ARPA presented its ideas for the Advanced Research Projects Agency Network (ARPANET), a small network of connected computers. The idea was that each host computer (not necessarily from the same manufacturer) would be attached to a specialized computer, called an *interface message processor* (IMP). The IMPs, in turn, would be connected to each other. Each IMP had to be able to communicate with other IMPs as well as with its own attached host.

Birth of the Internet

In 1972, Vint Cerf and Bob Kahn, both of whom were part of the core ARPANET group, collaborated on what they called the *Internetting Project. TCPI/P* Cerf and Kahn's landmark 1973 paper outlined the protocols to achieve end-to-end delivery of data. This was a new version of NCP. This paper on transmission control protocol (TCP) included concepts such as encapsulation, the datagram, and the functions of a gateway.

Transmission Control Protocol (TCP) and Internet Protocol (IP). IP would handle datagram routing while TCP would be responsible for higher level functions such as segmentation, reassembly, and error detection. The new combination became known as TCPIIP.

MILNET

In 1983, ARPANET split into two networks: Military Network (MILNET) for military users and ARPANET for nonmilitary users.

CSNET

Another milestone in Internet history was the creation of CSNET in 1981. Computer Science Network (CSNET) was a network sponsored by the National Science Foundation (NSF).

NSFNET

With the success of CSNET, the NSF in 1986 sponsored the National Science Foundation Network (NSFNET), a backbone that connected five supercomputer centers located throughout the United States.

ANSNET

In 1991, the U.S. government decided that NSFNET was not capable of supporting the rapidly increasing Internet traffic. Three companies, IBM, Merit, and Verizon, filled the void by forming a nonprofit organization called Advanced Network & Services (ANS) to build a new, high-speed Internet backbone called Advanced Network Services Network (ANSNET). Internet Today, we witness a rapid growth both in the infrastructure and new applications. The Internet today is a set of pier networks that provide services to the whole world. What has made the internet so popular is the invention of new applications.

World Wide Web

The 1990s saw the explosion of Internet applications due to the emergence of the World Wide Web (WWW). The Web was invented at CERN by Tim Berners-Lee. This invention has added the commercial applications to the Internet.

Draft Standard. A proposed standard is elevated to draft standard status after at least two successful independent and interoperable implementations. Barring difficulties, a draft standard, with modifications if specific problems are encountered, normally becomes an Internet standard.

Historic The historic RFCs are significant from a historical perspective. They either have been superseded by later specifications or have never passed the necessary maturity levels to become an Internet standard.

Experimental An RFC classified as experimental describes work related to an experimental situation that does not affect the operation of the Internet. Such an RFC should not be implemented in any functional Internet service.

RFCs are classified into five *requirement levels:* required, recommended, elective, limited use, and not recommended.

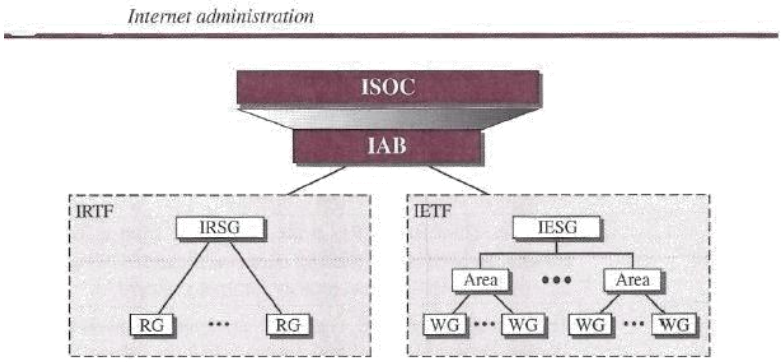
Required An RFC is labeled *required* if it must be implemented by all Internets systems to achieve minimum conformance. For example, IF and ICMP are required protocols.

Recommended An RFC labeled recommended is not required for minimum conformance; it is recommended because of its usefulness. For example, FTP and TELNET are recommended protocols.

Elective An RFC labeled elective is not required and not recommended. However, a system can use it for its own benefit.

Limited Use An RFC labeled limited use should be used only in limited situations. Most of the experimental RFCs fall under this category.

Not Recommended *An* RFC labeled not recommended is inappropriate for general use. Normally a historic (deprecated) RFC may fall under this category.



INTERNET ADMINISTRATION

The Internet, with its roots primarily in the research domain, has evolved and gained a broader user base with significant commercial activity. Various groups that coordinate Internet issues have guided this growth and development. Appendix G gives the addresses, e-rnail addresses, and telephone numbers for some of these groups. Shows the general organization of Internet administration. E-rnail addresses and telephone numbers for some of these groups. Below figure shows the general organization of Internet administration.

Isoc

The Internet Society (ISOC) is an international, nonprofit organization formed in 1992 to provide support for the Internet standards process. ISOC accomplishes this through maintaining and supporting other Internet administrative bodies such as lAB, IETF,IRTF, and IANA (see the following sections). ISOC also promotes research and other scholarly activities relating to the Internet.

lAB

The Internet Architecture Board (lAB) is the technical advisor to the ISOC. The main purposes of the lAB are to oversee the continuing development of the *TCP/IP* Protocol Suite and to serve in a technical advisory capacity to research members of the Internet community. lAB accomplishes this through its two primary components, the Internet Engineering Task Force (IETF) and the Internet Research Task Force (IRTF). Another responsibility of the lAB is the editorial management of the RFCs, described earlier. lAB is also the external liaison between the Internet and other standards organizations and forums.

JETF

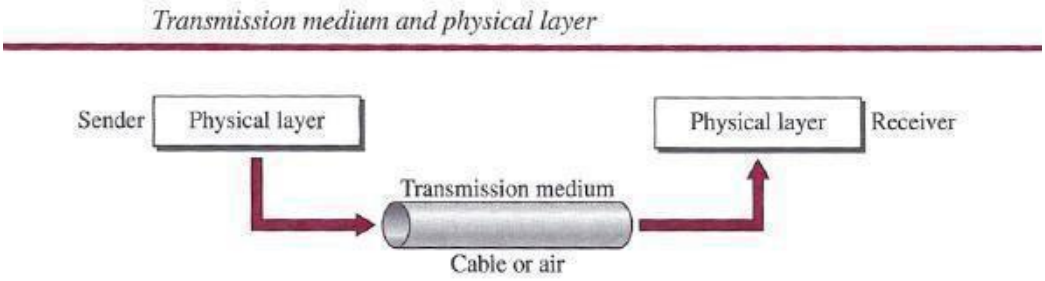
The Internet Engineering Task Force (IETF) is a forum of working groups managed by the Internet Engineering Steering Group (IESG). IETF is responsible for identifying operational problems and proposing solutions to these problems. IETF also develops and reviews specifications intended as Internet standards. The working groups are collected into areas, and each area concentrates on a specific topic. Currently nine areas have been defined. The areas include applications, protocols, routing, network management next generation (lPng), and securit

JRTF

The Internet Research Task Force (IRTF) is a forum of working groups managed by the Internet Research Steering Group (IRSG). IRTF focuses on long-term research topics related to Internet protocols, applications, architecture, and technology.

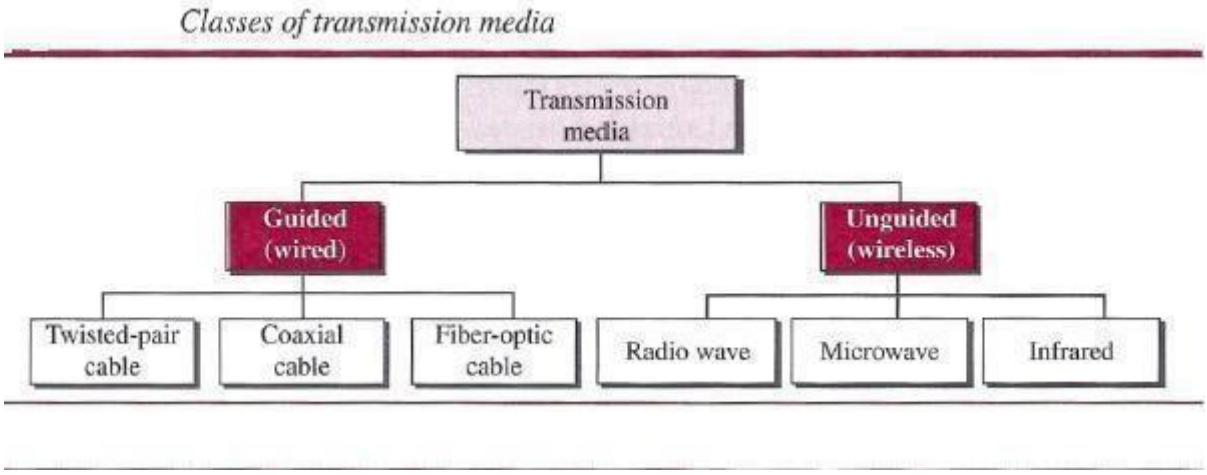
TRANSMISSION MEDIA

Transmission media are actually located below the physical layer and are directly controlled by the physical layer. We could say that transmission media belong to layer zero. Below figure shows the position of transmission media in relation to the physical layer.



In telecommunications, transmission media can be divided into two broad categories:

guided and unguided. Guided media include twisted-pair cable, coaxial cable, and fiber-optic cable. Unguided medium is free space.

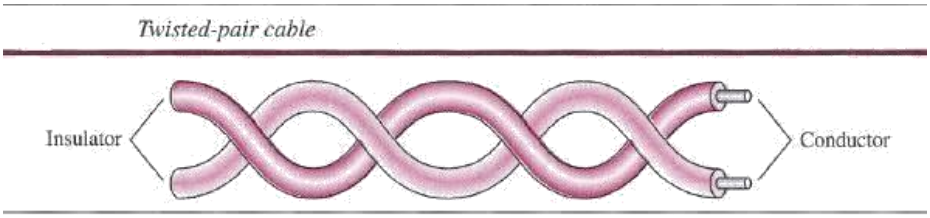


GUIDED MEDIA

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

Twisted-Pair Cable

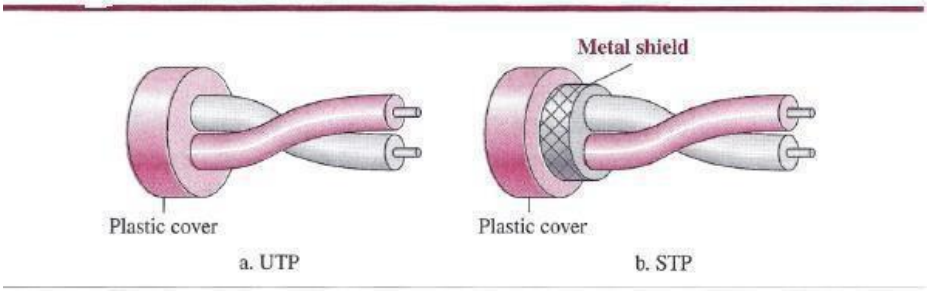
A twisted pair consists of two conductors (normally copper), each with its own plastic insulation, twisted together, as shown in following figure.



One of the wires is used to carry signals to the receiver, and the other is used only as a ground reference. The receiver uses the difference between the two. In addition to the signal sent by the sender on one of the wires, interference (noise) and crosstalk may affect both wires and create unwanted signals. If the two wires are parallel, the effect of these unwanted signals is not the same in both wires because they are at different locations relative to the noise or crosstalk sources (e.g., one is closer and the other is farther). This results in a difference at the receiver. By twisting the pairs, a balance is maintained. For example, suppose in one twist, one wire is closer to the noise source and the other is farther; in the next twist, the reverse is true. Twisting makes it probable that both wires are equally affected by external influences (noise or crosstalk). This means that the receiver, which calculates the difference between the two, receives no unwanted signals. The unwanted signals are mostly canceled out. From the above discussion, it is clear that the number of twists per unit of length (e.g., inch) has some effect on the quality of the cable.

Unshielded Versus Shielded Twisted-Pair Cable

The most common twisted-pair cable used in communications is referred to as *unshielded twisted-pair* (UTP). IBM has also produced a version of twisted-pair cable for its use, called *shielded twisted-pair* (STP). STP cable has a metal foil or braided mesh covering that encases each pair of insulated conductors. Although metal casing improves the quality of cable by preventing the penetration of noise or crosstalk, it is bulkier and more expensive. Below figure

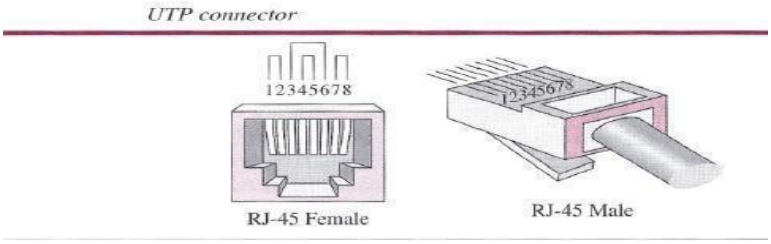


Categories

The Electronic Industries Association (EIA) has developed standards to classify unshielded twisted-pair cable into seven categories. Categories are determined by cable quality, with 1 as the lowest and 7 as the highest. Each EIA category is suitable for specific uses. Table below shows these categories.

Connectors

The most common UTP connector is **RJ45** (RJ stands for registered jack), as shown in below figure. The RJ45 is a keyed connector, meaning the connector can be inserted in only one way.



Performance

One way to measure the performance of twisted-pair cable is to compare attenuation versus frequency and distance. A twisted-pair cable can pass a wide range of frequencies. However, below figure shows that with increasing frequency, the attenuation, measured in decibels per kilometer (dB/km), sharply increases with frequencies above 100 kHz. Note that *gauge* is a measure of the thickness of the wire.

Applications

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

Coaxial Cable

Coaxial cable (or *coax)* carries signals of higher frequency ranges than those in twisted pair cable, in part because the two media are constructed quite differently. Instead of having two wires, coax has a central core conductor of solid or stranded wire (usually copper) enclosed in an insulating sheath, which is, in turn, encased in an outer conductor of metal foil, braid, or a combination of the two. The outer metallic wrapping serves both as a shield against noise and as the second conductor, which completes the circuit. This outer conductor is also enclosed in an insulating sheath, and the whole cable is protected by a plastic cover.

Coaxial Cable Standards

Coaxial cables are categorized by their Radio Government (RG) ratings. Each RG number denotes a unique set of physical specifications, including the wire gauge of the inner conductor, the thickness and type of the inner insulator, the construction of the shield, and the size and type of the outer casing. Each cable defined by an RG rating is adapted for a specialized function, as shown in below table.

Categories of coaxial cables

Category	*Impedance*	*Use*
RG-59	75 Ω	Cable TV
RG-58	50 Ω	Thin Ethernet
RG-11	50 Ω	Thick Ethernet

Coaxial Cable Connectors

To connect coaxial cable to devices, we need coaxial connectors. The most common type of connector used today is the Bayonet Neill-Concelman (BNC) connector. Below figure shows three popular types of these connectors: the BNC connector, the BNC T connector, and the BNC terminator. The BNC connector is used to connect the end of the cable to a device, such as a TV set. The BNC T connector is used in Ethernet networks (see Chapter 13) to branch out to a connection to a computer or other device. The BNC terminator is used at the end of the cable to prevent the reflection of the signal.

Performance

As we did with twisted-pair cable, we can measure the performance of a coaxial cable. the attenuation is much higher in coaxial cable than in twisted-pair cable. In other words, although coaxial cable has a much higher bandwidth, the signal weakens rapidly and requires the frequent use of repeaters.

Applications

Coaxial cable was widely used in analog telephone networks where a single coaxial network could carry 10,000 voice signals. Later it was used in digital telephone networks where a single coaxial cable could carry digital data up to 600 Mbps. However, coaxial cable in telephone networks has largely been replaced today with fiber optic cable. Cable TV networks also use coaxial cables. In the traditional cable TV network, the entire network used coaxial cable. Later, however, cable TV providers replaced most of the media with fiber-optic cable; hybrid networks use coaxial cable only at the network boundaries, near the consumer premises. Cable TV uses RG-59 coaxial cable. Another common application of coaxial cable is in traditional Ethernet LANs (see Because of its high bandwidth, and consequently high data rate, coaxial cable was chosen for digital transmission in early Ethernet LANs. The lOBase-2, or Thin Ethernet, uses RG-58 coaxial cable with BNC connectors to transmit data at 10 Mbps with a range of 185 m. The lOBase5, or Thick Ethernet, uses RG-ll (thick coaxial cable) to transmit 10 Mbps with a range of 5000 m. Thick Ethernet has specialized connectors.

Fiber-Optic Cable

A fiber-optic cable is made of glass or plastic and transmits signals in the form of light. To understand optical fiber, we first need to explore several aspects of the nature of light. Light travels in a straight line as long as it is moving through a single uniform substance. If a ray of light traveling through one substance suddenly enters another substance (of a different density), the ray changes direction. Below figure shows how a ray of light changes direction when going from a denser to a less dense substance. As the figure shows, if the angle **of incidence** *I* (the angle the ray makes with the line perpendicular to the interface between the two substances) is less than the **critical angle,** the ray **refracts** and moves closer to the surface. If the angle of incidence is equal to the critical angle, the light bends along the interface. If the angle is greater than the critical angle, the ray **reflects** (makes a turn) and travels again in the denser substance . Note that the critical angle is a property of the substance, and its value differs from one substance to another. Optical fibers use reflection to guide light through a channel. A glass or plastic core is surrounded by a cladding of less dense glass or plastic. The difference in density of the two materials must be such that a beam of light moving through the core is reflected off the cladding instead of being refracted into it. ***Propagation Modes*** Current technology supports two modes (multimode and single mode) for propagating light along optical channels, each requiring fiber with different physical characteristics. Multimode can be implemented in two forms: step-index or graded-index .

UNGUIDED MEDIA: WIRELESS

Unguided medium transport electromagnetic waves without using a physical conductor. This type of communication is often referred to as *wireless communication.* Signals are normally broadcast through free space and thus are available to anyone who has a device capable of receiving them. Below figure 7.17 shows the part of the electromagnetic spectrum, ranging from 3 kHz to 900 THz, used for wireless communication. Unguided signals can travel from the source to the destination in several ways: ground propagation, sky propagation, and line-of-sight propagation. In **ground propagation,** radio waves travel through the lowest portion of the atmosphere, hugging the earth. These low-frequency signals emanate in all directions from the transmitting antenna and follow the curvature of the planet. Distance depends on the amount of power in the signal: The greater the power, the greater the distance. **In sky propagation,** higher-frequency radio waves radiate upward into the ionosphere (the layer of atmosphere where particles exist as ions) where they are reflected back to earth. This type of transmission allows for greater distances with lower output power. **In line-of-sight propagation,** very high-frequency signals are transmitted in straight lines directly from antenna to antenna. The section of the electromagnetic spectrum defined as radio waves and microwaves is divided into eight ranges, called *bands*, each regulated by government authorities. These bands are rated from *very low frequency* (VLF) to *extremely high frequency* (EHF). Below table lists these bands, their ranges, propagation methods, and some applications

Radio Waves

Although there is no clear-cut demarcation between radio waves and microwaves, electromagnetic waves ranging in frequencies between 3 kHz and 1 GHz are normally called radio waves; waves ranging in frequencies between I and 300 GHz are called microwaves. However, the behavior of the waves, rather than the frequencies, is a better criterion for

classification. Radio waves, for the most part, are Omni-directional. When an antenna transmits radio waves, they are propagated in all directions.

This means that the sending and receiving antennas do not have to be aligned. A sending antenna sends waves that can be received by any receiving antenna. The Omni-directional property has a disadvantage, too. The radio waves transmitted by one antenna are susceptible to interference by another antenna that may send signals using the same frequency or band. Radio waves, particularly those waves that propagate in the sky mode, can travel long distances. This makes radio waves a good candidate for long-distance broadcasting such as AM radio.

Omni directional Antenna

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

Applications

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

Radio waves are used for multicast communications, such as radio and television, and paging systems.

Microwaves

Electromagnetic waves having frequencies between 1 and 300 GHz are called microwaves. Microwaves are unidirectional. When an antenna transmits microwaves, they can be narrowly focused. This means that the sending and receiving antennas need to be aligned. The unidirectional property has an obvious advantage. A pair of antennas can be aligned without interfering with another pair of aligned antennas.

The following describes some characteristics of microwave propagation:

- Microwave propagation is line-of-sight. Since the towers with the mounted antennas need to be in direct sight of each other, towers that are far apart need to be very tall. The curvature of the earth as well as other blocking obstacles does not allow two short towers to communicate by using microwaves. Repeaters are often needed for long distance communication.
- Very high-frequency microwaves cannot penetrate walls. This characteristic can be a disadvantage if receivers are inside buildings.
- The microwave band is relatively wide, almost 299 GHz. Therefore wider sub bands can be assigned, and a high data rate is possible.
- Use of certain portions of the band requires permission from authorities.

Unidirectional Antenna

Microwaves need unidirectional antennas that send out signals in one direction. Two types of antennas are used for microwave communications: the parabolic dish and the horn. A parabolic dish antenna is based on the geometry of a parabola: Every line parallel to the line of symmetry (line of sight) reflects off the curve at angles such that all the lines intersect in a common point called the focus. The parabolic dish works as a funnel, catching a wide range

of waves and directing them to a common point. In this way, more of the signal is recovered than would be possible with a single-point receiver.

Outgoing transmissions are broadcast through a horn aimed at the dish. The microwaves hit the dish and are deflected outward in a reversal of the receipt path. A horn antenna looks like a gigantic scoop. Outgoing transmissions are broadcast up a stem (resembling a handle) and deflected outward in a series of narrow parallel beams by the curved head. Received transmissions are collected by the scooped shape of the horn, in a manner similar to the parabolic dish, and are deflected down into the stem.

Applications

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

Microwaves are used for unicast communication such as cellular telephones, satellite networks, and wireless LANs.

Infrared

Infrared waves, with frequencies from 300 GHz to 400 THz (wavelengths from 1 mm to 770 nrn), can be used for short-range communication. Infrared waves, having high frequencies, cannot penetrate walls. This advantageous characteristic prevents interference between one system and another; a short-range communication system in one room cannot be affected by another system in the next room. When we use our infrared remote control, we do not interfere with the use of the remote by our neighbors. However, this same characteristic makes infrared signals useless for long-range communication. In addition, we cannot use infrared waves outside a building because the sun's rays contain infrared waves that can interfere with the communication.

Applications

The infrared band, almost 400 THz, has an excellent potential for data transmission. Such a wide bandwidth can be used to transmit digital data with a very high data rate. The *Infrared Data Association* elrDA), an association for sponsoring the use of infrared waves, has established standards for using these signals for communication between devices such as keyboards, mice, PCs, and printers. For example, some manufacturers provide a special port called the IrDA port that allows a wireless keyboard to communicate with a PC. The standard originally defined a data rate of 75 kbps for a distance up to 8 m. The recent standard defines a data rate of 4 Mbps.Infrared signals defined by IrDA transmit through line of sight; the IrDA port on the keyboard needs to point to the PC for transmission to occur. Infrared signals can be used for short-range communication in a closed area using line-of-sight propagation.

Multimedia

Recent developments in the multimedia applications such as voice over IP (telephony), video over IP (Skype), view sharing (YouTube), and television over IP (PPLive) has increased the number of users and the amount of time each user spends on the network.

Peer-to-Peer Applications

Peer-to-peer networking is also a new area of communication with a lot of potential.

Design Issues Data-Link Layer:

The data-link layer is located between the physical and the network layers. The data link layer provides services to the network layer; it receives services from the physical layer. Let us discuss services provided by the data-link layer. The duty scope of the data-link layer is node-to-node. When a packet is travelling in the Internet, the data-link layer of a node (host or router) is responsible for delivering a datagram to the next node in the path. For this purpose, the data-link layer of the sending node needs to encapsulate the datagram received from the network in a frame, and the data-link layer of the receiving node needs to decapsulate the datagram from the frame. In other words, the data-link layer of the source host needs only to encapsulate, the data-link layer of the destination host needs to decapsulate, but each intermediate node needs to both encapsulate and decapsulate. One may ask why we need encapsulation and decapsulation at each intermediate node. The reason is that each link may be using a different protocol with a different frame format. Even if one link and the next are using the same protocol, encapsulation and decapsulation are needed because the link-layer addresses are normally different. An analogy may help in this case. Assume a person needs to travel from her home to her friend's home in another city. The traveller can use three transportation tools. She can take a taxi to go to the train station in her own city, then travel on the train from her own city to the city where her friend lives, and finally

Framing

Definitely, the first service provided by the data-link layer is framing. The data-link layer at each node needs to encapsulate the datagram (packet received from the network layer) in a frame before sending it to the next node. The node also needs to decapsulate the datagram from the frame received on the logical channel. Although we have shown only a header for a frame, we will see in future chapters that a frame may have both a header and a trailer. Different data-link layers have different formats for framing. A packet at the data-link layer is normally called *a frame.*

Flow Control

Whenever we have a producer and a consumer, we need to think about flow control. If the producer produces items that cannot be consumed, accumulation of items occurs.

The sending data-link layer at the end of a link is a producer of frames; the receiving data-link layer at the other end of a link is a consumer. If the rate of produced frames is higher than the rate of consumed frames, frames at the receiving end need to be buffered while waiting to be consumed (processed). Definitely, we cannot have an unlimited buffer size at the receiving side. We have two choices. The first choice is to let the receiving data-link layer drop the frames if its buffer is full. The second choice is to let the receiving data-link layer send a feedback to the sending data-link layer to ask it to stop or slow down. Different data-link-layer protocols use different strategies for flow control. Since flow control also occurs at the transport layer, with a higher degree of importance, we discuss this issue in Chapter 23 when we talk about the transport layer.

Error Control

At the sending node, a frame in a data-link layer needs to be changed to bits, transformed to electromagnetic signals, and transmitted through the transmission media. At the receiving node, electromagnetic signals are received, transformed to bits, and put together to create a frame. Since electromagnetic signals are susceptible to error, a frame is susceptible to error. The error needs first to be detected. After detection, it needs to be either corrected at the receiver node or discarded and retransmitted by the sending node. Since error detection and correction is an issue in every layer (node-to node or host-to-host).

Congestion Control

Although a link may be congested with frames, which may result in frame loss, most data-link-layer protocols do not directly use a congestion control to alleviate congestion, although some wide-area networks do. In general, congestion control is considered an issue in the network layer or the transport layer because of its end-to-end nature.

CYCLIC CODES

Cyclic codes are special linear block codes with one extra property. In a cyclic code, if a codeword is cyclically shifted (rotated), the result is another codeword. For example, if 1011000 is a codeword and we cyclically left-shift, then 0110001 is also a codeword. In this case, if we call the bits in the first word *ao* to *a6,* and the bits in the second word *bo* to *b6,* we can shift the bits by using the following: In the rightmost equation, the last bit of the first word is wrapped around and becomes the first bit of the second word.

Cyclic Redundancy Check

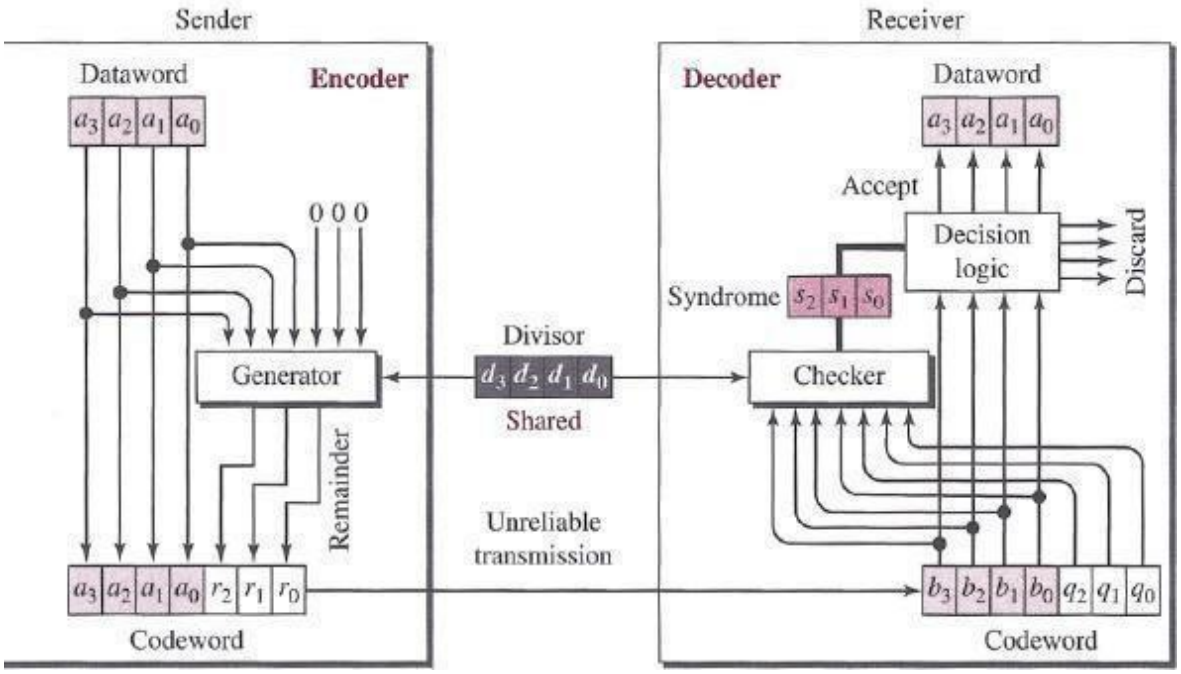
We can create cyclic codes to correct errors. However, the theoretical background required is beyond the scope of this book. In this section, we simply discuss a subset of cyclic codes called the cyclic redundancy check (CRC), which is used in networks such as LANs and WANs. Table below shows an example of a CRC code. We can see both the linear and cyclic properties of this code.

A CRC code with C(7, 4)

Dataword	*Codeword*	*Dataword*	*Codeword*
0000	0000000	1000	1000101
0001	0001011	1001	1001110
0010	0010110	1010	1010011
0011	0011101	1011	1011000
0100	0100111	1100	1100010
0101	0101100	1101	1101001
0110	0110001	1110	1110100
0111	0111010	1111	1111111

Figure below shows one possible design for the encoder and decoder.

CRC encoder and decoder

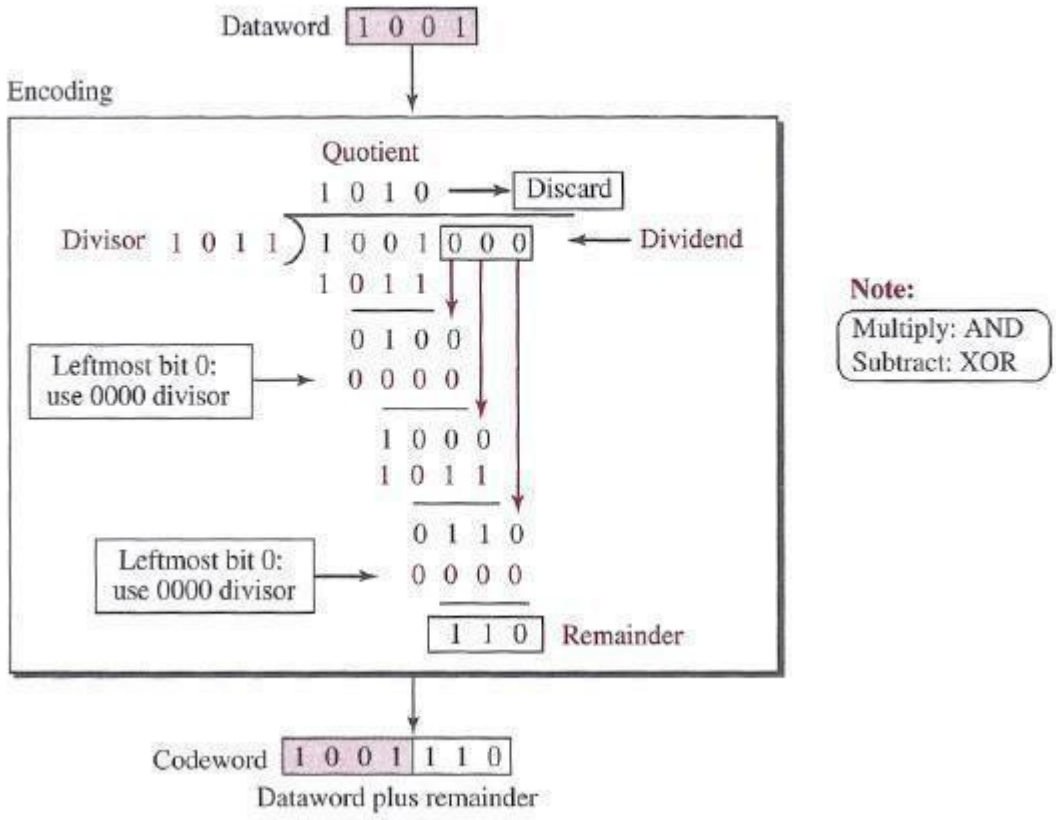


In the encoder, the data word has *k* bits (4 here); the codeword has *n* bits (7 here). The size of the data word is augmented by adding *n* - *k* (3 here) Os to the right-hand side of the word. The *n-bit* result is fed into the generator. The generator uses a divisor of size *n* - *k* + 1 (4 here), predefined and agreed upon. The generator divides the augmented data word by the divisor (modulo-2 division). The quotient of the division is discarded; the remainder *(r2rlrO)* is appended to the data word to create the codeword. The decoder receives the codeword (possibly corrupted in transition). A copy of all *n* bits is fed to the checker, which is a replica of the generator. The remainder produced by the checker is a syndrome of *n* - *k* (3 here) bits, which is fed to the decision logic analyzer. The analyzer has a simple function. If the syndrome bits are all Os, the 4 leftmost bits of the codeword are accepted as the data word (interpreted as no error); otherwise, the 4 bits are discarded (error).

Encoder

Let us take a closer look at the encoder. The encoder takes a data word and augments it with *n* - *k* number of Os. It then divides the augmented data word by the divisor, as shown in below figure.

Division in CRC encoder



Decoder

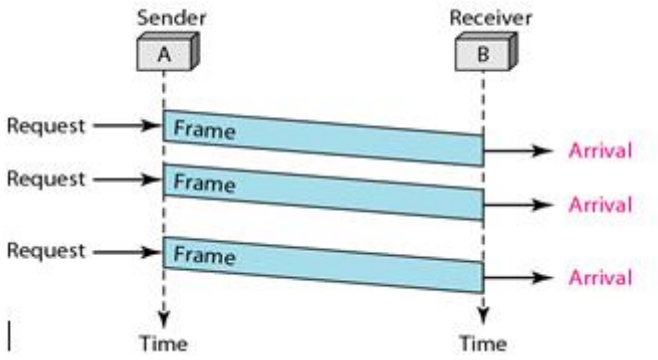
The codeword can change during transmission. The decoder does the same division process as the encoder. The remainder of the division is the syndrome. If the syndrome is all Os, there is no error with a high probability; the data word is separated from the received codeword and accepted. Otherwise, everything is discarded. Figure 10.7 shows two cases: The left-hand figure shows the value of the syndrome when no error has occurred; the syndrome is 000. The right-hand part of the figure shows the case in which there is a single error. The syndrome is not all Os (it is 011).

ELEMENT DATA LINK PROTOCOLS AND SLIDING WINDOW PROTOCOL

Traditionally four protocols have been defined for the data-link layer to deal with flow and error control: Simple, Stop-and-Wait, Go-Back-N, and Selective-Repeat. Although the first two protocols still are used at the data-link layer, the last two have disappeared.

Simple Protocol

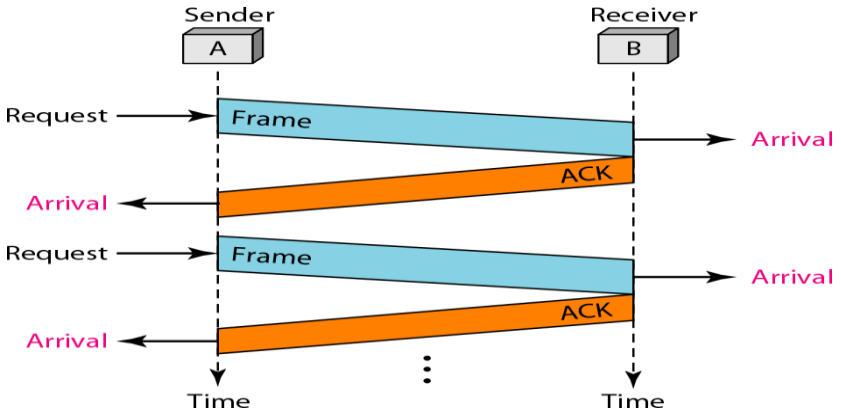
Our first protocol is a simple protocol with neither flow nor error control. We assume that the receiver can immediately handle any frame it receives. In other words, the receiver can never be overwhelmed with incoming frames. Below figure shows the layout for this protocol.



The data-link layer at the sender gets a packet from its network layer, makes a frame out of it, and sends the frame. The data-link layer at the receiver receives a frame from the link, extracts the packet from the frame, and delivers the packet to its network layer. The data-link layers of the sender and receiver provide transmission services for their network layers.

Stop-and- Wait Protocol

Our second protocol is called the Stop-and- Wait protocol, which uses both flow and error control. We show a primitive version of this protocol here, but we discuss the more sophisticated version in Chapter 23 when we have learned about sliding windows. In this protocol, the sender sends one frame at a time and waits for an acknowledgment before sending the next one. To detect corrupted frames, we need to add a CRC to each data frame. When a frame arrives at the receiver site, it is checked. If its CRC is incorrect, the frame is corrupted and silently discarded. The silence of the receiver is a signal for the sender that a frame was either corrupted or lost. Every time the sender sends a frame, it starts a timer. If an acknowledgment arrives before the timer expires, the timer is stopped and the sender sends the next frame (if it has one to send). If the timer expires, the sender resends the previous frame, assuming that the frame was either lost or corrupted. This means that the sender needs to keep a copy of the frame until its acknowledgment arrives. When the corresponding acknowledgment arrives, the sender discards the copy and sends the next frame if it is ready. Below figure shows the outline for the Stop-and-Wait protocol. Note that only one frame and one acknowledgment can be in the channels at any time.



HDLC

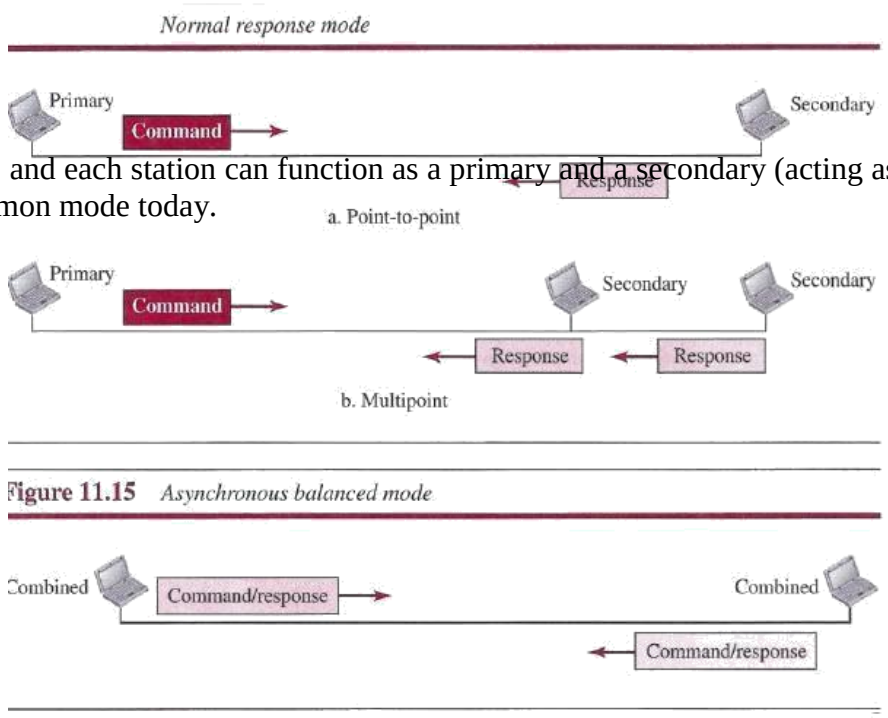
High-level Data Link Control (HDLC) is a bit-oriented protocol for communication over point - to-point and multipoint links. It implements the Stop-and- Wait protocol we discussed earlier. Although this protocol is more a theoretical issue than practical, most of the concept defined in this protocol is the basis for other practical protocols such as PPP, which we discuss next, or the Ethernet protocol.

Configurations and Transfer Modes

HDLC provides two common transfer modes that can be used in different configurations:

Normal response mode (NRM)* and *Asynchronous balanced mode (ABM)

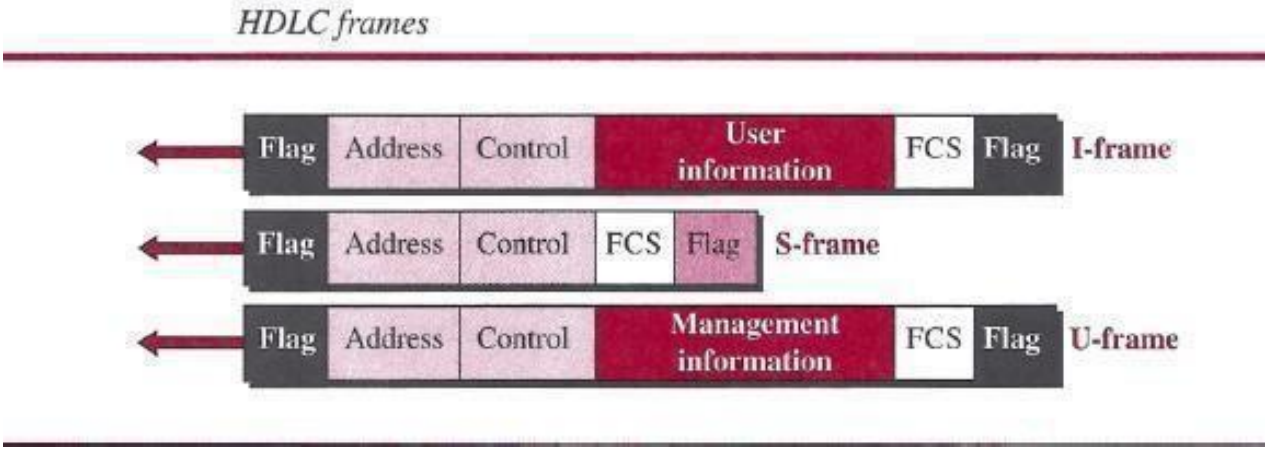
In *normal response mode (NRM)*, the station configuration is unbalanced. We have one primary station and multiple secondary stations. A *primary station* can send commands; a *secondary station* can only respond. The NRM is used for both point-to-point and multipoint links, as shown in below Figure.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



to-point, and each station can function as a primary and a secondary (acting as peers) this is the common mode today.

Figure 11.15 *Asynchronous balanced mode*

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



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

➢ D *Flag field.* This field contains synchronization pattern 01111110, which identifies both the beginning and the end of a frame.

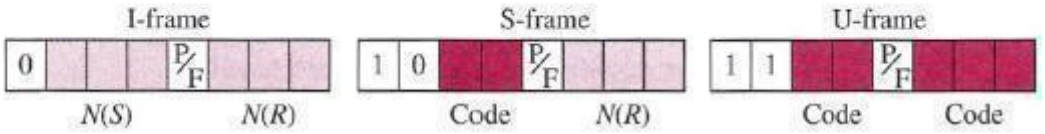
➢

D *Address field.* This field contains the address of the secondary station. If a primary station created the frame, it contains a *to* address. If a secondary station creates the frame, it contains *a from* address. The address field can be one byte or several bytes long, depending on the needs of the network.

- *Control field.* The control field is one or two bytes used for flow and error control.
- *Information field.* The information field contains the user's data from the network layer or management information. Its length can vary from one network to another.
- *FCS field.* The frame check sequence (FCS) is the HDLC error detection field. It can contain either a 2- or 4-byte CRC.

The control field determines the type of frame and defines its functionality. So let us discuss the format of this field in detail. The format is specific for the type of frame, as shown in below Figure.

Control field format for the different frame types



Control Field for I-Frames

I- frames are designed to carry user data from the network layer. In addition, they can include flow- and error-control information (piggybacking). The subfields in the control field are used to define these functions. The first bit defines the type. If the first bit of the control field is 0, this means the frame is an I-frame. The next 3 bits, called *N(S),* define the sequence number of the frame. Note that with 3 bits, we can define a sequence number between 0 and 7. The last 3 bits, called *N(R),* correspond to the acknowledgment number when piggybacking is used. The single bit between *N(S)* and *N(R)* is called the *PIF* bit. The *PIF* field is a single bit with a dual purpose. It has meaning only when it is set (bit = 1) and can mean poll or final. It means *poll* when the frame is sent by a primary station to a secondary (when the address field contains the address of the receiver). It means *final* when the frame is sent by a secondary to a primary (when the address field contains the address of the sender).

Control Field for S-Frames

Supervisory frames are used for flow and error control whenever piggybacking is either impossible or inappropriate. S-frames do not have information fields. If the first 2 bits of the control field are 10, this means the frame is an S-frame. The last 3 bits, called *N(R),* correspond to the acknowledgment number (ACK) or negative acknowledgment number (NAK), depending on the type of S-frame. The 2 bits called *code* are used to define the type of S-frame itself. With 2 bits, we can have four types of S-frames, as described below:

- ***Receive ready (RR)*** If the value of the code subfield is 00, it is an RR S-frame. This kind of frame acknowledges the receipt of a safe and sound frame or group of frames. In this case, the value of the *N(R)* field defines the acknowledgment number.

- ***Receive not ready (RNR)*** If the value of the code subfield is 10, it is an RNR S frame. This kind of frame is an RR frame with additional functions. It acknowledges the receipt of a frame or group of frames, and it announces that the receiver is busy and cannot receive more frames. It acts as a kind of congestion-control mechanism by asking the sender to slow down. The value of *N(R)* is the acknowledgment number.

- ***Reject (REJ)*** If the value of the code subfield is 01, it is an REJ S-frame. This is a NAK frame, but not like the one used for Selective Repeat ARQ. It is a NAK that can be used in *Go-Back-N* ARQ to improve the efficiency of the process by informing the sender, before the sender timer expires, that the last frame is lost or damaged. The value of *N(R)* is the negative acknowledgment number.

- ***Selective reject (SREJ)*** If the value of the code subfield is 11, it is an SREJ S frame. This is a NAK frame used in Selective Repeat ARQ. Note that the HDLC Protocol uses the term *selective reject* instead of *selective repeat.* The value of *N(R)* is the negative acknowledgment number.

Control Field or V-Frames

Unnumbered frames are used to exchange session management and control information between connected devices. Unlike S-frames, U-frames contain an information field, but one used for system management information, not user data. As with S-frames, however, much of the information carried by If-frames is contained in codes included in the control field. If-frame codes are divided into two sections: a 2-bit prefix before the *PI* F bit and a 3-bit suffix after the *PIP* bit. Together, these two segments (5 bits) can be used to create up to 32 different types of U-frames.

Control Field for V-Frames

Unnumbered frames are used to exchange session management and control information between connected devices. Unlike S-frames, U-frames contain an information field, but one used for system management information, not user data. As with S-frames, however, much of the information carried by U-frames is contained in codes included in the control field. U-frame codes are divided into two sections: a 2-bit prefix before the *PIP* bit and a 3-bit suffix after the *P/F* bit. Together, these two segments (5 bits) can be used to create up to 32 different types of If-frames.

POINT-TO-POINT PROTOCOL (PPP)

One of the most common protocols for point-to-point access is the **Point-to-Point Protocol (PPP).** Today, millions of Internet users who need to connect their home computers to the server of an Internet service provider use PPP. The majority of these users have a traditional modem; they are connected to the Internet through a telephone line, which provides the services of the physical layer. But to control and manage the transfer of data, there is a need for a point-to-point protocol at the data-link layer. PPP is by far the most common.

Services

The designers of PPP have included several services to make it suitable for a point-to point protocol, but have ignored some traditional services to make it simple.

Services Provided by PPP

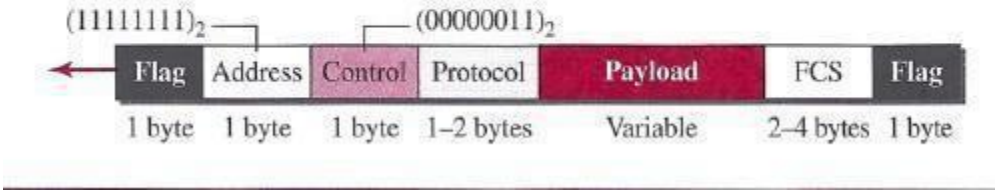
PPP defines the format of the frame to be exchanged between devices. It also defines how two devices can negotiate the establishment of the link and the exchange of data. PPP is designed to accept payloads from several network layers (not only IP). Authentication is also provided in the protocol, but it is optional. The new version of PPP, called *Multilink PPP*, provides connections over multiple links. One interesting feature of PPP is that it provides network address configuration. This is particularly useful when a home user needs a temporary network address to connect to the Internet.

Framing

PPP uses a character-oriented (or byte-oriented) frame. Below figure shows the format of a **PPP** frame. The description of each field follows:

- ***Flag A*** PPP frame starts and ends with a l-byte flag with the bit pattern 01111110.

PPP frame format



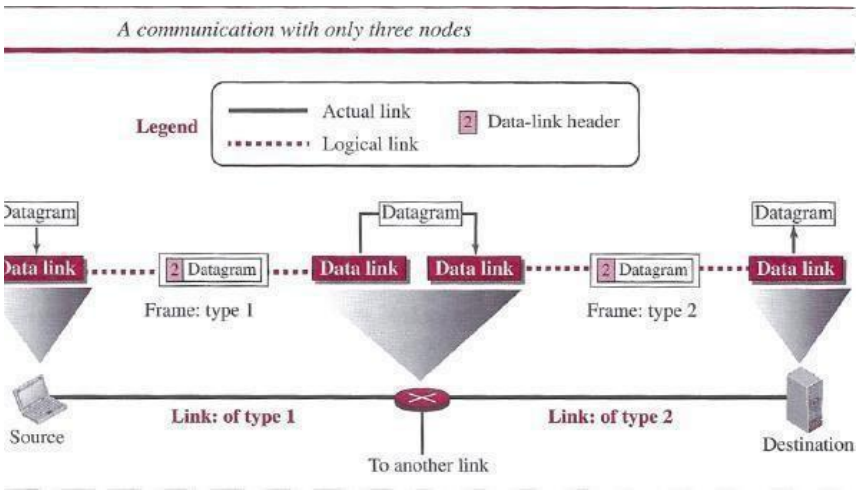
  - ***Address*** The address field in this protocol is a constant value and set to 11111111 (broadcast address).

- **D *Control*** *This* field is set to the constant value 00000011 (imitating unnumbered frames in HDLC). As we will discuss later, PPP does not provide any flow control. Error control is also limited to error detection.

- ***Protocol*** The protocol field defines what is being carried in the data field: either user data or other information. This field is by default 2 bytes long, but the two parties can agree to use only 1 byte.

- ***Payload field*** This field carries either the user data or other information that we will discuss shortly. The data field is a sequence of bytes with the default of a maximum of 1500 bytes; but this can be changed during negotiation. The data field is byte-stuffed if the flag byte pattern appears in this field. Because there is no field defining the size of the data field, padding is needed if the size is less than the maximum default value or the maximum negotiated value D *FCS.* The frame check sequence (FCS) is simply a 2-byte or 4-byte standard CRC.

Reach her friend's home using another taxi. Here we have a source node, a destination node, and two intermediate nodes. The traveler needs to get into the taxi at the source node, get out of the taxi and get into the train at the first intermediate node (train station in the city where she lives), get out of the train and get into another taxi at the second intermediate node (train station in the city where her friend lives), and finally get out of the taxi when she arrives at her destination. A kind of encapsulation occurs at the source node, encapsulation and decapsulation occur at the intermediate nodes, and decapsulation occurs at the destination node. For simplicity, we have assumed that we have only one router between the source and destination. The datagram received by the data-link layer of the source host is encapsulated in a frame. The frame is logically transported from the source host to the router. The frame is decapsulated at the data-link layer of the router and encapsulated at another frame. The new frame is logically transported from the router to the destination host. Note that, although we have shown only two data-link layers at the router, the router actually has three data-link layers because it is connected to three physical links.



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Whenever we have a producer and a consumer, we need to think about flow control. If the producer produces items that cannot be consumed, accumulation of items occurs. The sending data-link layer at the end of a link is a producer of frames; the receiving data-link layer at the other end of a link is a consumer. If the rate of produced frames is higher than the rate of consumed frames, frames at the receiving end need to be buffered while waiting to be consumed (processed). Definitely, we cannot have an unlimited buffer size at the receiving side. We have two choices. The first choice is to let the receiving data-link layer drop the frames if its buffer is full. The second choice is to let the receiving data-link layer send a feedback to the sending data-link layer to ask it to stop or slow down. Different data-link-layer protocols use different strategies for flow control. Since flow control also occurs at the transport layer, with a higher degree of importance, we discuss this issue in Chapter 23 when we talk about the transport layer.

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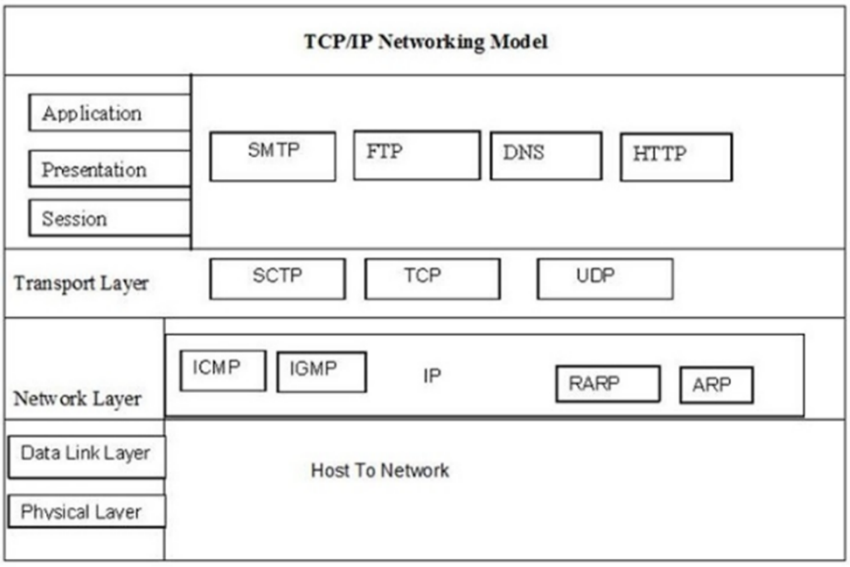
Layering

Layering means to break up the sending of messages into separate components and activities. Each component handles a different part of the communication. This can be referred to as the Transmission Control Protocol/Internet Protocol (TCP/IP) model.

TCP/IP Protocol Suite

TCP Stands For Transmission Control Protocol. IP Stands For Internet Protocol

- The Main Use of the TCP/IP is a Network Model Designed to support Network Communication, Even if the Computer Are from the different manufactures pc's .
- TCP/IP is a Practical Model Developed to meet the Needs of the Original Internet design. TCP/IP are the set of Two Protocols it consist of Numerous Protocols at different Layers.
- The TCP/IP model consists of four layers: the application layer, transport layer, network layer and physical layer.



1. Application layer

- In TCP/IP, the Application layer protocols provide services to the application software running on a computer. The application layer uses HTTP,FTP, and SMTP protocols. The application layer provides an interface between the software running on a computer and the network itself.
- **FTP (File Transfer Protocol)**
- It is one of the widely used application layer protocol of the TCP/IP protocol suite. FTP is basically used to exchange data between two host devices over the Internet securely.
- It is referred to as one of the safest modes of file sharing among systems, and thus it is deployed by large industries, universities, and offices.

SMTP(Simple Mail Transfer Protocol)

SMTP is the standardization for transmission of electronic mails on the Internet.It is used by the e-mail server for sending and receiving messages, but the client host-based application only uses it for sending messages to the mail server. For receiving purposes, they use POP3 or IMAP.

DNS (Domain Name Server)

DNS is a directory service that provides a mapping between the name of a host on the network and its numerical address. DNS is required for the functioning of the internet

HTTP

HTTP stands for Hypertext Transfer Protocol. It is an application protocol Hypertext Transfer Protocol (HTTP) is a method for encoding and transporting information between a client (such as a web browser) and a web server. HTTP is the primary protocol for transmission of information across the Internet.

2. THE TRANSPORT LAYER

- It is responsible for maintaining the communication between the sender and receiver. TCP or UDP (User Datagram Protocol) is used for this purpose.
- At the sending node, the transport layer receives the message from the application layer. When the message reaches the transport layer, one of the transport layer protocols, i.e., TCP or UDP, is selected.
- TCP supports segmentation. So, if the message is large, TCP divides it into smaller pieces and adds a header to form a TCP segment.
- On the other hand, UDP does not support segmentation, so the applications using UDP should send messages short enough to fit into one UDP datagram.

3. THE 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. :
- Logical Addressing
- Routing

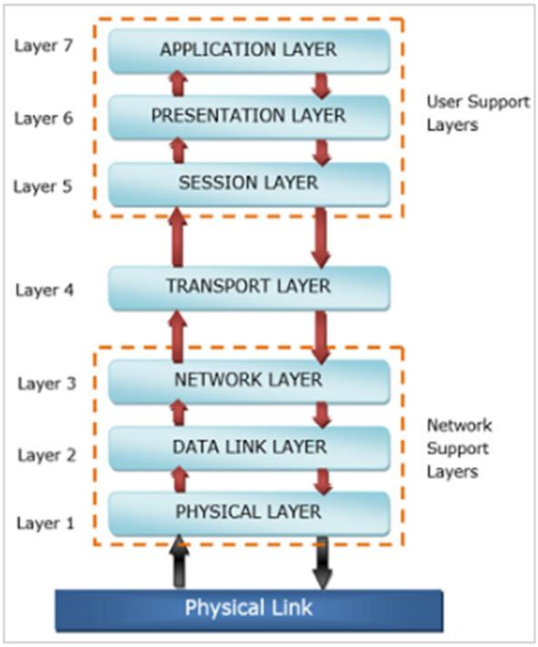
- **Logical Addressing**
- Every computer in a network has a unique IP address. The network layer assigns sender and receiver's IP addresses to each segment or datagram to form an IP Packet. IP addresses are assigned to ensure that each IP packet can reach the correct destination present in different networks.
- **Routing**
- Routing is a method of moving an IP packet from source to destination present in different networks. Routing is not needed if the source and destination computers are present in the same network.
- For communications within a network, the task is usually simple. module takes the destination IP address from the IP packet and returns the MAC address of the destination computer. It is then used to create an Ethernet frame which is delivered directly to the destination as it is present in the same network, . no routing is needed.

4. Physical layer

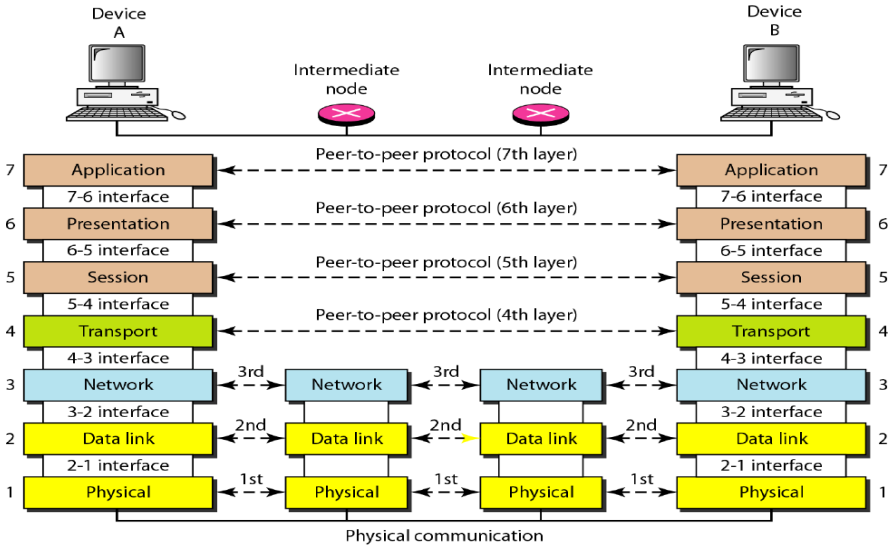
- The physical layer is responsible for carrying individual bits in a frame across the link. Although the physical layer is the lowest level in the TCPIIP protocol suite, the communication between two devices at the physical layer is still a logical communication because there is another, hidden layer, the transmission media, under the physical layer.
- Two devices are connected by a transmission medium (cable or air). We need to know that the transmission medium does not carry bits; it carries electrical or optical signals.

OSI Reference Model

- OSI stands for Open Systems Interconnection Created by International Standards Organization (ISO).
- This layered model is a conceptualized view of how one system should communicate with the other, using various protocols defined in each layer. Further, each layer is designated to a well-defined part of communication system.

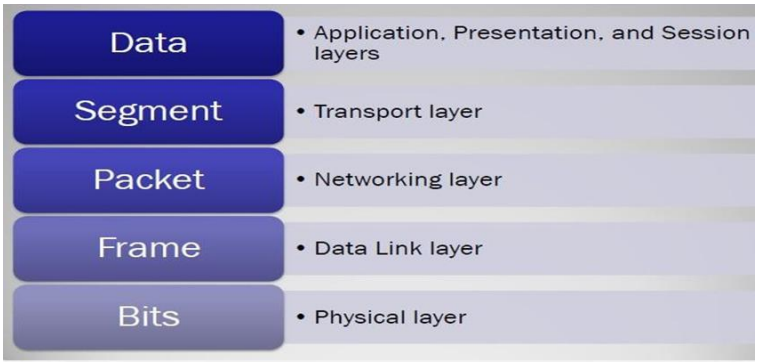


The Interaction between layers in the OSI model



1. Physical Layer

- The Physical Layer is the lowermost layer in the OSI model and its major responsibility includes the actual propagation of the unstructured data bits (0's and 1's) across the network, from the physical layer of the sending device to the physical layer of the receiving device.
- The Physical layer contains information in the form of bits. It transmits individual bits from one node to the next node. The transmission media defined by the physical layer include metallic cable, optical fiber, and the wireless radio-wave.



2. Data-Link Layer

It is the second layer of the OSI model. The data link layer is responsible for providing error-free communication across the physical link connecting the primary and secondary nodes within a network.

Functions of the Data-link layer

Framing: The data link layer translates the physical's raw bit stream into packets known as Frames. The Data link layer adds the header and trailer to the frame. The header which is added to the frame contains the hardware destination and source address.

- Error handling − It is used to soles the damaged, lost, and duplicate frames.
- Flow Control − It keeps a fast transmitter from flooding a slow receiver.
- Access Control − In access control, if many hosts have usage of the medium, When a single communication channel is shared by multiple devices, the MAC sub-layer of data link layer helps to determine which device has control over the channel at a given time.

3. Network Layer

- The network layer provides details that enable data to be routed between devices in an environment using multiple networks, sub-networks, or both.
- The networking components that operate at the network layer include routers and their software. It determines which network configuration is most appropriate for the function provided by the network and addresses and routes data within a network by establishing, maintaining, and terminating connectors between them.
- It provides the upper layers of the hierarchy with independence from the data transmission and switching technologies used to interconnect systems.

4. Transport Layer

We can say that the transport layer controls and ensures the end-to-end integrity of the data message propagated through the network between two devices, providing the reliable, transparent transfer of data between the endpoints.

Segmentation and Reassembly − In this, a message is divided into small pieces. Reassemble the message correctly upon arriving at the destination.

Reliability − It ensures that packets arrive at their destination. Reassembles out-of-order messages.

Service Decisions − It is used to check what types of service to provide error-free point-to-point, datagram, etc.

Mapping − It determines which messages belong to which connections.

Naming − It must be translated into an internal address and route, send to node XYZ.

Flow Control − It keeps a fast transmitter from flooding a slow receiver.

Error Control − To retransmit the damaged segments.

5. Session Layer

The session layer creates communication channels between devices. It is responsible for opening sessions, ensuring they remain open and functional while the data is being transferred, and close the session when the communication ends. The session layer can also set checkpoints during a data transfer. If a session is interrupted, then the devices can resume data transfer from the last checkpoint.

- Session Layer Responsibilities −
- Network log-on and log-off procedures
- User authentication
- Determines the type dialog available − simplex, half-duplex, and full-duplex.
- Synchronization of data flow for recovery purposes.
- Creation of dialog units and activity units.

6. Presentation Layer

The presentation layer prepares the data for its upper layer or the application layer. It defines how two devices should encode, encrypt, and compress the data. The presentation layer receives any data transmitted by the application layer and prepares it for transmission over the session layer. It specifies how the end-user applications should format the data. This layer provides for the translation between the local representation of data and the representation of data that will be used for transfer between the end-users.

7. Application Layer

The application layer is the topmost layer in the OSI model and acts as the general manager of the network by proving access to the OSI environment. This layer provides distributed information services and controls the sequence of activities within an application and also the sequence of events between the computer application and the user of the application. The application layer uses HTTP, FTP, POP, SMTP, and DNS protocols that allow the software to send and receive information and present meaningful data to users.

Difference between OSI Model And TCP/IP Model

OSI MODEL	TCP/IP MODEL
Contains 7 Layers	Contains 4 Layers
Uses Strict Layering resulting in vertical layers.	Uses Loose Layering resulting in horizontal layers.
Supports both connectionless & connection-oriented communication in the Network layer, but only connection-oriented communication in Transport Layer	Supports only connectionless communication in the Network layer, but both connectionless & connection-oriented communication in Transport Layer
It distinguishes between Service, Interface and Protocol.	Does not clearly distinguish between Service, Interface and Protocol.
Protocols are better hidden and can be replaced relatively easily as technology changes (No transparency)	Protocols are not hidden and thus cannot be replaced easily. (Transparency) Replacing IP by a substantially different protocol would be virtually impossible
OSI reference model was devised before the corresponding protocols were designed.	The protocols came first and the model was a description of the existing protocols