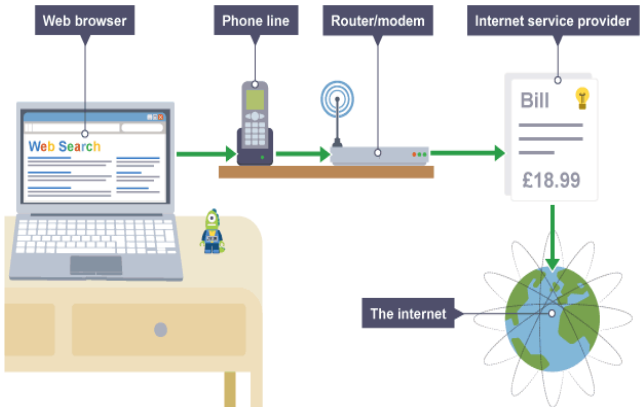
Network hardware:

Networks are created when two or more computers are connected. Files are sent over a network as data packets. Networks can be made in different topologies.

Networking hardware

Computers need networking hardware in order to connect to each other. Routers, hubs, switches and bridges are all pieces of networking equipment that can perform slightly different tasks. A router can often incorporate hubs, switches and wireless access within the same hardware.



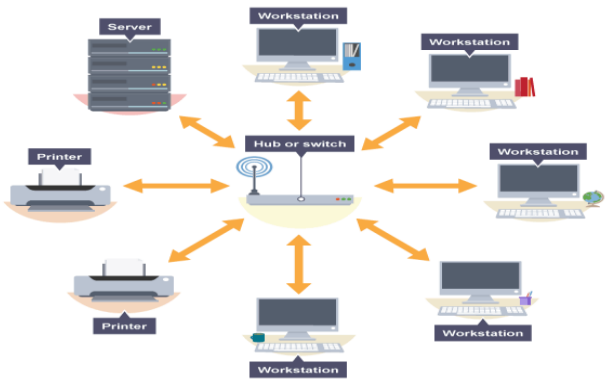
Routers

A **router** can form a LAN by connecting devices within a building. It also makes it possible to connect different networks together. Homes and businesses use a router to connect to the internet. A router can often incorporate a modem within the hardware.

Modems

A **modem** enables a computer to connect to the internet over a telephone line. A modem converts digital signals from a computer to analogue signals that are then sent down the telephone line. A modem on the other end converts the analogue signal back to a digital signal which another computer can understand.

Hubs, bridges and switches



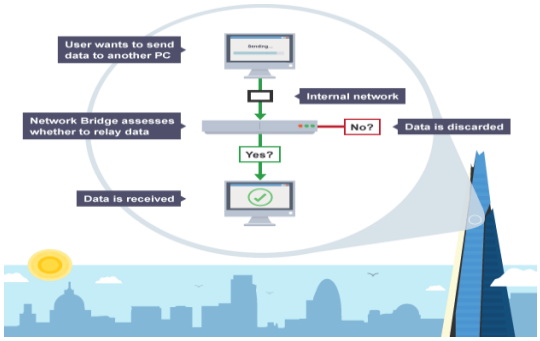
Hubs, **bridges** and **switches** allow multiple devices to connect to the router and they transfer data to all devices on a network. A router is a more complex device that usually includes the capability of hubs, bridges and switches.

Hubs

A hub broadcasts data to all devices on a network. This can use a lot of bandwidth as it results in unnecessary data being sent - not all computers might need to receive the data. A hub would be useful to link up a few games consoles for a local multiplayer game using a wired LAN.

Bridges

A **bridge** is used to connect two separate LAN networks. A computer can act as a bridge through the operating system. A bridge looks for the receiving device before it sends the message. This means that it will not send a message if the receiving computer is not there. It will check to see if the receiver has already had the message. This can help save unnecessary data transfers, which improves the performance of a network.



Switches

A **switch** performs a similar role to a hub and a bridge but is more powerful. It stores the MAC addresses of devices on a network and filters data packets to see which devices have asked for them. This makes a switch more efficient when demand is high. If, for example, a game involved lots of data being passed between machines, then a switch could reduce the amount of latency.

Wireless access points

Wireless access points (WAPs) are required to connect to a network wirelessly. WAPs are usually built into the broadband router.

Device addresses

Data packets include the addresses of the devices they are going to and coming from. Computers need a network interface card to connect to a network. All devices on a network have a MAC address.

MAC address

Every piece of hardware on a network has a unique **MAC address**. This is embedded in the hardware when the product is made in the factory, and the user cannot change it. On a computer, the MAC address is a

unique code built into a NIC. No two computers have the same MAC address. A MAC address is made up of 48 bits of data, usually written as 12 hexadecimal characters.

Network interface card (NIC)

NICs enable desktop and laptop computers to connect to a network. NICs are small circuit boards that connect to the motherboard. Smartphones also use a GSM chip to connect to the telephone network. Games consoles contain a NIC card so users can access the internet, download games and play online.

Types of network

There are different networking models for how to connect computers over a network. Computers that request information are called clients and computers that provide information are servers. But the client and server relationship can be organised in different ways.

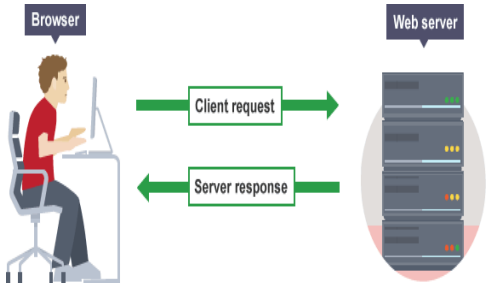
The most widely-used models are client-server or peer-to-peer (P2P).

Client-server

The client-server model is the relationship between two computers in which one, the client, makes a service request from another, the server. The key point about a client-server model is that **the client is dependent on the server to provide and manage the information**.

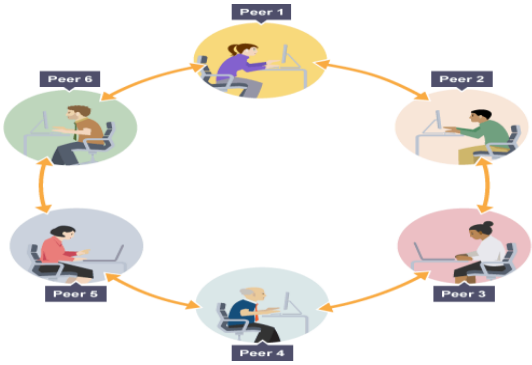
For example, websites are stored on web servers. A web browser is the client which makes a request to the server, and the server sends the website to the browser.

Popular websites need powerful servers to serve thousands or millions of clients, all making requests at the same time. The client side of a web application is often referred to as the front end. The server side is referred to as the back end.



Peer-to-peer (P2P)

In a P2P network, no single provider is responsible for being the server. Each computer stores files and acts as a server. Each computer has equal responsibility for providing data.



In the client-server model, many users trying to access a large file, such as a film, would put strain on one server. In the peer-to-peer model, many users on the network could store the same file. Each computer can then send sections of the file, sharing the workload. Each client can download and share files with other users.

P2P is ideal for sharing files. P2P would be unsuitable for a service such as booking tickets, as one server needs to keep track of how many tickets are left. Also, on P2P networks no single computer is responsible for storing a file - anyone can delete files as they wish.

Differences between client-server and P2P networks

	Client-server	**P2P**
Security	The server controls security of the network.	No central control over security.
Management	The server manages the network. Needs a dedicated team of people to manage the server.	No central control over the network. Anyone can set up.
Dependency	Clients are dependent on the server.	Clients are not dependent on a central server.
Performance	The server can be upgraded to be made more powerful to cope with high demand.	If machines on the network are slow they will slow down other machines.
Backups	Data is all backed up on the main server.	Each computer has to be backed up. Data can easily be deleted by users.

REPEATER

HUB

SWITCH

BRIDGE

GATEWAY

ROUTER

CABLES

Network software:

Network software encompasses a broad range of software used for design, implementation, and operation and monitoring of computer networks. Traditional networks were hardware based with software embedded. With the advent of Software – Defined Networking (SDN), software is separated from the hardware thus making it more adaptable to the ever-changing nature of the computer network.

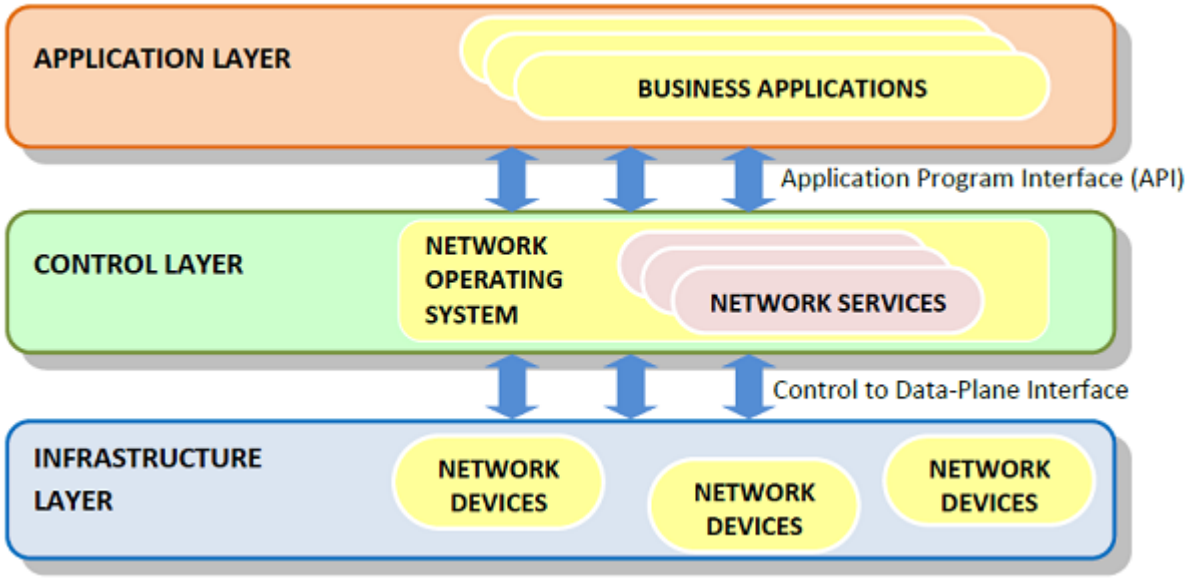
Functions of Network Software

- Helps to set up and install computer networks
- Enables users to have access to network resources in a seamless manner
- Allows administrations to add or remove users from the network
- Helps to define locations of data storage and allows users to access that data

- Helps administrators and security system to protect the network from data breaches, unauthorized access and attacks on a network
- Enables network virtualizations

SDN Framework

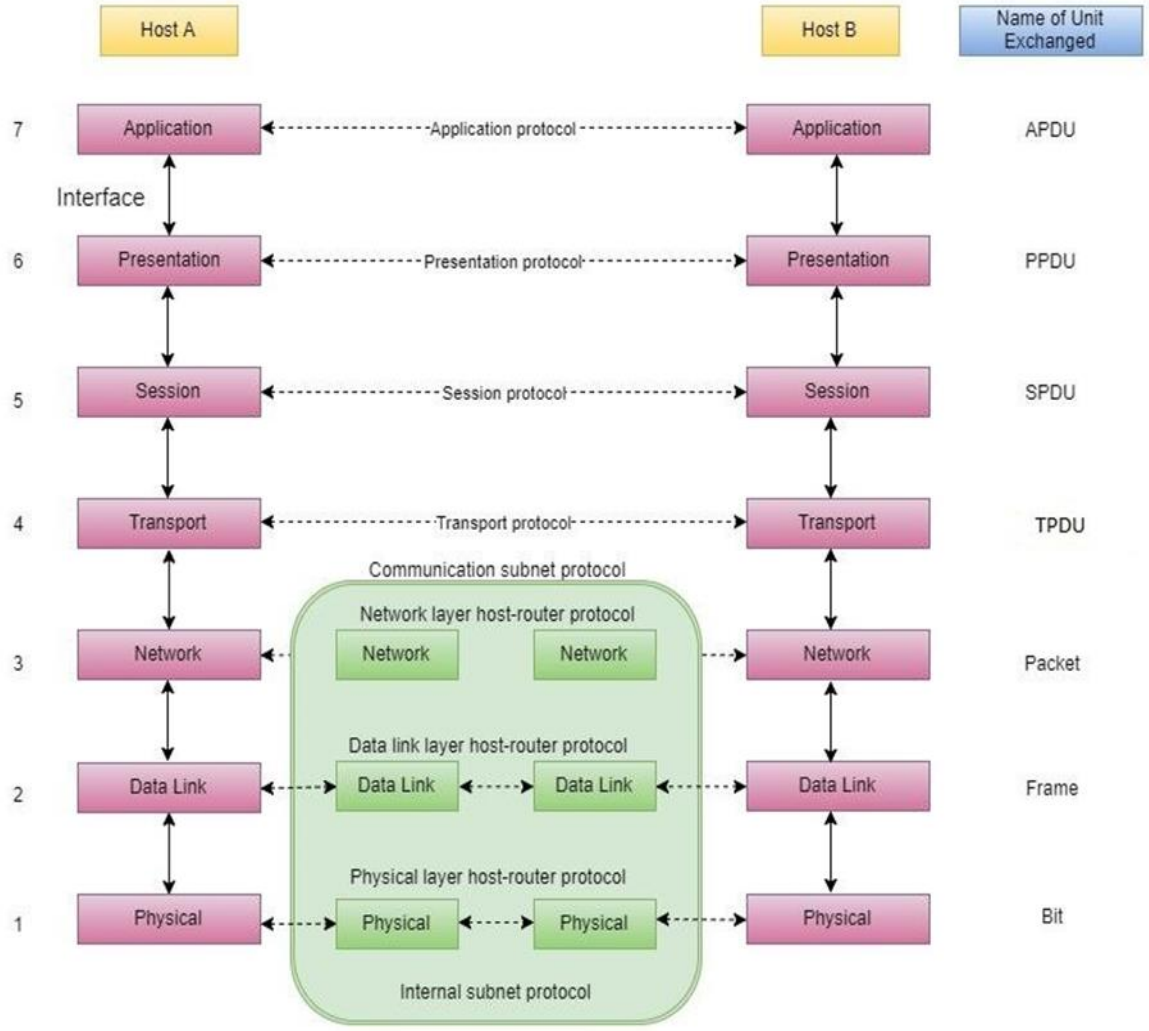
The Software Defined Networking framework has three layers as depicted in the following diagram −



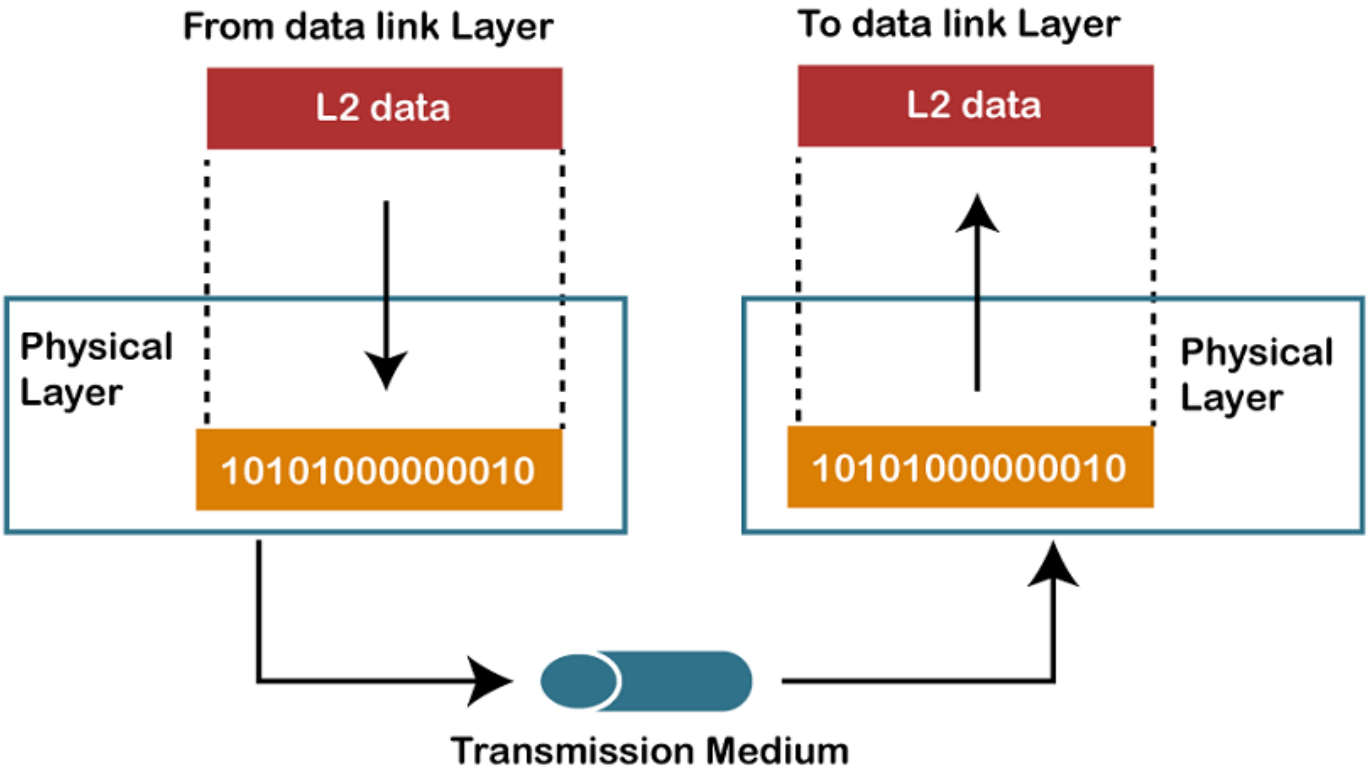
- **APPLICATION LAYER** − SDN applications reside in the Application Layer. The applications convey their needs for resources and services to the control layer through APIs.
- **CONTROL LAYER** − The Network Control Software, bundled into the Network Operating System, lies in this layer. It provides an abstract view of the underlying network infrastructure. It receives the requirements of the SDN applications and relays them to the network components.
- **INFRASTRUCTURE LAYER** − Also called the Data Plane Layer, this layer contains the actual network components. The network devices reside in this layer that shows their network capabilities through the Control to data-Plane Interface.

OSIREFERENCE MODEL

OSI stands for **Open Systems Interconnection**. It has been developed by ISO – '**International Organization of Standardization**', in the year 1984. It is 7 layer architecture with each layer having specific functionality to perform. All these 7 layers work collaboratively to transmit the data from one person to another across the globe.


Host A
Host B
Name of Unit Exchanged
7
Application
Application protocol
Application
APDU
Interface
6
Presentation
Presentation protocol
Presentation
PPDU
5
Session
Session protocol
Session
SPDU
4
Transport
Transport protocol
Transport
TPDU
Communication subnet protocol
Network layer host-router protocol
3
Network
Network
Network
Network
Packet
Data link layer host-router protocol
2
Data Link
Data Link
Data Link
Data Link
Frame
Physical layer host-router protocol
1
Physical
Physical
Physical
Physical
Bit
Internal subnet protocol

1. Physical Layer (Layer 1) :



The lowest layer of the OSI reference model is the physical layer. It is responsible for the actual physical connection between the devices. The physical layer contains information in the form of **bits.** It is responsible for transmitting individual bits from one node to the next. When receiving data, this layer will get the signal received and convert it into 0s and 1s and send them to the Data Link layer, which will put the frame back together.

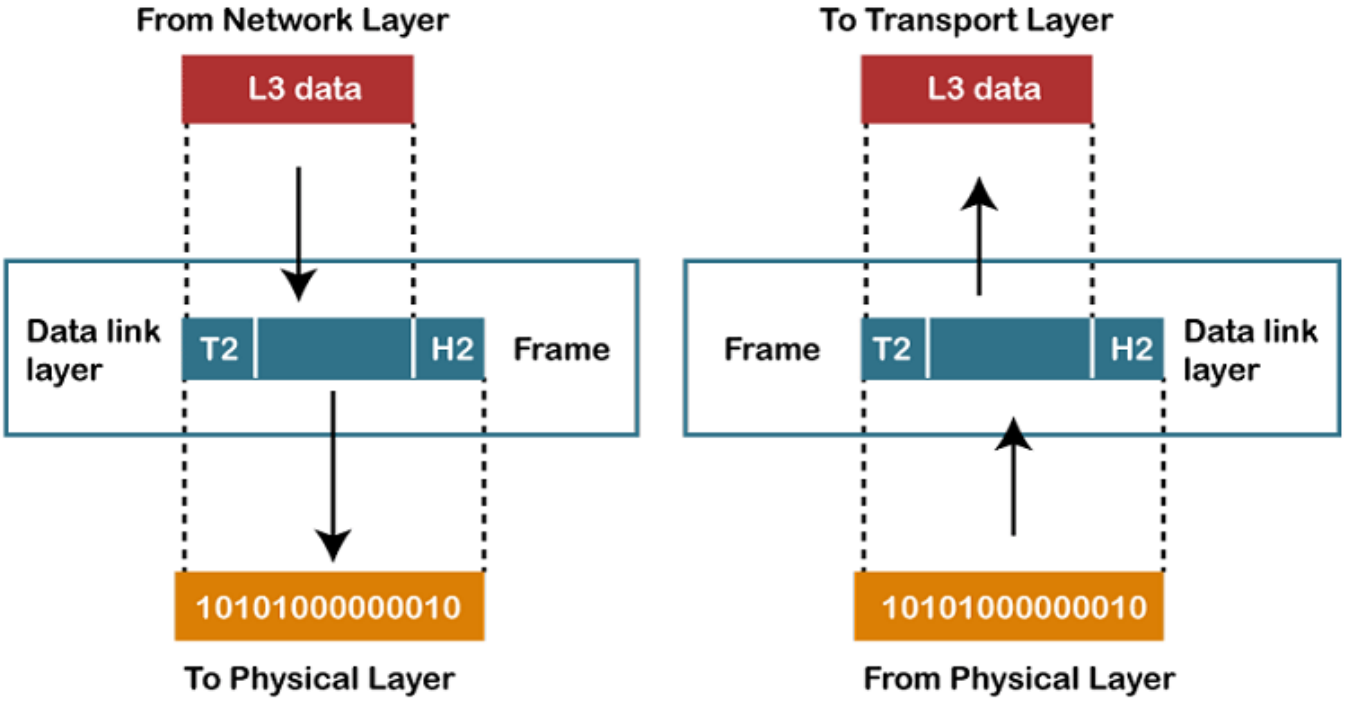
The functions of the physical layer are :

1. **Bit synchronization:** The physical layer provides the synchronization of the bits by providing a clock. This clock controls both sender and receiver thus providing synchronization at bit level.
2. **Bit rate control:** The Physical layer also defines the transmission rate i.e. the number of bits sent per second.
3. **Physical topologies:** Physical layer specifies the way in which the different, devices/nodes are arranged in a network i.e. bus, star or mesh topology.
4. **Transmission mode:** Physical layer also defines the way in which the data flows between the two connected devices. The various transmission modes possible are: Simplex, half-duplex and full-duplex.

* **Hub, Repeater, Modem, Cables are Physical Layer devices.**

** **Network Layer, Data Link Layer and Physical Layer are also known as** Lower Layers **or** Hardware Layers**.**

2. Data Link Layer (DLL) (Layer 2) :

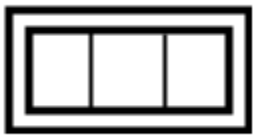


The data link layer is responsible for the node to node delivery of the message. The main function of this layer is to make sure data transfer is error-free from one node to another, over the physical layer. When a packet arrives in a network, it is the responsibility of DLL to transmit it to the Host using its MAC address.
Data Link Layer is divided into two sub layers:

1. Logical Link Control (LLC)
    - It is responsible for transferring the packets to the Network layer of the receiver that is receiving.
    - It identifies the address of the network layer protocol from the header.
    - It also provides flow control.
2. Media Access Control (MAC)
    - A Media access control layer is a link between the Logical Link Control layer and the network's physical layer.
    - It is used for transferring the packets over the network

The packet received from Network layer is further divided into frames depending on the frame size of NIC (Network Interface Card). DLL also encapsulates Sender and Receiver's MAC address in the header.

The Receiver's MAC address is obtained by placing an ARP(Address Resolution Protocol) request onto the wire asking "Who has that IP address?" and the destination host will reply with its MAC address.

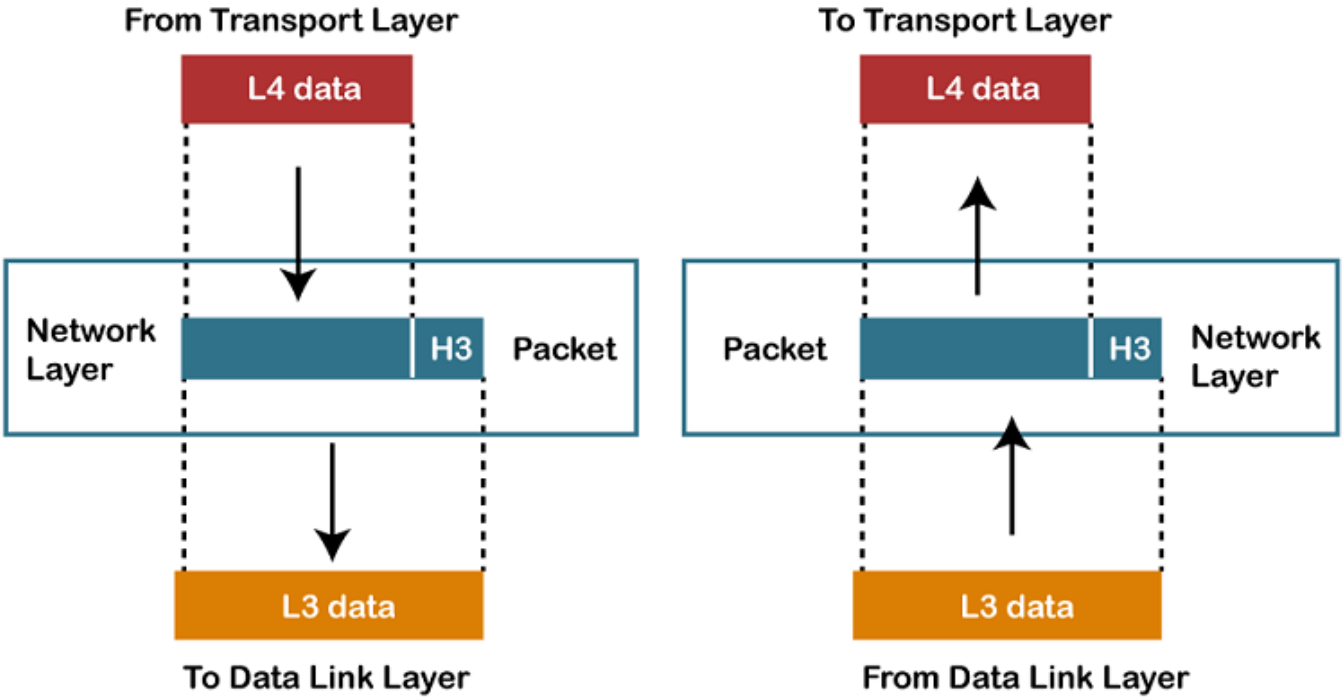


The functions of the data Link layer are :

1. **Framing:** Framing is a function of the data link layer. It provides a way for a sender to transmit a set of bits that are meaningful to the receiver. This can be accomplished by attaching special bit patterns to the beginning and end of the frame.
2. **Physical addressing:** After creating frames, Data link layer adds physical addresses (MAC address) of sender and/or receiver in the header of each frame.
3. **Error control:** Data link layer provides the mechanism of error control in which it detects and retransmits damaged or lost frames.
4. **Flow Control:** The data rate must be constant on both sides else the data may get corrupted thus, flow control coordinates that amount of data that can be sent before receiving acknowledgement.
5. **Access control:** When a single communication channel is shared by multiple devices, MAC sub-layer of data link layer helps to determine which device has control over the channel at a given time.

* ***Packet in Data Link layer is referred as*** *Frame.*
** ***Data Link layer is handled by the NIC (Network Interface Card) and device drivers of host machines.***
*** ***Switch & Bridge are Data Link Layer devices.***

3. Network Layer (Layer 3):

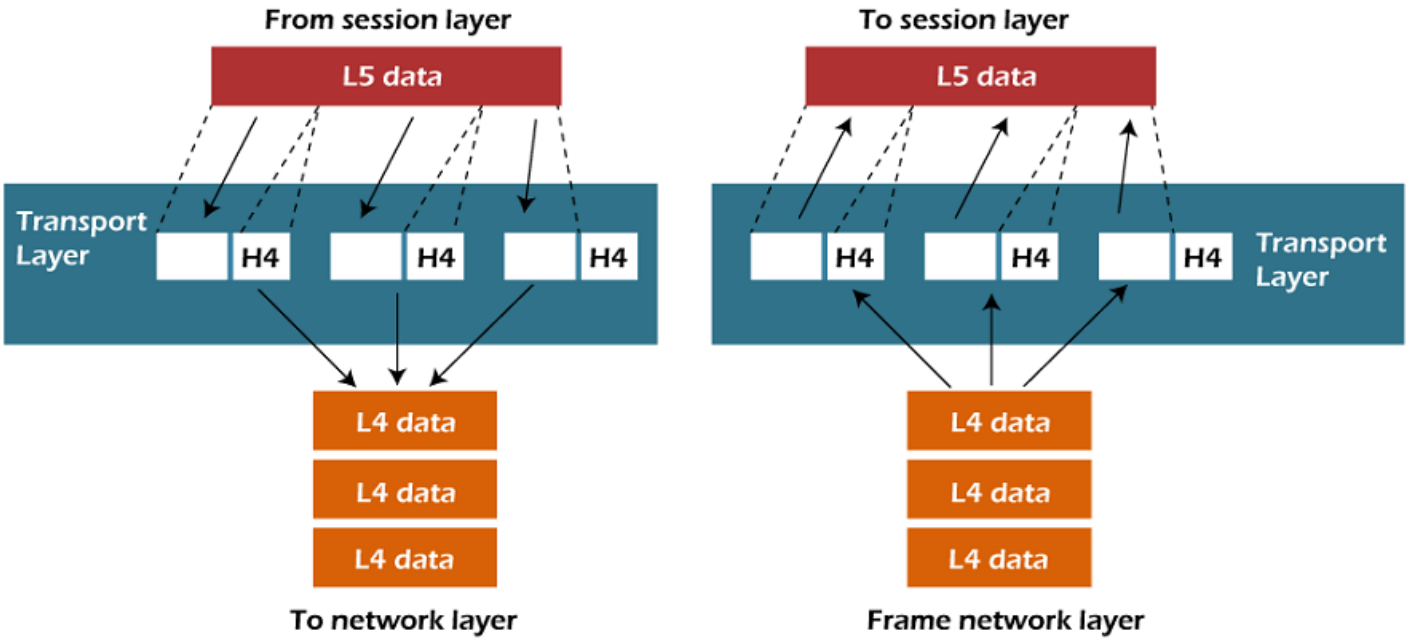


It is a layer 3 that manages device addressing, tracks the location of devices on the network. It determines the best path to move data from source to the destination based on the network conditions, the priority of service, and other factors. The Data link layer is responsible for routing and forwarding the packets. Routers are the layer 3 devices, they are specified in this layer and used to provide the routing services within an internetwork. The protocols used to route the network traffic are known as Network layer protocols. Examples of protocols are IP and Ipv6.

The functions of the Network layer are:

1. **Routing:** The network layer protocols determine which route is suitable from source to destination. This function of network layer is known as routing.
2. **Logical Addressing:** In order to identify each device on internetwork uniquely, network layer defines an addressing scheme. The sender & receiver's IP address are placed in the header by network layer. Such an address distinguishes each device uniquely and universally.

4.Transport Layer (Layer 4) :



- The Transport layer is a Layer 4 ensures that messages are transmitted in the order in which they are sent and there is no duplication of data.
- The main responsibility of the transport layer is to transfer the data completely.
- It receives the data from the upper layer and converts them into smaller units known as segments.
- This layer can be termed as an end-to-end layer as it provides a point-to-point connection between source and destination to deliver the data reliably.

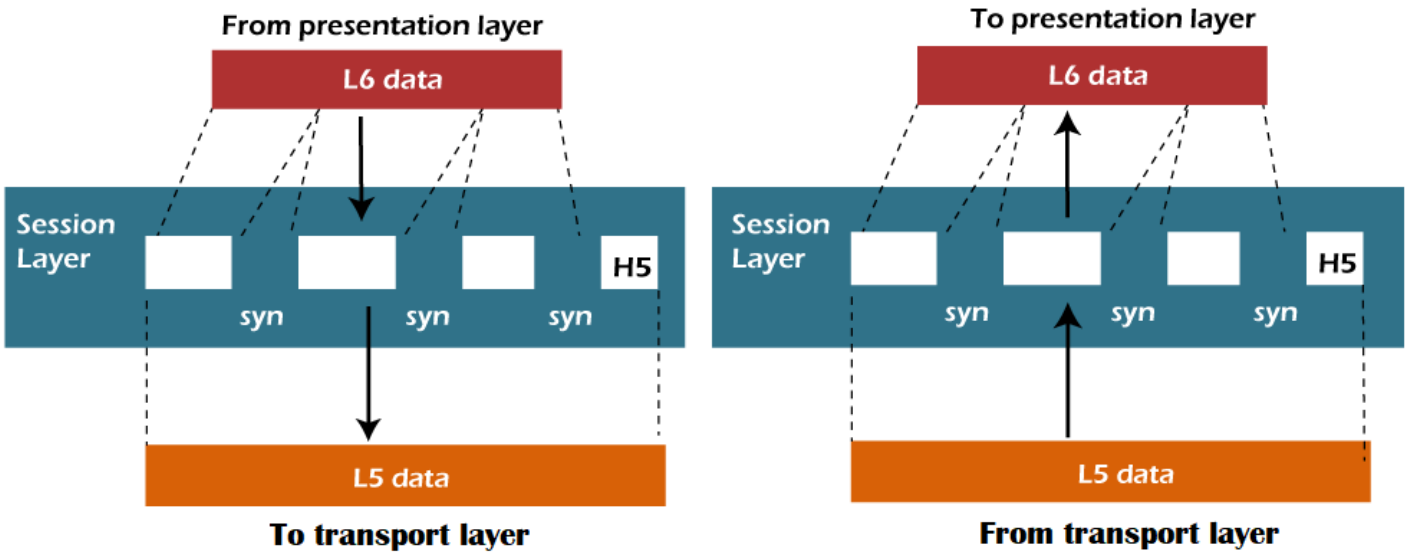
The two protocols used in this layer are:

- **Transmission Control Protocol**
  - It is a standard protocol that allows the systems to communicate over the internet.
  - It establishes and maintains a connection between hosts.
  - When data is sent over the TCP connection, then the TCP protocol divides the data into smaller units known as segments. Each segment travels over the internet using multiple routes, and they arrive in different orders at the destination. The transmission control protocol reorders the packets in the correct order at the receiving end.
- **User Datagram Protocol**
  - User Datagram Protocol is a transport layer protocol.
  - It is an unreliable transport protocol as in this case receiver does not send any acknowledgment when the packet is received, the sender does not wait for any acknowledgment. Therefore, this makes a protocol unreliable.

The functions of the transport layer are :

1. **Segmentation and Reassembly:** This layer accepts the message from the (session) layer , breaks the message into smaller units . Each of the segment produced has a header associated with it. The transport layer at the destination station reassembles the message.
2. **Service Point Addressing:** In order to deliver the message to correct process, transport layer header includes a type of address called service point address or port address. Thus by specifying this address, transport layer makes sure that the message is delivered to the correct process.

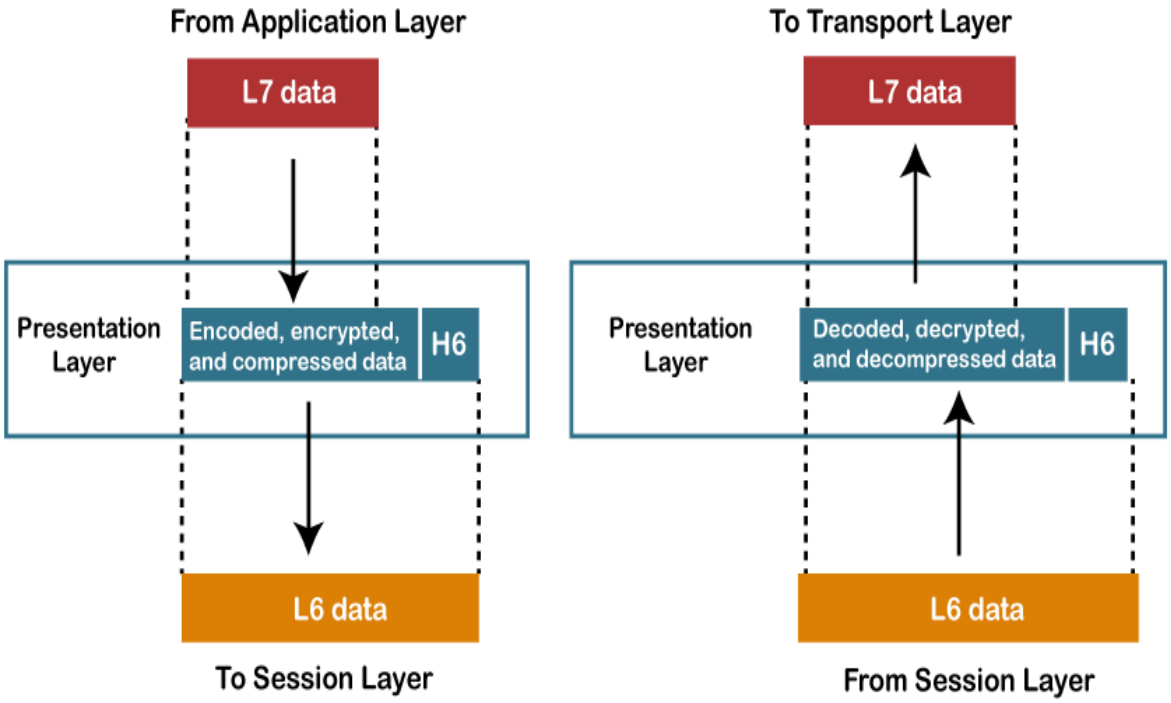
5.Session Layer (Layer 5) :



This layer is responsible for establishment of connection, maintenance of sessions, authentication and also ensures security between the communicating devices.
The functions of the session layer are :

1. **Session establishment, maintenance and termination:** The layer allows the two processes to establish, use and terminate a connection.
2. **Synchronization :** This layer allows a process to add checkpoints which are considered as synchronization points into the data. These synchronization point help to identify the error so that the data is re-synchronized properly, and ends of the messages are not cut prematurely and data loss is avoided.
3. **Dialog Controller:** The session layer allows two systems to start communication with each other in half-duplex or full-duplex.

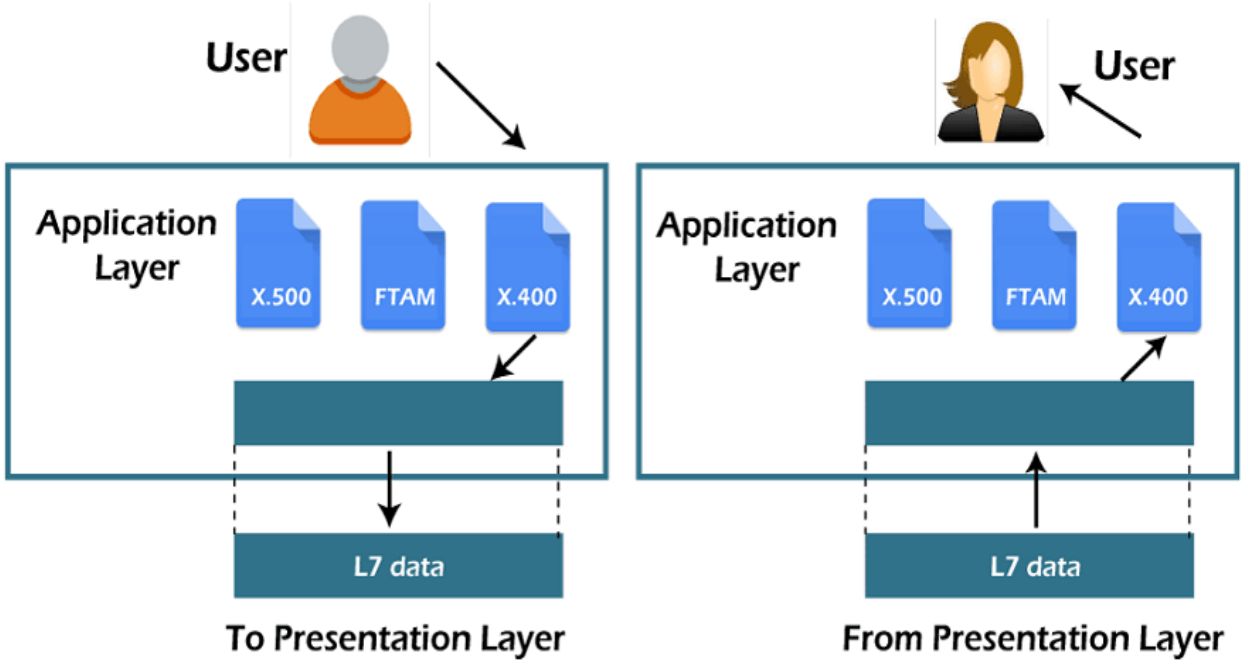
6.Presentation Layer (Layer 6) :



Presentation layer is also called the **Translation layer**. The data from the application layer is extracted here and manipulated as per the required format to transmit over the network. The functions of the presentation layer are :

1. **Translation :** For example, ASCII to EBCDIC.
2. **Encryption/ Decryption:** Data encryption translates the data into another form or code. The encrypted data is known as the cipher text and the decrypted data is known as plain text. A key value is used for encrypting as well as decrypting data.
3. **Compression:** Reduces the number of bits that need to be transmitted on the network.

7.Application Layer (Layer 7) :



At the very top of the OSI Reference Model stack of layers, we find Application layer which is implemented by the network applications. These applications produce the data, which has to be transferred over the network. This layer also serves as a window for the application services to access the network and for displaying the received information to the user.

The functions of the Application layer are :

- **File transfer, access, and management (FTAM):** An application layer allows a user to access the files in a remote computer, to retrieve the files from a computer and to manage the files in a remote computer.
- **Mail services:** An application layer provides the facility for email forwarding and storage.
- **Directory services**: An application provides the distributed database sources and is used to provide that global information about various objects.

TCP/IP REFERENCE MODEL

Prerequisite – Layers of OSI Model

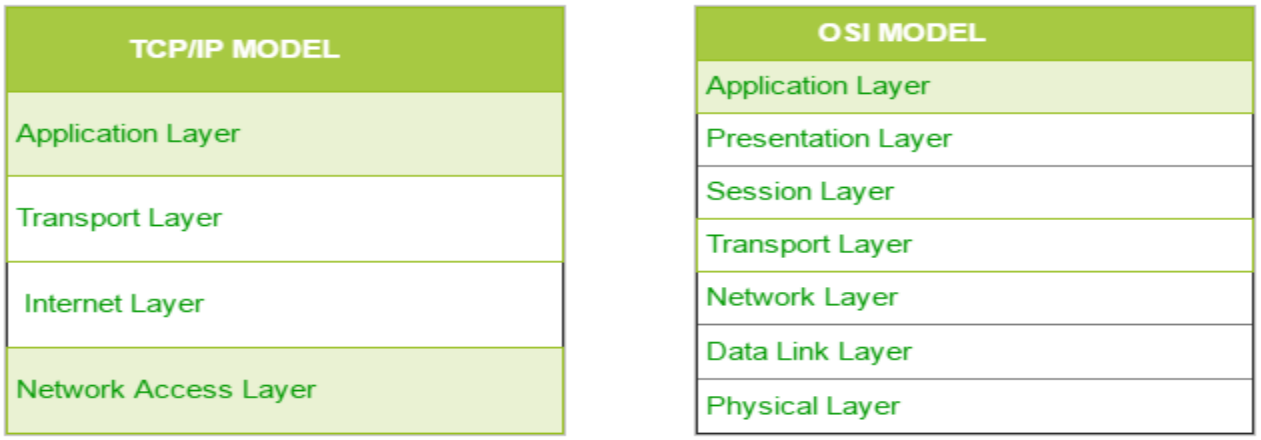
TCP/IP that is Transmission Control Protocol and Internet Protocol was developed by Department of **Defence's Project Research Agency** (ARPA, later DARPA) as a part of a research project of network interconnection to connect remote machines.

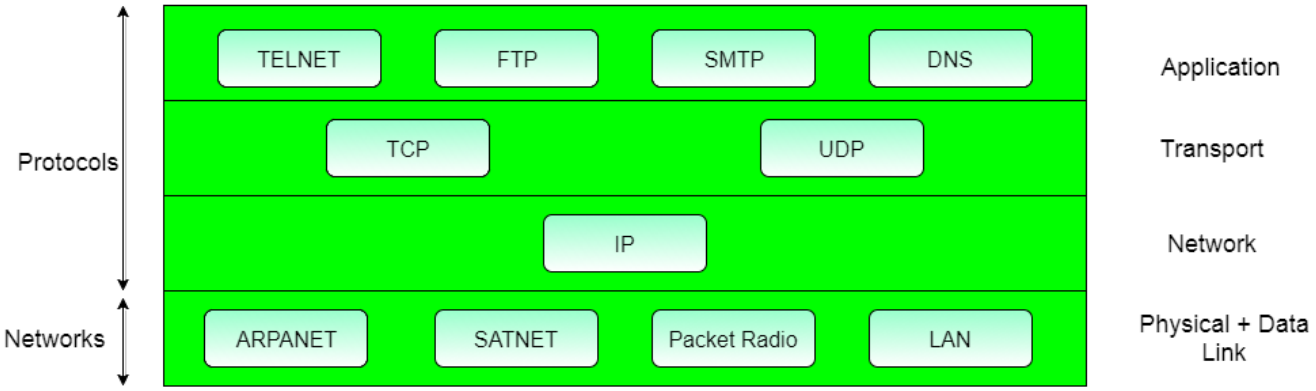
The **OSI Model** we just looked at is just a reference/logical model. It was designed to describe the functions of the communication system by dividing the communication procedure into smaller and simpler components. But when we talk about the TCP/IP model, it was designed and developed by

Department of Defense (DoD) is based on standard protocols. It stands for Transmission Control Protocol/Internet Protocol. The **TCP/IP model** is a concise version of the OSI model. It contains four layers, unlike seven layers in the OSI model. The layers are:

1. Process/Application Layer
2. Host-to-Host/Transport Layer
3. Internet Layer
4. Network Access/Link Layer

The diagrammatic comparison of the TCP/IP and OSI model is as follows :





Difference between TCP/IP and OSI Model:

TCP/IP	OSI
TCP refers to Transmission Control Protocol.	OSI refers to Open Systems Interconnection.
TCP/IP has 4 layers.	OSI has 7 layers.

TCP/IP is more reliable	OSI is less reliable
TCP/IP does not have very strict boundaries.	OSI has strict boundaries
TCP/IP follows a horizontal approach.	OSI follows a vertical approach.
TCP/IP uses both session and presentation layer in the application layer itself.	OSI uses different session and presentation layers.
TCP/IP developed protocols then model.	OSI developed model then protocol.
Transport layer in TCP/IP does not provide assurance delivery of packets.	In OSI model, transport layer provide s assurance delivery of packets.
TCP/IP model network layer only provides connection less services.	Connection less and connection oriented both services are provided by network layer in OSI model.
Protocols cannot be replaced easily in TCP/IP model.	While in OSI model, Protocols are better covered and is easy to replace with the change in technology.

The first layer is the Process layer on the behalf of the sender and Network Access layer on the behalf of the receiver. During this article, we will be talking on the behalf of the receiver.

1. Network Access Layer –

This layer corresponds to the combination of Data Link Layer and Physical Layer of the OSI model. It looks out for hardware addressing and the protocols present in this layer allows for the physical transmission of data.We just talked about ARP (Address Resolution Protocol) being a protocol of Internet layer, but there is a conflict about declaring it as a protocol of Internet Layer or Network access layer. It is described as residing in layer 3, being encapsulated by layer 2 protocols.

2. Internet Layer –

This layer parallels the functions of OSI's Network layer. It defines the protocols which are responsible for logical transmission of data over the entire network. The main protocols residing at this layer are :

1. **IP –** stands for Internet Protocol and it is responsible for delivering packets from the source host to the destination host by looking at the IP addresses in the packet headers. IP has 2 versions: IPv4 and IPv6. IPv4 is the one that most of the websites are using currently. But IPv6 is growing as the number of IPv4 addresses are limited in number when compared to the number of users.

2. **ICMP –** stands for Internet Control Message Protocol. It is encapsulated within IP datagram's and is responsible for providing hosts with information about network problems.
3. **ARP –** stands for Address Resolution Protocol. Its job is to find the hardware address of a host from a known IP address. ARP has several types: Reverse ARP, Proxy ARP, Gratuitous ARP and Inverse ARP.

3. Host-to-Host Layer –

This layer is analogous to the transport layer of the OSI model. It is responsible for end-to-end communication and error-free delivery of data. It shields the upper-layer applications from the complexities of data. The two main protocols present in this layer are :

1. **Transmission Control Protocol (TCP) –** It is known to provide reliable and error-free communication between end systems. It performs sequencing and segmentation of data. It also has acknowledgment feature and controls the flow of the data through flow control mechanism. It is a very effective protocol but has a lot of overhead due to such features. Increased overhead leads to increased cost.
2. **User Datagram Protocol (UDP) –** On the other hand does not provide any such features. It is the go-to protocol if your application does not require reliable transport as it is very cost-effective. Unlike TCP, which is connection-oriented protocol, UDP is connectionless.

4. Application Layer –

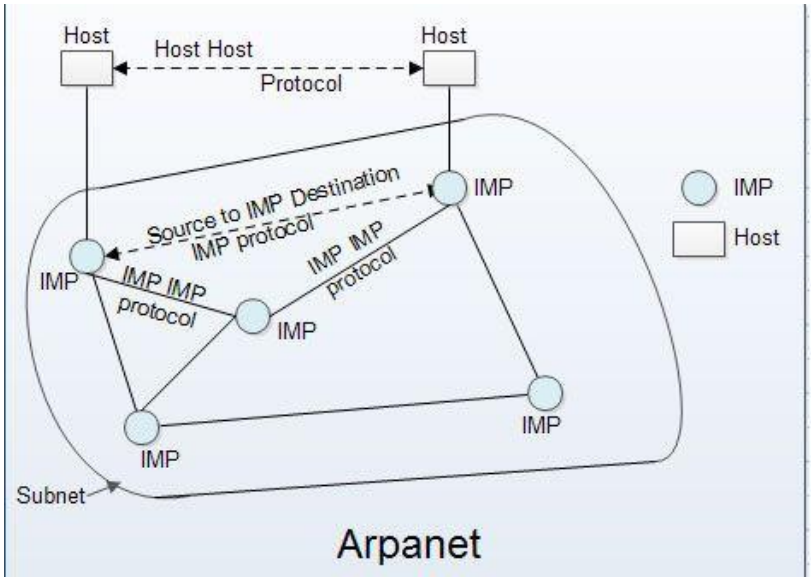
This layer performs the functions of top three layers of the OSI model: Application, Presentation and Session Layer. It is responsible for node-to-node communication and controls user-interface specifications. Some of the protocols present in this layer are: HTTP, HTTPS, FTP, TFTP, Telnet, SSH, SMTP, SNMP, NTP, DNS, DHCP, NFS, X Window, LPD. Have a look at Protocols in Application Layer for some information about these protocols. Protocols other than those present in the linked article are:

1. **HTTP and HTTPS –** HTTP stands for Hypertext transfer protocol. It is used by the World Wide Web to manage communications between web browsers and servers. HTTPS stands for HTTP-Secure. It is a combination of HTTP with SSL(Secure Socket Layer). It is efficient in cases where the browser need to fill out forms, sign in, authenticate and carry out bank transactions.
2. **SSH –** SSH stands for Secure Shell. It is a terminal emulations software similar to Telnet. The reason SSH is more preferred is because of its ability to maintain the encrypted connection. It sets up a secure session over a TCP/IP connection.
3. **NTP –** NTP stands for Network Time Protocol. It is used to synchronize the clocks on our computer to one standard time source. It is very useful in situations like bank transactions. Assume the following situation without the presence of NTP. Suppose you carry out a transaction, where your computer reads the time at 2:30 PM while the server records it at 2:28 PM. The server can crash very badly if it's out of sync.

ARPANET :

ARPANET stands for **Advanced Research Projects Agency NET**. ARPANET was first network which consisted of distributed control. It was first to implement TCP/IP protocols. It was basically beginning of Internet with use of these technologies. It was designed with a basic idea in mind that was to communicate with scientific users among an institute or university.

History of ARPANET: ARPANET was introduced in the year 1969 by Advanced Research Projects Agency (ARPA) of US Department of Defence. It was established using a bunch of PCs at various colleges and sharing of information and messages was done. It was for playing as long separation diversions and individuals were asked to share their perspectives. In the year 1980, ARPANET was handed over to different military network, Defence Data Network.



Characteristics of ARPANET :

1. It is basically a type of WAN.
2. It used concept of Packet Switching Network.
3. It used Interface Message Processors(IMPs) for sub-netting.
4. ARPANETs software was split into two parts- a host and a subnet.

Advantages of ARPANET :

- ARPANET was designed to service even in a Nuclear Attack.
- It was used for collaborations through E-mails.
- It created an advancement in transfer of important files and data of defense.

Limitations of ARPANET :

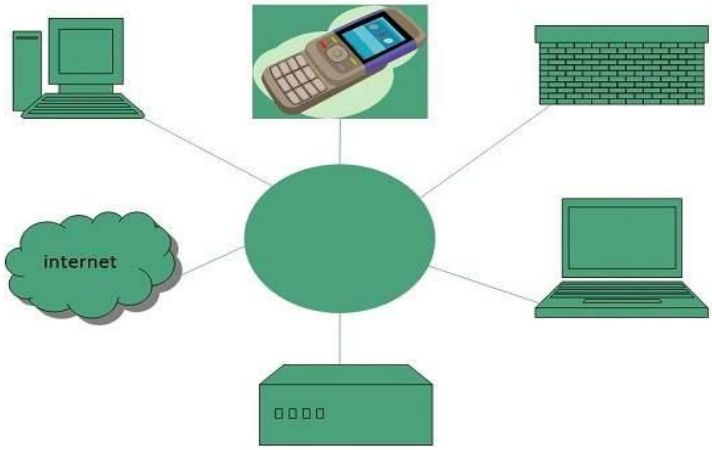
- Increased number of LAN connections resulted in difficulty handling.
- It was unable to cope-up with advancement in technology.

INTERNET:

Internet is a global network that connects billions of computers across the world with each other and to the World Wide Web. It uses standard internet protocol suite (TCP/IP) to connect billions of computer users worldwide. It is set up by using cables such as optical fibers and other wireless and networking technologies. At present, internet is the fastest mean of sending or exchanging information and data between computers across the world.

Internet is defined as an Information super Highway, to access information over the web. However, It can be defined in many ways as follows:

- Internet is a world-wide global system of interconnected computer networks.
- Internet uses the standard Internet Protocol (TCP/IP).
- Every computer in internet is identified by a unique IP address.
- IP Address is a unique set of numbers (such as 110.22.33.114) which identifies a computer location.
- A special computer DNS (Domain Name Server) is used to give name to the IP Address so that user can locate a computer by a name.
- For example, a DNS server will resolve a name **http://www.tutorialspoint.com** to a particular IP address to uniquely identify the computer on which this website is hosted.
- Internet is accessible to every user all over the world.



Evolution

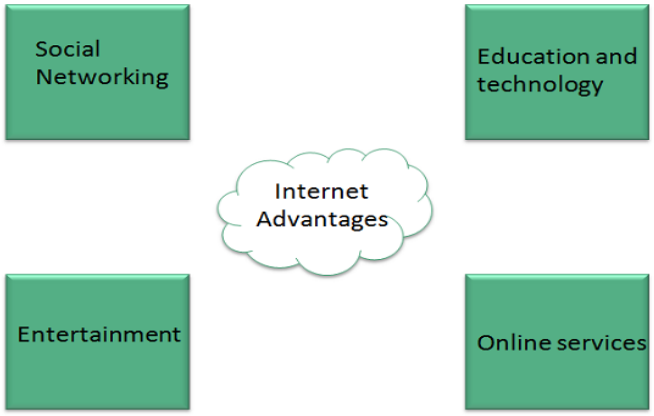
The concept of Internet was originated in 1969 and has undergone several technological & Infrastructural changes as discussed below:

- The origin of Internet devised from the concept of **Advanced Research Project Agency Network (ARPANET).**
- **ARPANET** was developed by United States Department of Defense.

- Basic purpose of ARPANET was to provide communication among the various bodies of government.
- Initially, there were only four nodes, formally called **Hosts.**
- In 1972, the **ARPANET** spread over the globe with 23 nodes located at different countries and thus became known as **Internet.**
- By the time, with invention of new technologies such as TCP/IP protocols, DNS, WWW, browsers, scripting languages etc.,Internet provided a medium to publish and access information over the web.

Advantages

Internet covers almost every aspect of life, one can think of. Here, we will discuss some of the advantages of Internet:



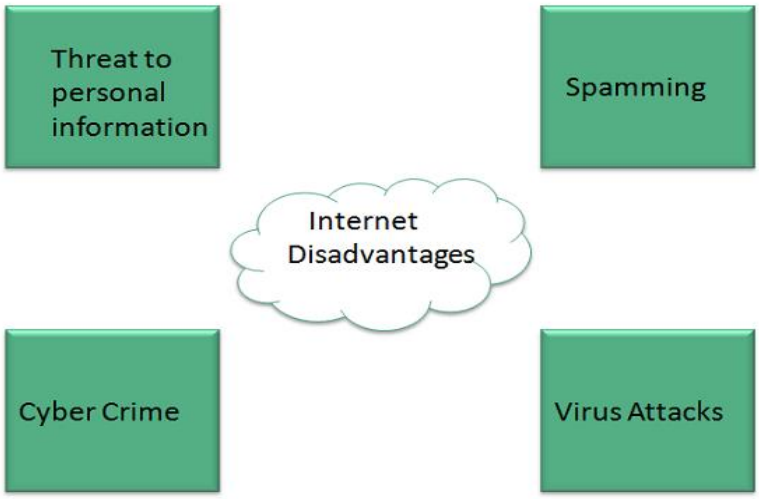
- Internet allows us to communicate with the people sitting at remote locations. There are various apps available on the wed that uses Internet as a medium for communication. One can find various social networking sites such as:
    - Facebook
    - Twitter
    - Yahoo
    - Google+
    - Flickr
    - Orkut
- One can surf for any kind of information over the internet. Information regarding various topics such as Technology, Health & Science, Social Studies, Geographical Information, Information Technology, Products etc can be surfed with help of a search engine.
- Apart from communication and source of information, internet also serves a medium for entertainment. Following are the various modes for entertainment over internet.
    - Online Television

- Online Games
- Songs
- Videos
- Social Networking Apps

- Internet allows us to use many services like:
  - Internet Banking
  - Matrimonial Services
  - Online Shopping
  - Online Ticket Booking
  - Online Bill Payment
  - Data Sharing
  - E-mail
- Internet provides concept of **electronic commerce**, that allows the business deals to be conducted on electronic systems

Disadvantages:

However, Internet has proved to be a powerful source of information in almost every field, yet there exists many disadvantages discussed below:

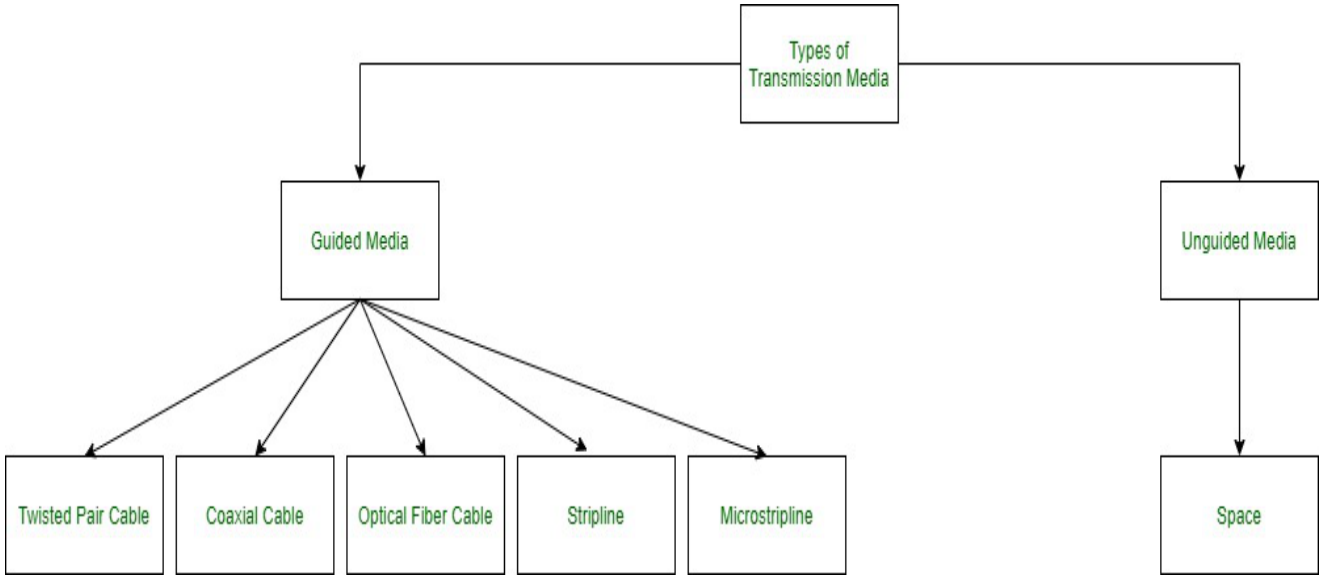


- There are always chances to loose personal information such as name, address, credit card number. Therefore, one should be very careful while sharing such information. One should use credit cards only through authenticated sites.
- Another disadvantage is the **Spamming**. Spamming corresponds to the unwanted e-mails in bulk. These e-mails serve no purpose and lead to obstruction of entire system.
- **Virus** can easily be spread to the computers connected to internet. Such virus attacks may cause your system to crash or your important data may get deleted.

- Also a biggest threat on internet is pornography. There are many pornographic sites that can be found, letting your children to use internet which indirectly affects the children healthy mental life.
- There are various websites that do not provide the authenticated information. This leads to misconception among many people.

Types of Transmission Media:

In data communication terminology, a transmission medium is a physical path between the transmitter and the receiver i.e it is the channel through which data is sent from one place to another. Transmission Media is broadly classified into the following types:

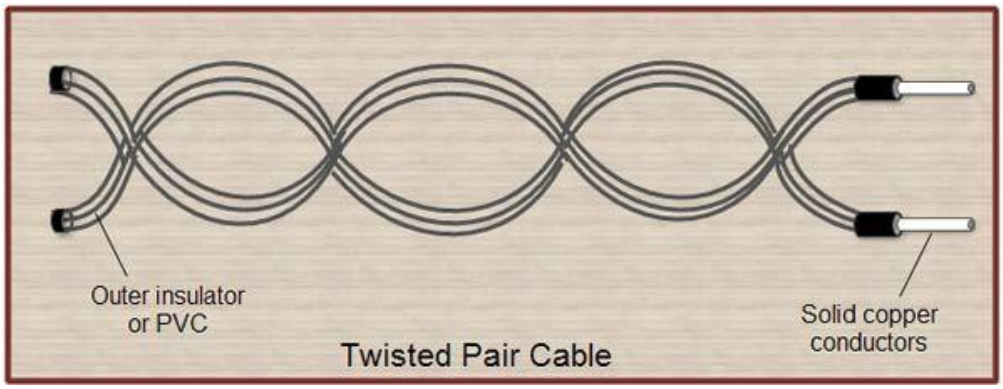


1. Guided Media: It is also referred to as Wired or Bounded transmission media. Signals being transmitted are directed and confined in a narrow pathway by using physical links. Features:

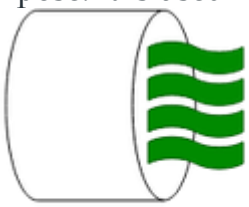
- High Speed
- Secure
- Used for comparatively shorter distances

There are 3 major types of Guided Media:

(i) Twisted Pair Cable – It consists of 2 separately insulated conductor wires wound about each other. Generally, several such pairs are bundled together in a protective sheath. They are the most widely used Transmission Media. Twisted Pair is of two types:



- **Unshielded Twisted Pair (UTP):** This type of cable has the ability to block interference and does not depend on a physical shield for this purpose. It is used for telephonic applications.



Unshielded Twisted Pair

Advantages:

⇢ Least expensive

⇢ Easy to install

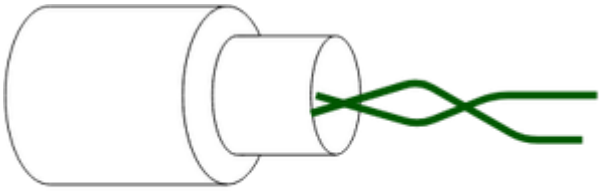
⇢ High-speed capacity

⇢ Susceptible to external interference

⇢ Lower capacity and performance in comparison to STP

⇢ Short distance transmission due to attenuation

- **Shielded Twisted Pair (STP):** This type of cable consists of a special jacket to block external interference. It is used in fast-data-rate Ethernet and in voice and data channels of telephone lines.



Shielded Twisted Pair

Advantages:

⇢ Better performance at a higher data rate in comparison to UTP

⇢ Eliminates crosstalk

⇢ Comparatively faster

⇢ Comparatively difficult to install and manufacture

⇢ More expensive

⇢ Bulky

(ii) Coaxial Cable – It has an outer plastic covering containing 2 parallel conductors each having a separate insulated protection cover. The coaxial cable transmits information in two modes: Baseband mode(dedicated cable bandwidth) and Broadband mode(cable bandwidth is split into separate ranges). Cable TVs and analog television networks widely use Coaxial cables.

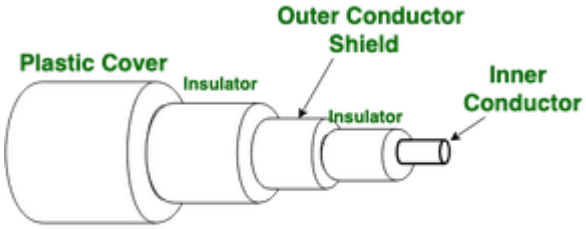


Figure of Optical Coaxial Cable

Advantages:

- High Bandwidth
- Better noise Immunity
- Easy to install and expand
- Inexpensive

Disadvantages:

- Single cable failure can disrupt the entire network

(iii) Optical Fibre Cable – It uses the concept of reflection of light through a core made up of glass or plastic. The core is surrounded by a less dense glass or plastic covering called the cladding. It is used for the transmission of large volumes of data.

The cable can be unidirectional or bidirectional. The WDM (Wavelength Division Multiplexer) supports two modes, namely unidirectional and bidirectional mode.

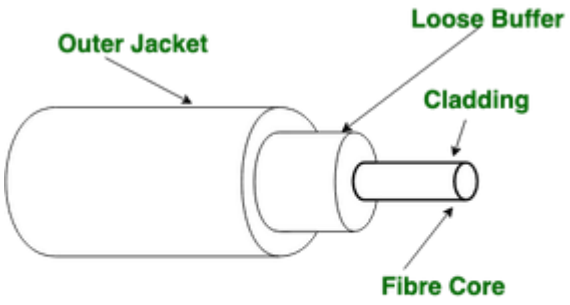


Figure of Optical Fibre Cable

Advantages:

- Increased capacity and bandwidth
- Lightweight
- Less signal attenuation
- Immunity to electromagnetic interference

- Resistance to corrosive materials

Disadvantages:

- Difficult to install and maintain
- High cost
- Fragile

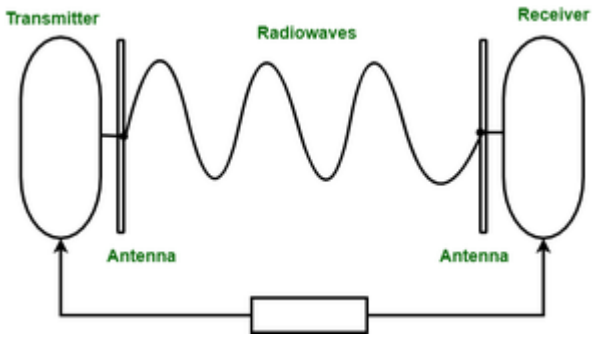
Unguided Media: It is also referred to as Wireless or Unbounded transmission media. No physical medium is required for the transmission of electromagnetic signals.

Features:

- The signal is broadcasted through air
- Less Secure
- Used for larger distances

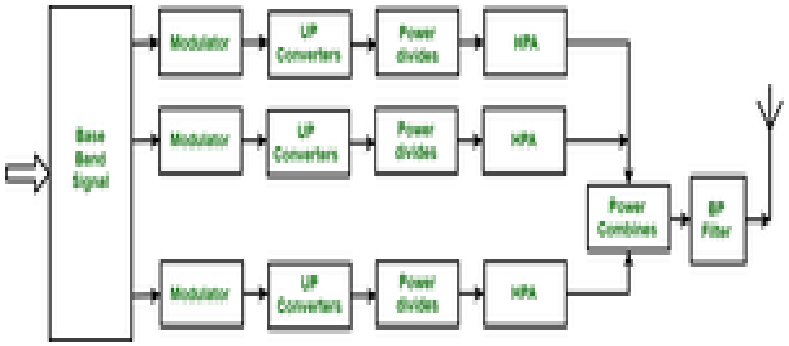
There are 3 types of Signals transmitted through unguided media:

(i)Radiowaves-These are easy to generate and can penetrate through buildings. The sending and receiving antennas need not be aligned. Frequency Range:3KHz – 1GHz. AM and FM radios and cordless phones use Radiowaves for transmission.

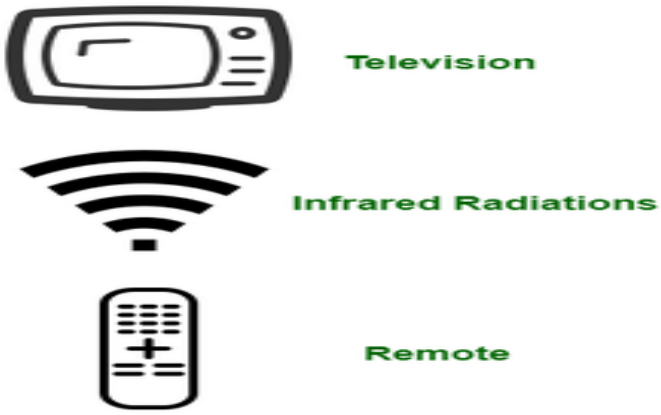


Further Categorized as (i) Terrestrial and (ii) Satellite.

(ii)Microwaves–It is a line of sight transmission i.e. the sending and receiving antennas need to be properly aligned with each other. The distance covered by the signal is directly proportional to the height of the antenna. Frequency Range:1GHz – 300GHz. These are majorly used for mobile phone communication and television distribution.



(iii)Infrared: Infrared waves are used for very short distance communication. They cannot penetrate through obstacles. This prevents interference between systems. Frequency Range:300GHz – 400THz. It is used in TV remotes, wireless mouse, keyboard, printer, etc.

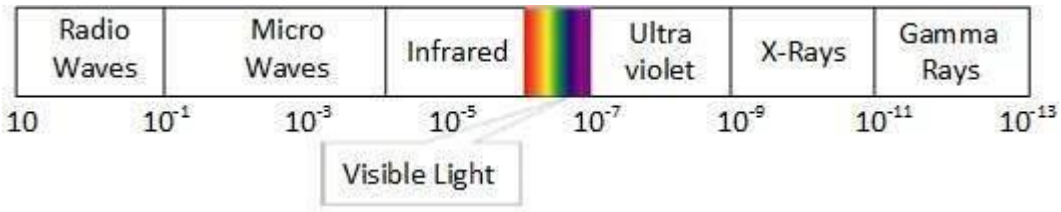


Wireless transmission:

Wireless transmission is a form of unguided media. Wireless communication involves no physical link established between two or more devices, communicating wirelessly. Wireless signals are spread over in the air and are received and interpreted by appropriate antennas.

When an antenna is attached to electrical circuit of a computer or wireless device, it converts the digital data into wireless signals and spread all over within its frequency range. The receptor on the other end receives these signals and converts them back to digital data.

A little part of electromagnetic spectrum can be used for wireless transmission.

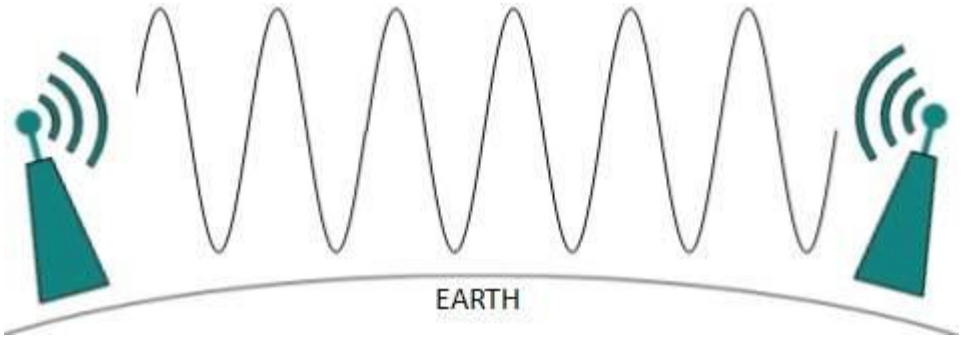


Radio Transmission

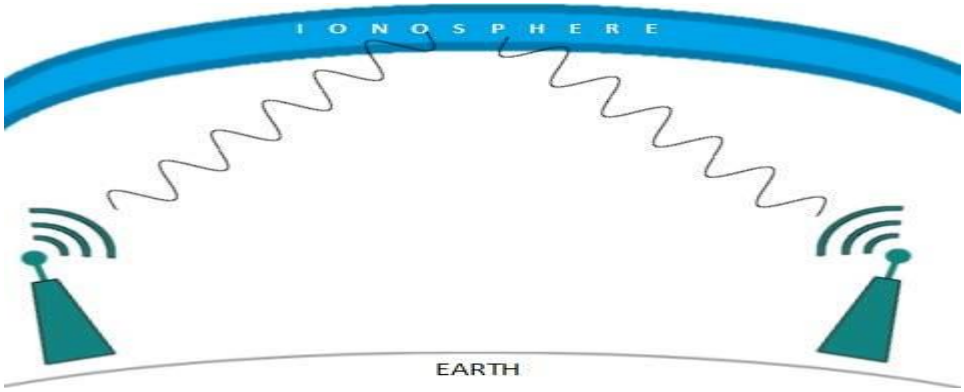
Radio frequency is easier to generate and because of its large wavelength it can penetrate through walls and structures alike. Radio waves can have wavelength from 1 mm – 100,000 km and have frequency ranging from 3 Hz (Extremely Low Frequency) to 300 GHz (Extremely High Frequency). Radio frequencies are sub-divided into six bands.

Radio waves at lower frequencies can travel through walls whereas higher RF can travel in straight line and bounce back. The power of low frequency waves decreases sharply as they cover long distance. High frequency radio waves have more power.

Lower frequencies such as VLF, LF, MF bands can travel on the ground up to 1000 kilometers, over the earth's surface.



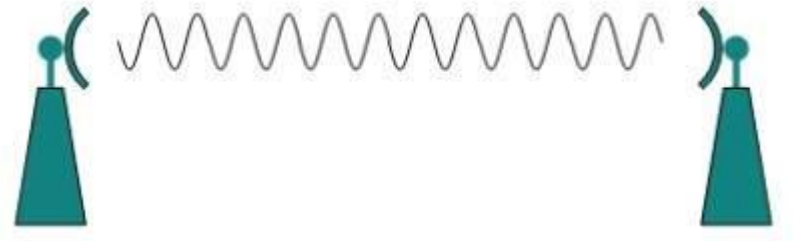
Radio waves of high frequencies are prone to be absorbed by rain and other obstacles. They use Ionosphere of earth atmosphere. High frequency radio waves such as HF and VHF bands are spread upwards. When they reach Ionosphere, they are refracted back to the earth.



Microwave Transmission

Electromagnetic waves above 100 MHz tend to travel in a straight line and signals over them can be sent by beaming those waves towards one particular station. Because Microwaves travels in straight lines, both sender and receiver must be aligned to be strictly in line-of-sight.

Microwaves can have wavelength ranging from 1 mm – 1 meter and frequency ranging from 300 MHz to 300 GHz.



Microwave antennas concentrate the waves making a beam of it. As shown in picture above, multiple antennas can be aligned to reach farther. Microwaves have higher frequencies and do not penetrate wall like obstacles.

Microwave transmission depends highly upon the weather conditions and the frequency it is using.

Infrared Transmission

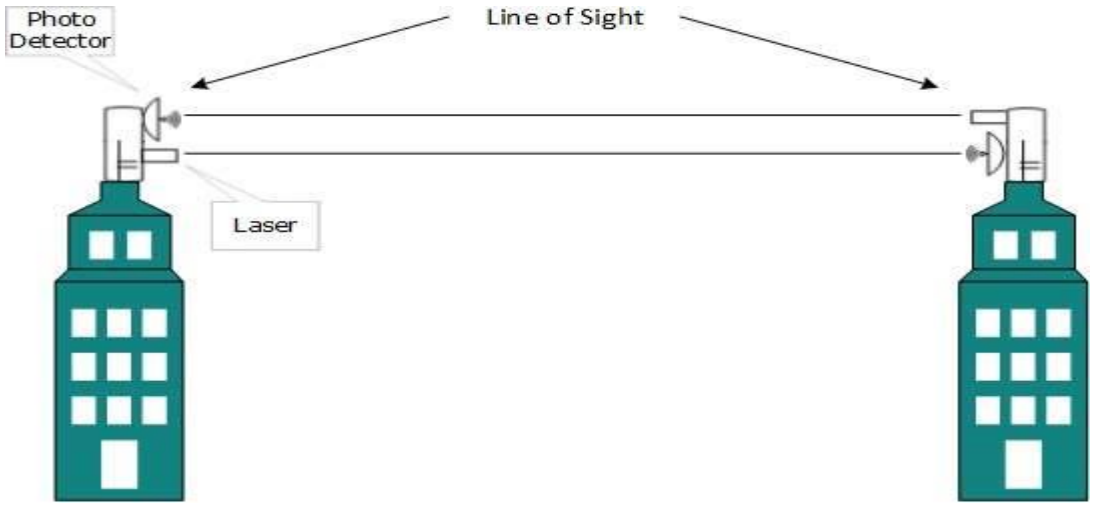
Infrared wave lies in between visible light spectrum and microwaves. It has wavelength of 700-nm to 1-mm and frequency ranges from 300-GHz to 430-THz.

Infrared wave is used for very short range communication purposes such as television and it's remote. Infrared travels in a straight line hence it is directional by nature. Because of high frequency range, Infrared cannot cross wall-like obstacles.

Light Transmission

Highest most electromagnetic spectrum which can be used for data transmission is light or optical signaling. This is achieved by means of LASER.

Because of frequency light uses, it tends to travel strictly in straight line.Hence the sender and receiver must be in the line-of-sight. Because laser transmission is unidirectional, at both ends of communication the laser and the photo-detector needs to be installed. Laser beam is generally 1mm wide hence it is a work of precision to align two far receptors each pointing to lasers source.



Laser works as Tx (transmitter) and photo-detectors works as Rx (receiver).

Lasers cannot penetrate obstacles such as walls, rain, and thick fog. Additionally, laser beam is distorted by wind, atmosphere temperature, or variation in temperature in the path.

Laser is safe for data transmission as it is very difficult to tap 1mm wide laser without interrupting the communication channel.

Data Link Layer

The Data Link Layer breaks the bit stream into discrete frames and computes the checksum for each frame. When a Frame arrives at the destination, the checksum is recomputed. If the newly computed checksum is different from one computed contained in the frame, the data link layer knows that an error has occurred and takes steps to deal with it.

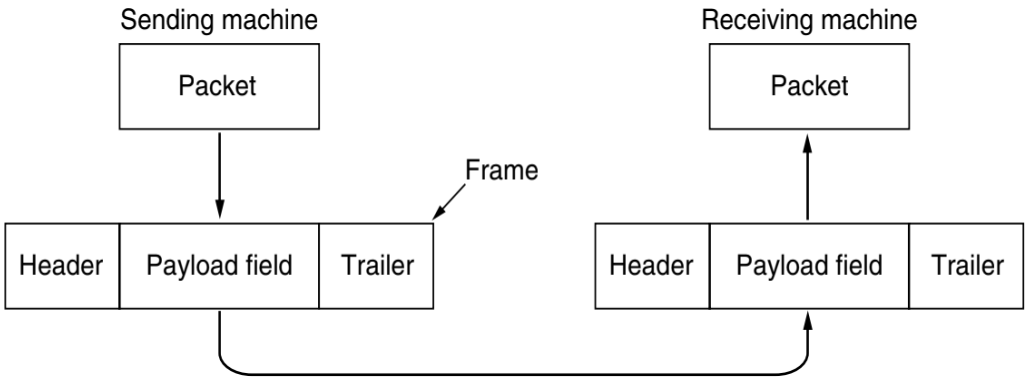
DATA LINK LAYER DESIGN ISSUES

The data link layer uses the services of the physical layer to send and receive bits over communication channels. It has a number of functions, including:

- Providing a well-defined service interface to the network layer.
- Dealing with transmission errors.
- Regulating the flow of data so that slow receivers are not swamped by fast senders.

To accomplish these goals, the data link layer takes the packets it gets from the network layer and encapsulates them into frames for transmission. Each frame contains a frame header, a payload field for holding the packet, and a frame trailer, as illustrated in Fig. 2-1. Frame management forms the heart of what the data link layer does. In the following sections we will examine all the above mentioned issues in detail.

Figure 2-1. Relationship between packets and frames.



Services Provided to the Network Layer

The principal service is transferring data from the network layer on the source machine to the network layer on the destination machine. On the source machine is an entity, call it a process, in the network layer that hands some bits to the data link layer for transmission to the destination. The job of the data link layer is to transmit the bits to the destination machine so they can be handed over to the net-work layer there, as shown in Fig. 2-2(a). The actual transmission follows the path of Fig. 2-2(b), but it is easier to think in terms of two data link layer processes communicating using a data link protocol. For this reason, we will implicitly use the model of Fig. 2-2(a) throughout this chapter.

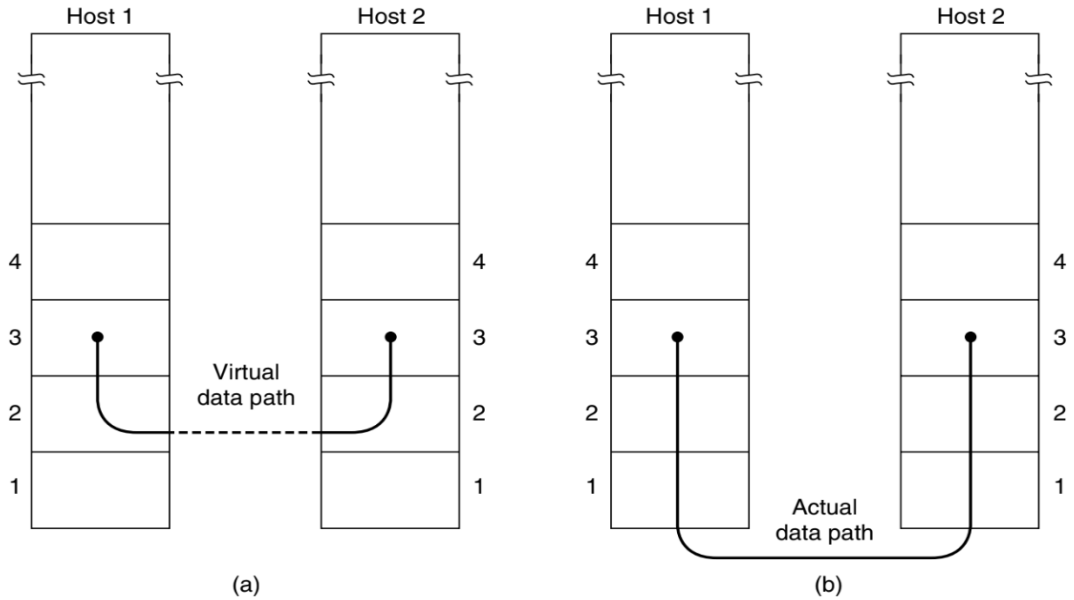


Figure 2-2. (a) Virtual communication. (b) Actual communication.

The data link layer can be designed to offer various services. The actual services that are offered vary from protocol to protocol. Three reasonable possibilities that we will consider inturn are:

1. Unacknowledged connectionless service.
   Ethernet is a good example of a data link layer that provides this class of service. This service is appropriate when the error rate is very low, It is also appropriate for real-time traffic, such as voice, in which late data are worse than bad data.

2. Acknowledged connectionless service.
   This service is useful over unreliable channels, such as wireless systems. 802.11 (WiFi) is a good example of this class of service.

3. Acknowledged connection-oriented It is appropriate over long, unreliable links such as a satellite channel or a long-distancetelephone circuit.

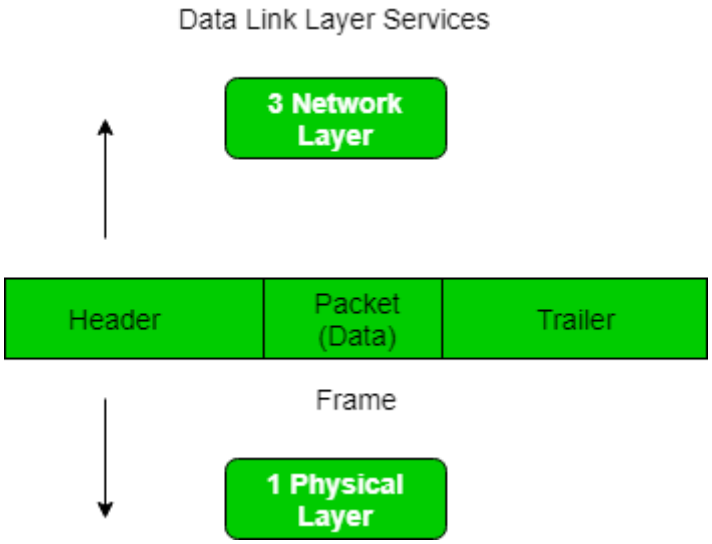
The usual approach is for the data link layer to break up the bit stream into discrete frames, compute a short token called a checksum for each frame, and include the checksum in the frame when it is transmitted. When a frame arrives at the destination, the checksum is recomputed. If the newly computed checksum is different from the one contained in the frame, the data link layer knows that an

error has occurred and takes steps to deal with it (e.g., discarding the bad frame and possibly also sending back an error report).

Breaking up the bit stream into frames is more difficult than it at first appears. A good design must make it easy for a receiver to find the start of new frames while using little of the channel bandwidth. We will look at four methods:

FRAMMING:

Framing is a function of the data link layer. It provides a way for a sender to transmit a set of bits that are meaningful to the receiver.



Problems in Framing –

- ✓ **Detecting start of the frame:** When a frame is transmitted, every station must be able to detect it. Station detect frames by looking out for special sequence of bits that marks the beginning of the frame i.e. SFD (Starting Frame Delimeter).
- ✓ **How do station detect a frame:** Every station listen to link for SFD pattern through a sequential circuit. If SFD is detected, sequential circuit alerts station. Station checks destination address to accept or reject frame.
- ✓ **Detecting end of frame:** When to stop reading the frame.

Types of framing – There are two types of framing:

1. Fixed size – The frame is of fixed size and there is no need to provide boundaries to the frame, length of the frame itself acts as delimiter.

- ✓ **Drawback:** It suffers from internal fragmentation if data size is less than frame size
- ✓ **Solution:** Padding

2. Variable size – In this there is need to define end of frame as well as beginning of next frame to distinguish. This can be done in two ways:

1. **Length field** – We can introduce a length field in the frame to indicate the length of the frame. Used in **Ethernet (802.3)**. The problem with this is that sometimes the length field might get corrupted.

End Delimeter (ED) – We can introduce an ED(pattern) to indicate the end of the frame. Used in **Token Ring**. The problem with this is that ED can occur in the data. This can be solved by:

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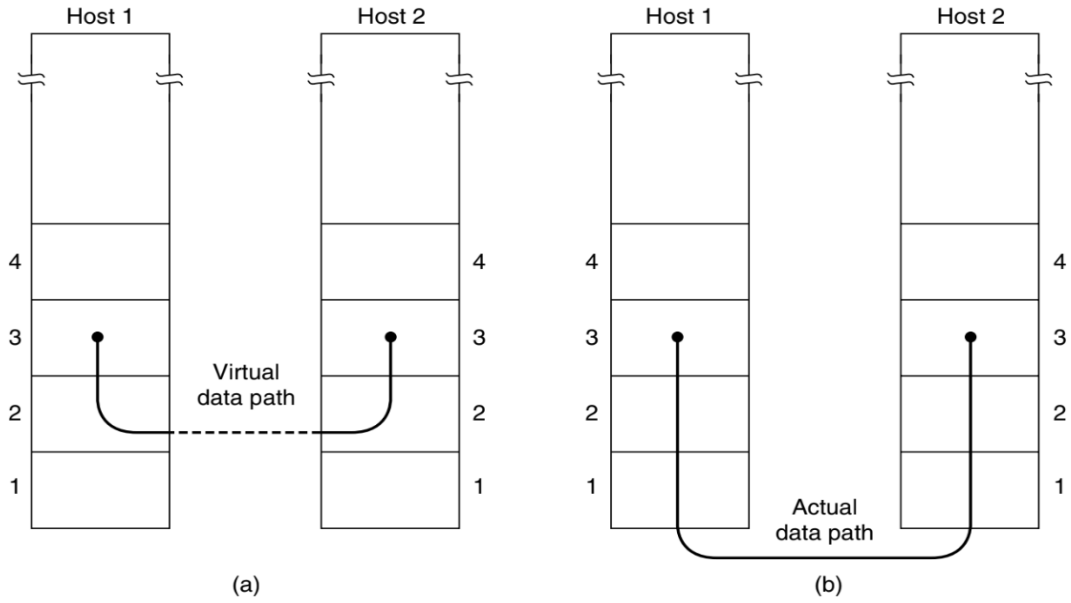

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Breaking up the bit stream into frames is more difficult than it at first appears. A good design must make it easy for a receiver to find the start of new frames while using little of the channel bandwidth. We will look at four methods:

FRAMING METHODS

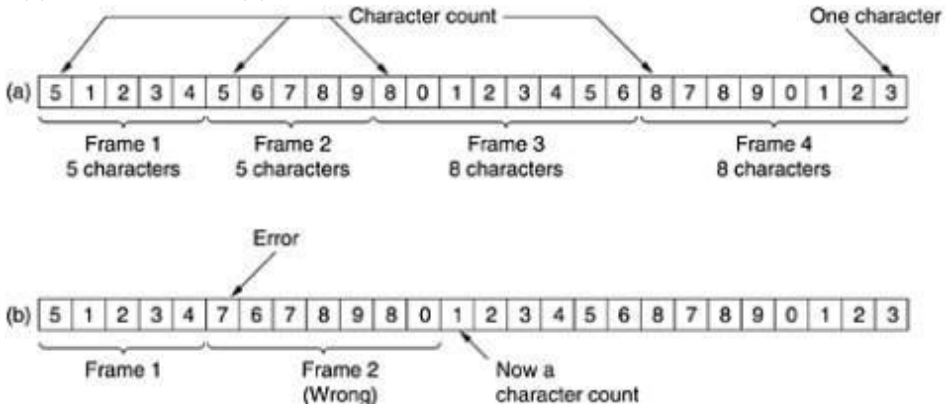
1. CHARATER COUNT METHOD
2. STARTING AND ENDING CHARACTERS, WITH CHARATER STUFFING
3. STARTING AND ENDING FLAGS, WITH BIT STUFFING

CHARATER COUNT METHOD:

In this method a field in the header will be used to specify the number of CHARACTERS in the frame. When data link layer at the destination sees the character count, it knows how many characters follow and hence where the end of the frame is.

The trouble with this algorithm is that the count can be garbed by a transmission error resulting the destination will get out of synchronization and will be unable to locate the start of the next frame. There is no way of telling where the next frame starts. For this reason this method is rarely used.

A Character Stream (a) Without errors (b) With one error



CHARATER STUFFING METHOD: In this method each frame willstart with a FLAG and ends with a FLAG.

The starting flag is **DLE STX ---- Data Link Escape Start of Text**

The ending flag is **DLE ETX ---------- Data link Escape End of Text.**

Ex 1. The given Data **ABRFCXDGJHKK12435ASBGXRR**

The Data will be sent as **DLE STX ABRFCXDGJHKK12435ASBGXRR DLE ETX**

Ex 2. The given Data **ASHGTRDXZBNHG DLE STX %$#54378**

The data will be sent as **DLE STX ASHGTRDXZBNHG DLE STX %$#54378 DLEETX**

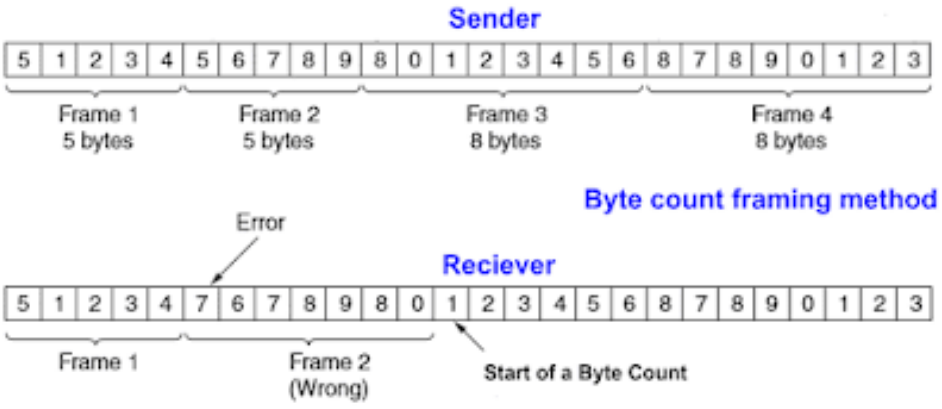
Dis Adv:

1. 24 bits are unnecessarily stuffed.
2. Transmission delay.

BIT STUFFING METHOD

Each frame begins and ends with a special bit pattern, 01111110 (in fact, a flag byte). Whenever the sender's data link layer encounters five consecutive 1s in the data, it automatically stuffs a 0 bit into the outgoing bit stream. This bit stuffing is analogous to byte stuffing, in which an escape byte is stuffed into the outgoing character stream before a flag byte in the data. When the receiver sees five consecutive incoming 1 bits, followed by a 0 bit, it automatically destuffs (i.e.,deletes) the 0 bit. Just as byte stuffing is completely transparent to the network layer in both computers, so is bit stuffing. If the user data contain the flag pattern, 01111110, this flag is transmitted as 011111010 but stored in the receiver's memory as 01111110. Figure 3-6 gives an example of bit stuffing.

1.



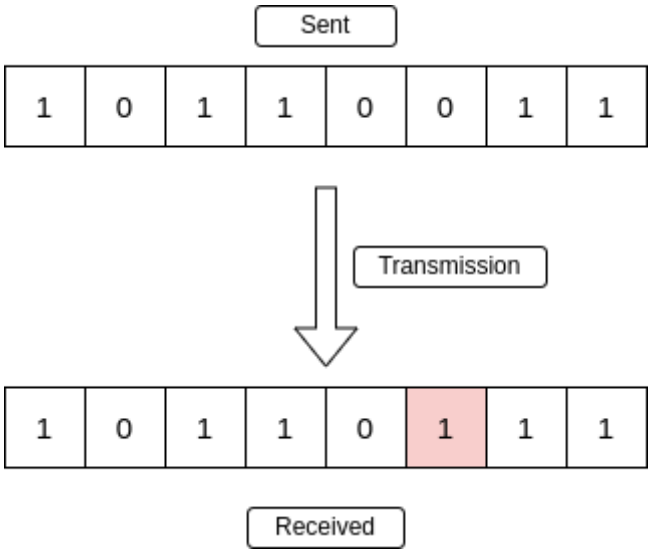
Error Detection

When data is transmitted from one device to another device, the system does not guarantee whether the data received by the device is identical to the data transmitted by another device. An Error is a situation when the message received at the receiver end is not identical to the message transmitted.

Types of Errors

Single-Bit Error

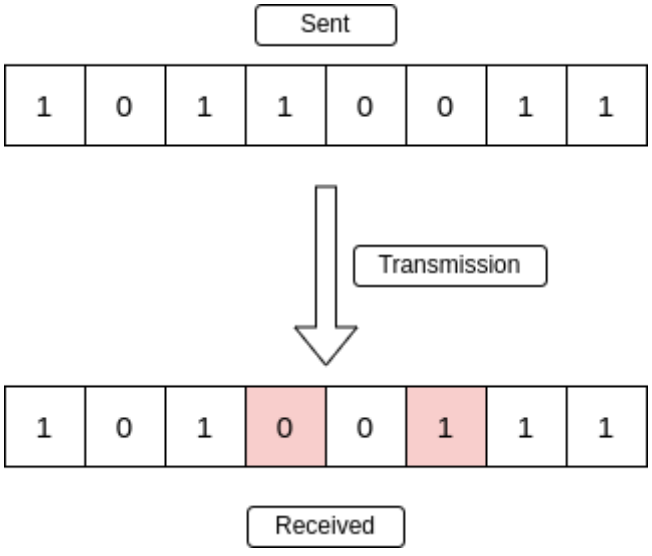
A single-bit error refers to a type of data transmission error that occurs when one bit (i.e., a single binary digit) of a transmitted data unit is altered during transmission, resulting in an incorrect or corrupted data unit.



Single-Bit Error

Multiple-Bit Error

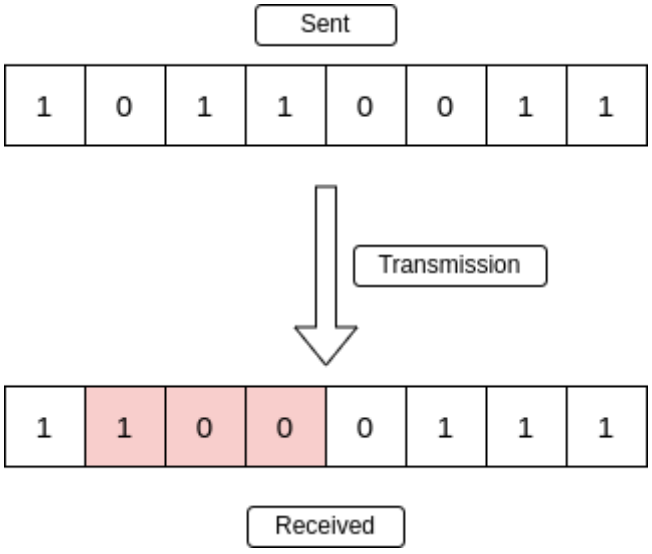
A multiple-bit error is an error type that arises when more than one bit in a data transmission is affected. Although multiple-bit errors are relatively rare when compared to single-bit errors, they can still occur, particularly in high-noise or high-interference digital environments.



Multiple-Bit Error

Burst Error

When several consecutive bits are flipped mistakenly in digital transmission, it creates a burst error. This error causes a sequence of consecutive incorrect values.



Burst Error

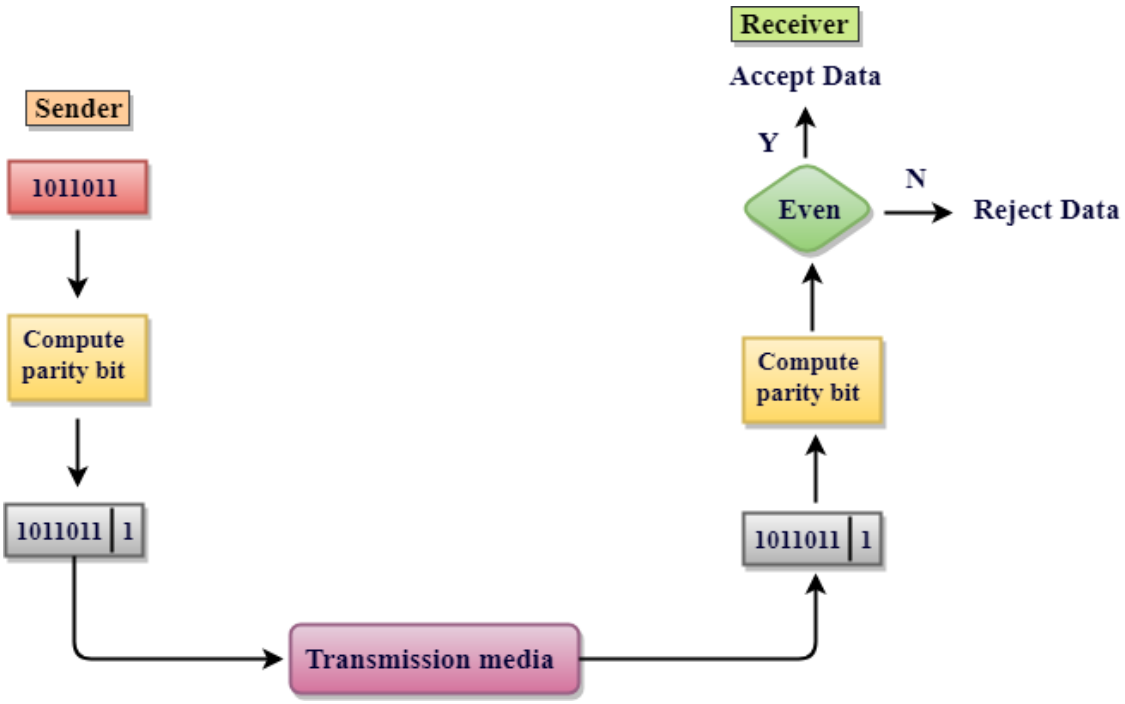
Error Detecting Techniques:

The most popular Error Detecting Techniques are:

- Single parity check
- Two-dimensional parity check
- Checksum
- Cyclic redundancy check

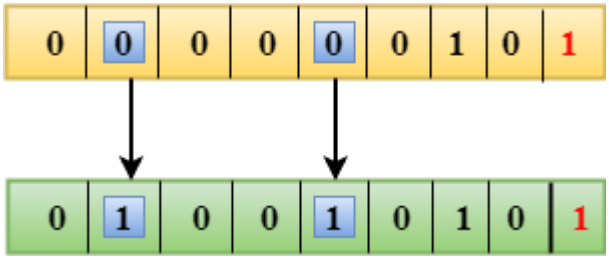
Single Parity Check

- Single Parity checking is the simple mechanism and inexpensive to detect the errors.
- In this technique, a redundant bit is also known as a parity bit which is appended at the end of the data unit so that the number of 1s becomes even. Therefore, the total number of transmitted bits would be 8 bits.
- If the number of 1s bits is odd, then parity bit 1 is appended and if the number of 1s bits is even, then parity bit 0 is appended at the end of the data unit.
- At the receiving end, the parity bit is calculated from the received data bits and compared with the received parity bit.
- This technique generates the total number of 1s even, so it is known as even-parity checking.



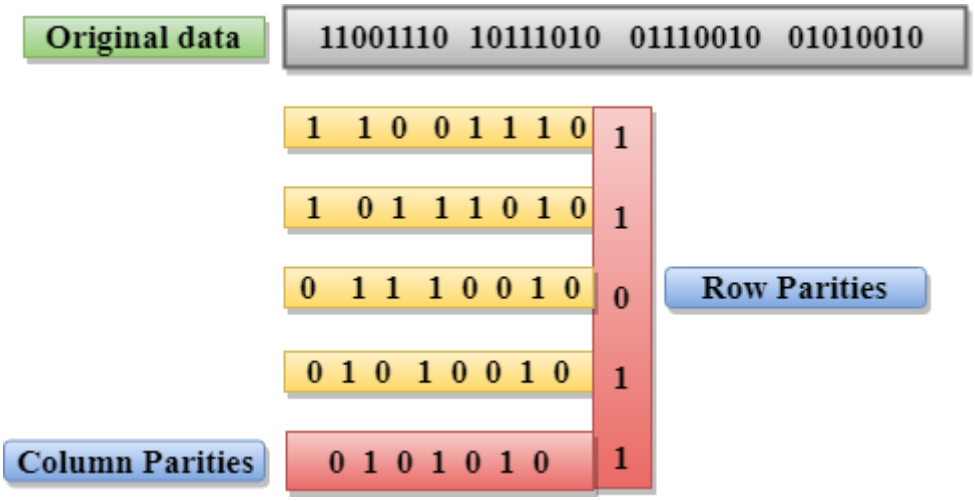
Drawbacks Of Single Parity Checking

- It can only detect single-bit errors which are very rare.
- If two bits are interchanged, then it cannot detect the errors.



Two-Dimensional Parity Check

- Performance can be improved by using **Two-Dimensional Parity Check** which organizes the data in the form of a table.
- Parity check bits are computed for each row, which is equivalent to the single-parity check.
- In Two-Dimensional Parity check, a block of bits is divided into rows, and the redundant row of bits is added to the whole block.
- At the receiving end, the parity bits are compared with the parity bits computed from the received data.



Drawbacks Of 2D Parity Check

- If two bits in one data unit are corrupted and two bits exactly the same position in another data unit are also corrupted, then 2D Parity checker will not be able to detect the error.
- This technique cannot be used to detect the 4-bit errors or more in some cases.

Checksum

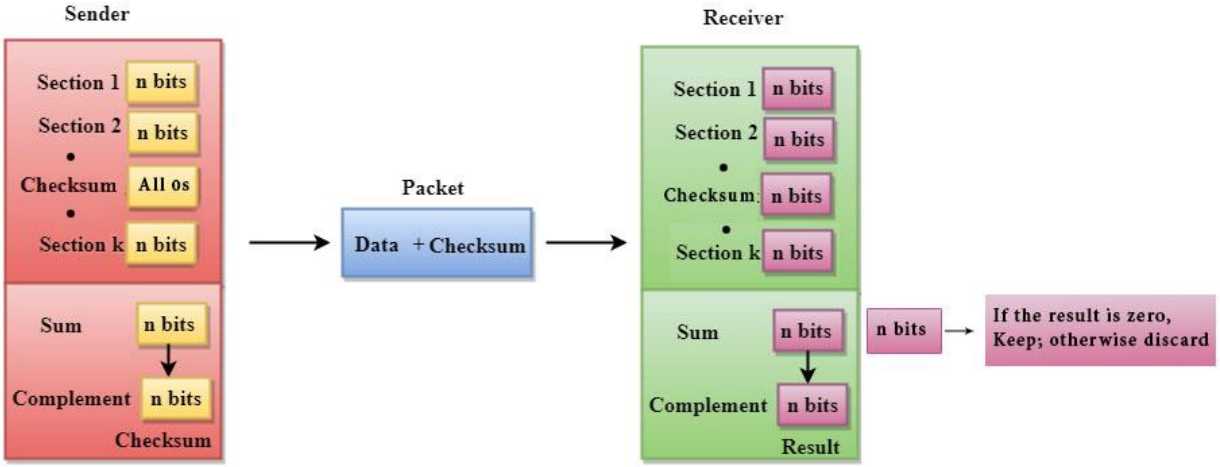
A Checksum is an error detection technique based on the concept of redundancy.

It is divided into two parts:

Checksum Generator

A Checksum is generated at the sending side. Checksum generator subdivides the data into equal segments of n bits each, and all these segments are added together by using one's complement arithmetic. The sum is complemented and appended to the original data, known as checksum field. The extended data is transmitted across the network.

Suppose L is the total sum of the data segments, then the checksum would be ?L



The Sender follows the given steps:

1. The block unit is divided into k sections, and each of n bits.
2. All the k sections are added together by using one's complement to get the sum.
3. The sum is complemented and it becomes the checksum field.

4. The original data and checksum field are sent across the network.

Checksum Checker

A Checksum is verified at the receiving side. The receiver subdivides the incoming data into equal segments of n bits each, and all these segments are added together, and then this sum is complemented. If the complement of the sum is zero, then the data is accepted otherwise data is rejected.

Cyclic Redundancy Check (CRC)

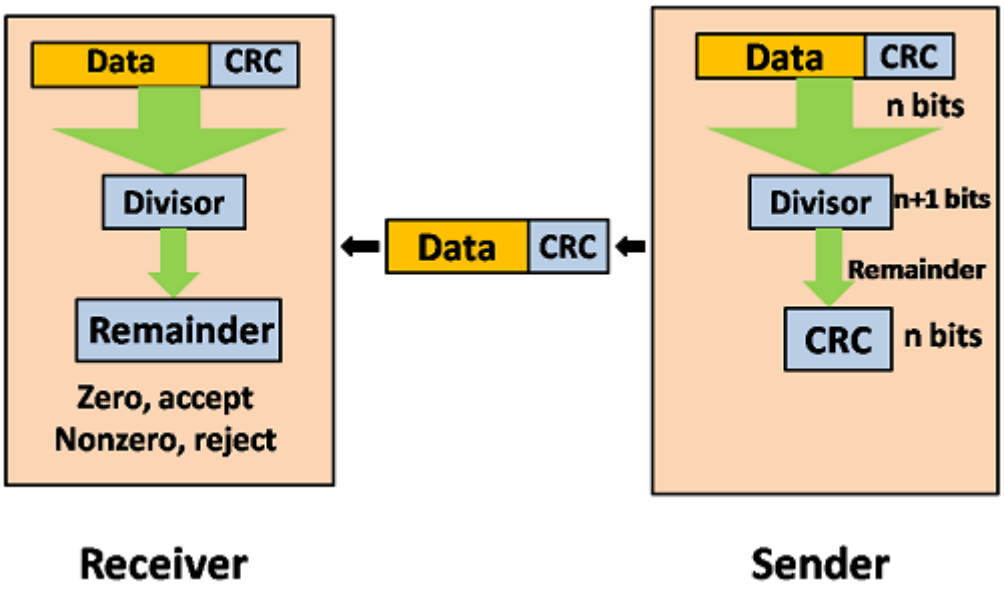
CRC is a redundancy error technique used to determine the error.

Following are the steps used in CRC for error detection:

- In CRC technique, a string of n 0s is appended to the data unit, and this n number is less than the number of bits in a predetermined number, known as division which is n+1 bits.
- Secondly, the newly extended data is divided by a divisor using a process is known as binary division. The remainder generated from this division is known as CRC remainder.
- Thirdly, the CRC remainder replaces the appended 0s at the end of the original data. This newly generated unit is sent to the receiver.
- The receiver receives the data followed by the CRC remainder. The receiver will treat this whole unit as a single unit, and it is divided by the same divisor that was used to find the CRC remainder.

If the resultant of this division is zero which means that it has no error, and the data is accepted.

If the resultant of this division is not zero which means that the data consists of an error. Therefore, the data is discarded.

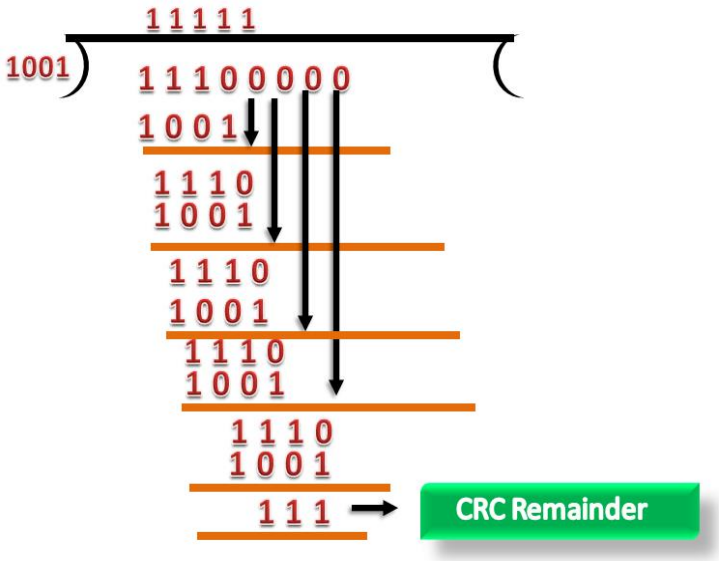


Let's understand this concept through an example:

Suppose the original data is 11100 and divisor is 1001.

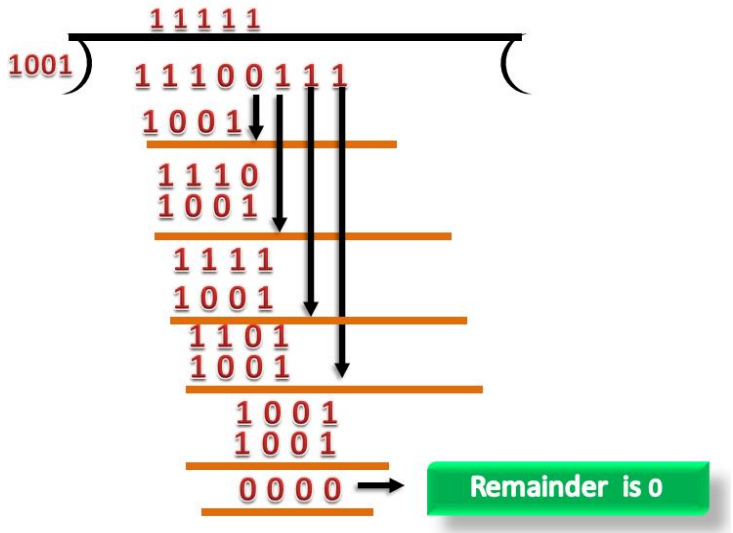
CRC Generator

- A CRC generator uses a modulo-2 division. Firstly, three zeroes are appended at the end of the data as the length of the divisor is 4 and we know that the length of the string 0s to be appended is always one less than the length of the divisor.
- Now, the string becomes 11100000, and the resultant string is divided by the divisor 1001.
- The remainder generated from the binary division is known as CRC remainder. The generated value of the CRC remainder is 111.
- CRC remainder replaces the appended string of 0s at the end of the data unit, and the final string would be 11100111 which is sent across the network.



CRC Checker

- The functionality of the CRC checker is similar to the CRC generator.
- When the string 11100111 is received at the receiving end, then CRC checker performs the modulo-2 division.
- A string is divided by the same divisor, i.e., 1001.
- In this case, CRC checker generates the remainder of zero. Therefore, the data is accepted.



Error Correction

Error Correction codes are used to detect and correct the errors when data is transmitted from the sender to the receiver.

Error Correction can be handled in two ways:

- **Backward error correction:** Once the error is discovered, the receiver requests the sender to retransmit the entire data unit.
- **Forward error correction:** In this case, the receiver uses the error-correcting code which automatically corrects the errors.

A single additional bit can detect the error, but cannot correct it.

For correcting the errors, one has to know the exact position of the error. For example, If we want to calculate a single-bit error, the error correction code will determine which one of seven bits is in error. To achieve this, we have to add some additional redundant bits.

Suppose r is the number of redundant bits and d is the total number of the data bits. The number of redundant bits r can be calculated by using the formula:

$$2^r >= d+r+1$$

The value of r is calculated by using the above formula. For example, if the value of d is 4, then the possible smallest value that satisfies the above relation would be 3.

To determine the position of the bit which is in error, a technique developed by R.W Hamming is Hamming code which can be applied to any length of the data unit and uses the relationship between data units and redundant units.

Hamming Code

Parity bits: The bit which is appended to the original data of binary bits so that the total number of 1s is even or odd.

Even parity: To check for even parity, if the total number of 1s is even, then the value of the parity bit is 0. If the total number of 1s occurrences is odd, then the value of the parity bit is 1.

Odd Parity: To check for odd parity, if the total number of 1s is even, then the value of parity bit is 1. If the total number of 1s is odd, then the value of parity bit is 0.

Algorithm of Hamming code:

- An information of 'd' bits are added to the redundant bits 'r' to form d+r.
- The location of each of the (d+r) digits is assigned a decimal value.
- The 'r' bits are placed in the positions 1,2,.....2^{k-1}.
- At the receiving end, the parity bits are recalculated. The decimal value of the parity bits determines the position of an error.

Relationship b/w Error position & binary number.

Error Position	Binary Number
0	000
1	001
2	010
3	011
4	100
5	101
6	110
7	111

Let's understand the concept of Hamming code through an example:

Suppose the original data is 1010 which is to be sent.

Total number of data bits 'd' = 4

Number of redundant bits r : $2^r >= d+r+1$

$2^r >= 4+r+1$

Therefore, the value of r is 3 that satisfies the above relation.

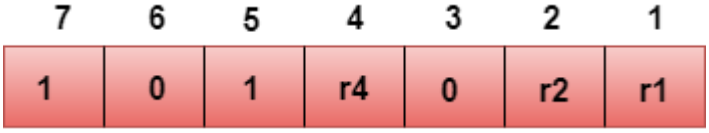
Total number of bits = d+r = 4+3 = 7;

Determining the position of the redundant bits

The number of redundant bits is 3. The three bits are represented by r1, r2, r4. The position of the redundant bits is calculated with corresponds to the raised power of 2. Therefore, their corresponding positions are **1, 2^1, 2^2**.

1. The position of r1 = 1
2. The position of r2 = 2
3. The position of r4 = 4

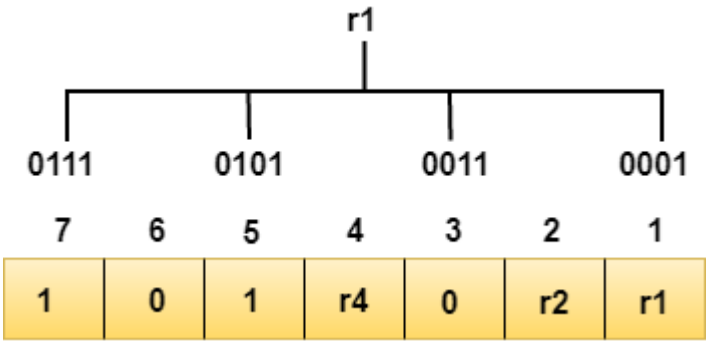
Representation of Data on the addition of parity bits:



Determining the Parity bits

Determining the r1 bit

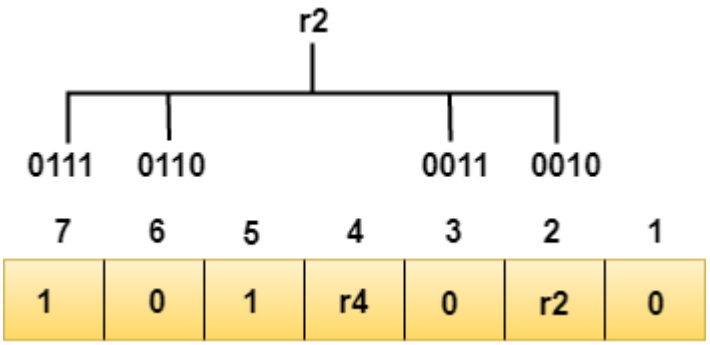
The r1 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the first position.



We observe from the above figure that the bit positions that includes 1 in the first position are 1, 3, 5, 7. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r1 is **even, therefore, the value of the r1 bit is 0**.

Determining r2 bit

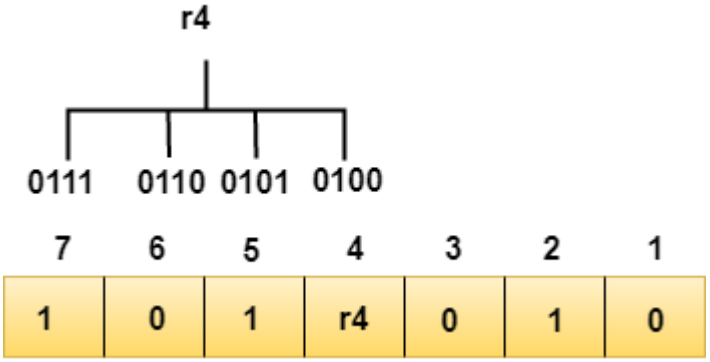
The r2 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the second position.



We observe from the above figure that the bit positions that includes 1 in the second position are **2, 3, 6, 7**. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r2 is **odd, therefore, the value of the r2 bit is 1**.

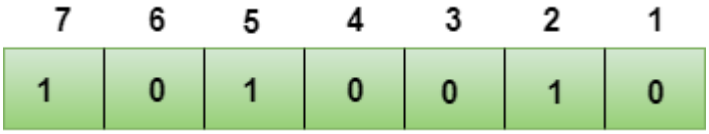
Determining r4 bit

The r4 bit is calculated by performing a parity check on the bit positions whose binary representation includes 1 in the third position.



We observe from the above figure that the bit positions that includes 1 in the third position are **4, 5, 6, 7**. Now, we perform the even-parity check at these bit positions. The total number of 1 at these bit positions corresponding to r4 is **even, therefore, the value of the r4 bit is 0**.

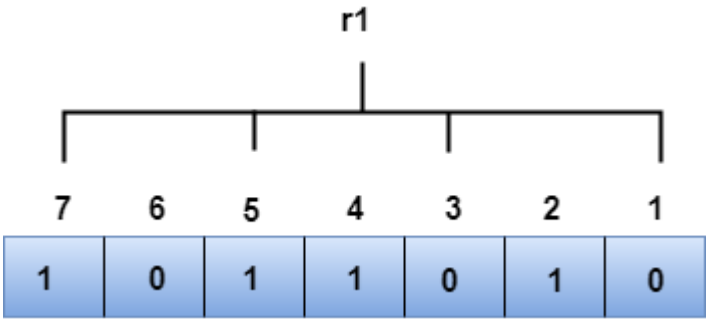
Data transferred is given below:



Suppose the 4th bit is changed from 0 to 1 at the receiving end, then parity bits are recalculated.

R1 bit

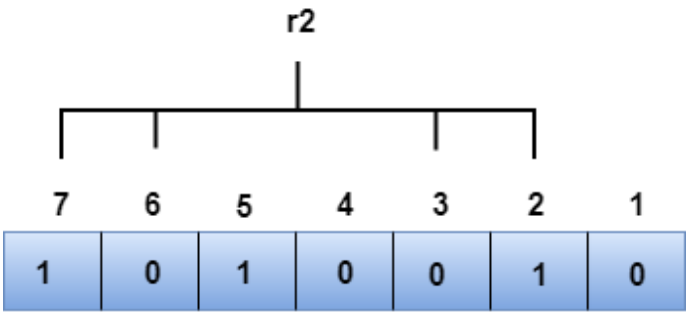
The bit positions of the r1 bit are 1,3,5,7



We observe from the above figure that the binary representation of r1 is 1100. Now, we perform the even-parity check, the total number of 1s appearing in the r1 bit is an even number. Therefore, the value of r1 is 0.

R2 bit

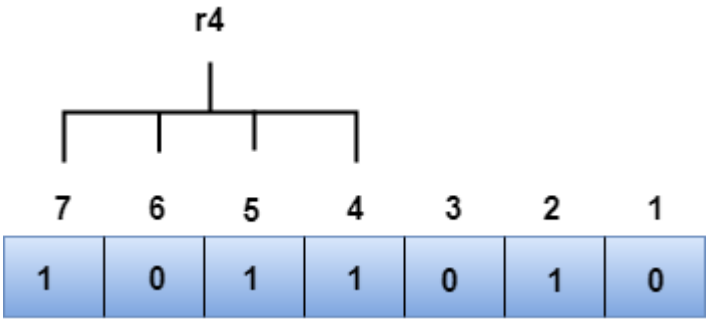
The bit positions of r2 bit are 2,3,6,7.



We observe from the above figure that the binary representation of r2 is 1001. Now, we perform the even-parity check, the total number of 1s appearing in the r2 bit is an even number. Therefore, the value of r2 is 0.

R4 bit

The bit positions of r4 bit are 4,5,6,7.



We observe from the above figure that the binary representation of r4 is 1011. Now, we perform the even-parity check, the total number of 1s appearing in the r4 bit is an odd number. Therefore, the value of r4 is 1.

- *The binary representation of redundant bits, i.e., r4r2r1 is 100, and its corresponding decimal value is 4. Therefore, the error occurs in a 4^{th} bit position. The bit value must be changed from 1 to 0 to correct the error.*