

Error Detection and Correction

There are many reasons such as noise, cross-talk etc., which may help data to get corrupted during transmission. The upper layers work on some generalized view of network architecture and are not aware of actual hardware data processing. Hence, the upper layers expect error-free transmission between the systems. Most of the applications would not function expectedly if they receive erroneous data. Applications such as voice and video may not be that affected and with some errors they may still function well.

Data-link layer uses some error control mechanism to ensure that frames (data bit streams) are transmitted with certain level of accuracy. But to understand how errors are controlled, it is essential to know what types of errors may occur.

Types of Errors

There may be three types of errors:

- **Single bit error**



In a frame, there is only one bit, anywhere though, which is corrupt.

- **Multiple bits error**



Frame is received with more than one bits in corrupted state.

- **Burst error**



Frame contains more than 1 consecutive bits corrupted.

Error control mechanism may involve two possible ways:

- Error detection
- Error correction

Error Detection

Errors in the received frames are detected by means of Parity Check and Cyclic Redundancy Check (CRC). In both cases, few extra bits are sent along with actual data to confirm that bits received at other end are same as they were sent. If the counter-check at receiver' end fails, the bits are considered corrupted.

Parity Check

One extra bit is sent along with the original bits to make number of 1s either even in case of even parity, or odd in case of odd parity.

The sender while creating a frame counts the number of 1s in it. For example, if even parity is used and number of 1s is even then one bit with value 0 is added. This way number of 1s remains even. If the number of 1s is odd, to make it even a bit with value 1 is added.

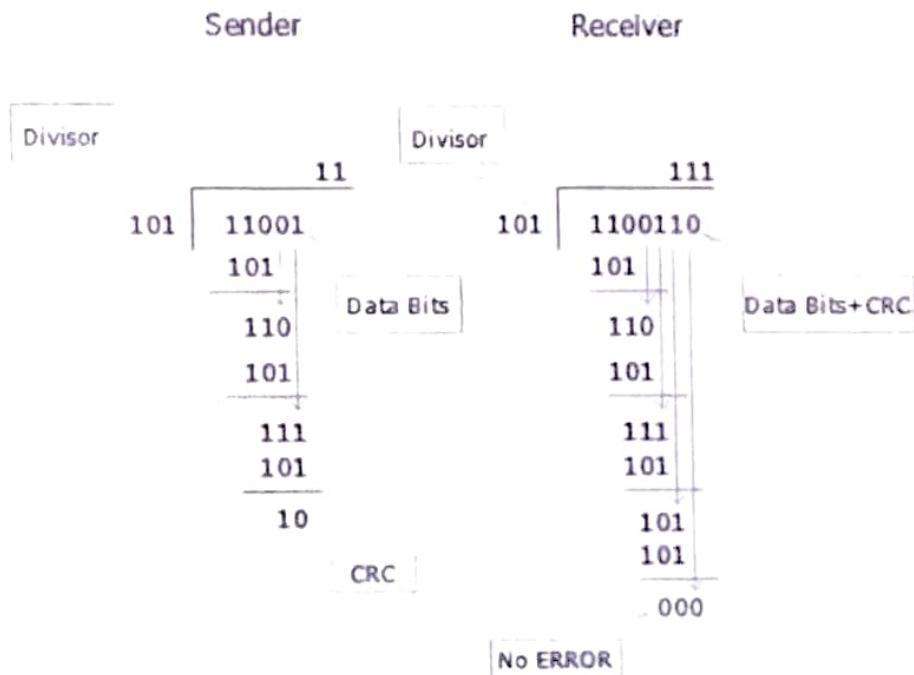


The receiver simply counts the number of 1s in a frame. If the count of 1s is even and even parity is used, the frame is considered to be not-corrupted and is accepted. If the count of 1s is odd and odd parity is used, the frame is still not corrupted.

If a single bit flips in transit, the receiver can detect it by counting the number of 1s. But when more than one bits are erroneous, then it is very hard for the receiver to detect the error.

Cyclic Redundancy Check (CRC)

CRC is a different approach to detect if the received frame contains valid data. This technique involves binary division of the data bits being sent. The divisor is generated using polynomials. The sender performs a division operation on the bits being sent and calculates the remainder. Before sending the actual bits, the sender adds the remainder at the end of the actual bits. Actual data bits plus the remainder is called a codeword. The sender transmits data bits as codewords.



At the other end, the receiver performs division operation on codewords using the same CRC divisor. If the remainder contains all zeros the data bits are accepted, otherwise it is considered as there some data corruption occurred in transit.

Error Correction

In the digital world, error correction can be done in two ways:

- **Backward Error Correction** When the receiver detects an error in the data received, it requests back the sender to retransmit the data unit.
- **Forward Error Correction** When the receiver detects some error in the data received, it executes error-correcting code, which helps it to auto-recover and to correct some kinds of errors.

The first one, Backward Error Correction, is simple and can only be efficiently used where retransmitting is not expensive. For example, fiber optics. But in case of wireless transmission retransmitting may cost too much. In the latter case, Forward Error Correction is used.

To correct the error in data frame, the receiver must know exactly which bit in the frame is corrupted. To locate the bit in error, redundant bits are used as parity bits for error detection. For example, we take ASCII words (7 bits data), then there could be 8 kind of information we need: first seven bits to tell us which bit is error and one more bit to tell that there is no error.

For m data bits, r redundant bits are used. r bits can provide 2^r combinations of information. In $m+r$ bit codeword, there is possibility that the r bits themselves may get corrupted. So the number of r bits used must inform about $m+r$ bit locations plus no-error information, i.e. $m+r+1$.

$$2^r \geq m+r+1$$

Data-link Control and Protocols

Data-link layer is responsible for implementation of point-to-point flow and error control mechanism.

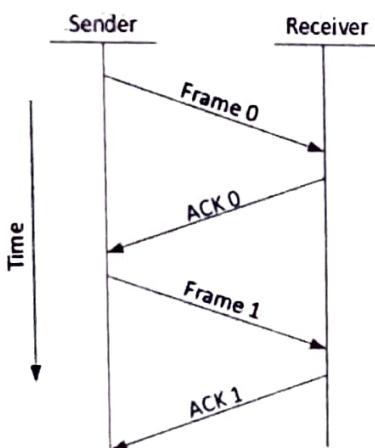
Flow Control

When a data frame (Layer-2 data) is sent from one host to another over a single medium, it is required that the sender and receiver should work at the same speed. That is, sender sends at a speed on which the receiver can process and accept the data. What if the speed (hardware/software) of the sender or receiver differs? If sender is sending too fast the receiver may be overloaded, (swamped) and data may be lost.

Two types of mechanisms can be deployed to control the flow:

- **Stop and Wait**

This flow control mechanism forces the sender after transmitting a data frame to stop and wait until the acknowledgement of the data-frame sent is received.



- **Sliding Window**

In this flow control mechanism, both sender and receiver agree on the number of data-frames after which the acknowledgement should be sent. As we learnt, stop and wait flow control mechanism wastes resources, this protocol tries to make use of underlying resources as much as possible.

Error Control

When data-frame is transmitted, there is a probability that data-frame may be lost in the transit or it is received corrupted. In both cases, the receiver does not receive the correct data-frame and sender does not know anything

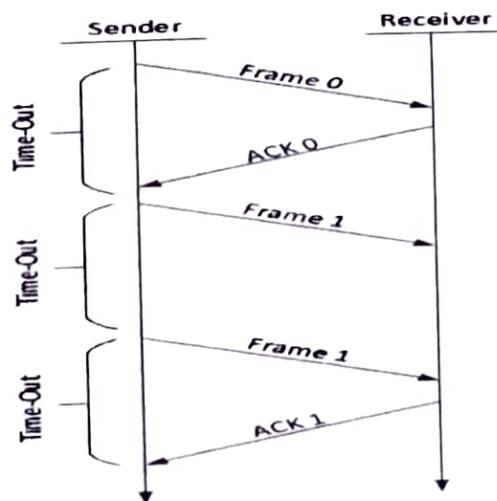
about any loss. In such case, both sender and receiver are equipped with some protocols which helps them to detect transit errors such as loss of data-frame. Hence, either the sender retransmits the data-frame or the receiver may request to resend the previous data-frame.

Requirements for error control mechanism:

- **Error detection** - The sender and receiver, either both or any, must ascertain that there is some error in the transit.
- **Positive ACK** - When the receiver receives a correct frame, it should acknowledge it.
- **Negative ACK** - When the receiver receives a damaged frame or a duplicate frame, it sends a NACK back to the sender and the sender must retransmit the correct frame.
- **Retransmission:** The sender maintains a clock and sets a timeout period. If an acknowledgement of a data-frame previously transmitted does not arrive before the timeout the sender retransmits the frame, thinking that the frame or its acknowledgement is lost in transit.

There are three types of techniques available which Data-link layer may deploy to control the errors by Automatic Repeat Requests (ARQ):

- Stop-and-wait ARQ

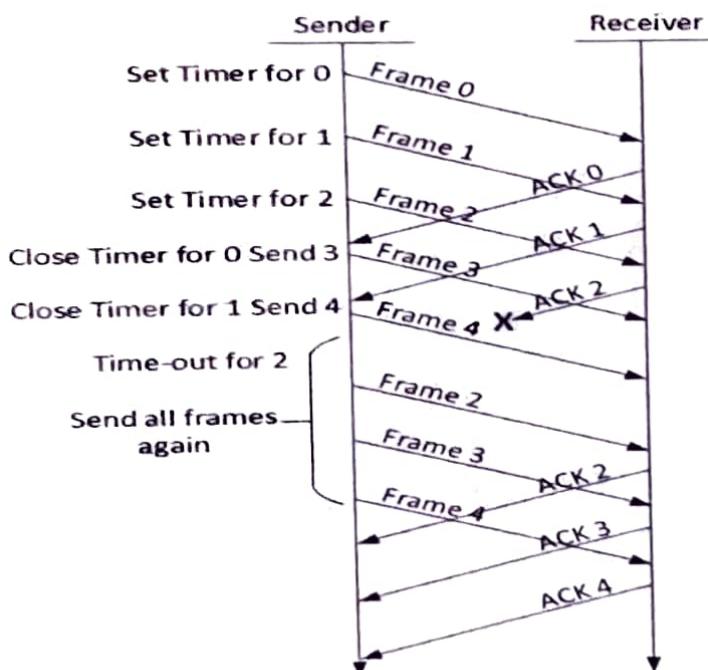


The following transition may occur in Stop-and-Wait ARQ:

- The sender maintains a timeout counter.
 - When a frame is sent, the sender starts the timeout counter.
 - If acknowledgement of frame comes in time, the sender transmits the next frame in queue.
 - If acknowledgement does not come in time, the sender assumes that either the frame or its acknowledgement is lost in transit. Sender retransmits the frame and starts the timeout counter.
 - If a negative acknowledgement is received, the sender retransmits the frame.
-

• Go-Back-N ARQ

Stop and wait ARQ mechanism does not utilize the resources at their best. When the acknowledgement is received, the sender sits idle and does nothing. In Go-Back-N ARQ method, both sender and receiver maintain a window.

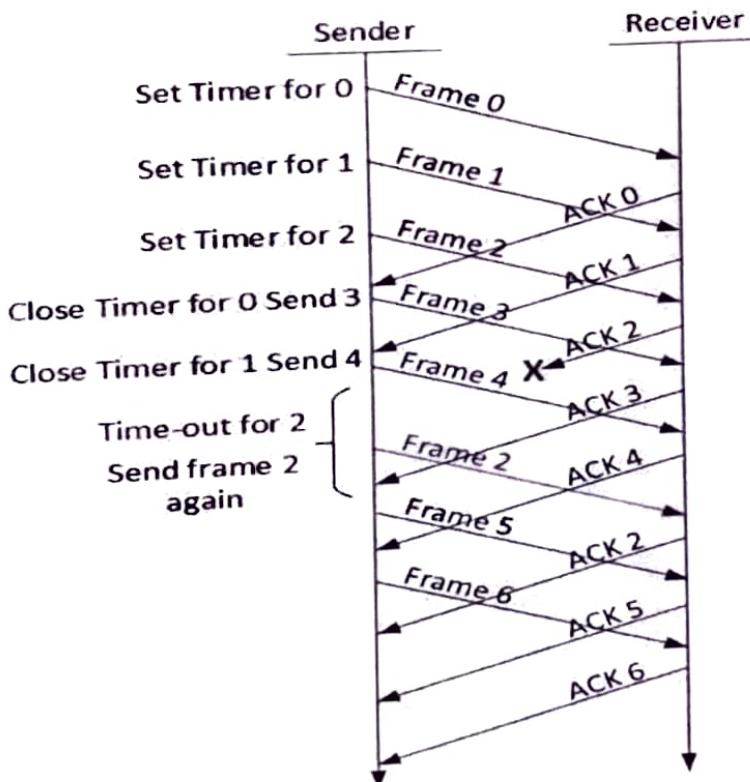


The sending-window size enables the sender to send multiple frames without receiving the acknowledgement of the previous ones. The receiving-window enables the receiver to receive multiple frames and acknowledge them. The receiver keeps track of incoming frame's sequence number.

When the sender sends all the frames in window, it checks up to what sequence number it has received positive acknowledgement. If all frames are positively acknowledged, the sender sends next set of frames. If sender finds that it has received NACK or has not receive any ACK for a particular frame, it retransmits all the frames after which it does not receive any positive ACK.

- **Selective Repeat ARQ**

In Go-back-N ARQ, it is assumed that the receiver does not have any buffer space for its window size and has to process each frame as it comes. This enforces the sender to retransmit all the frames which are not acknowledged.



TRANSMISSION CONTROL PROTOCOL (TCP)

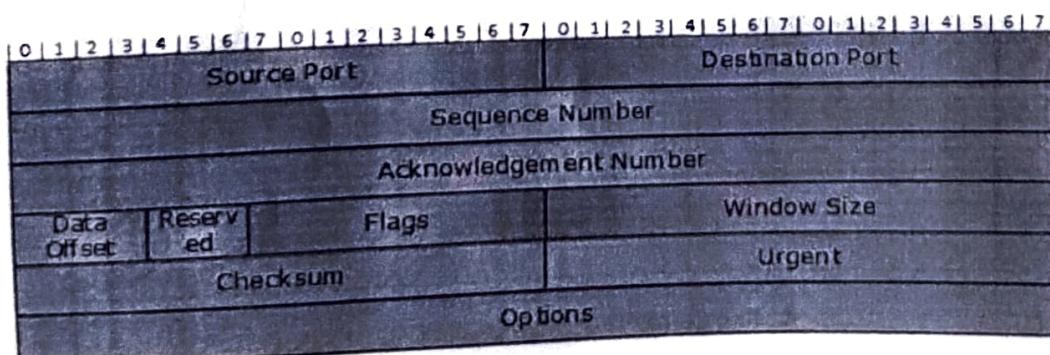
The transmission Control Protocol (TCP) is one of the most important protocols of Internet Protocols suite. It is most widely used protocol for data transmission in communication network such as internet.

Features

- TCP is reliable protocol. That is, the receiver always sends either positive or negative acknowledgement about the data packet to the sender, so that the sender always has bright clue about whether the data packet is reached the destination or it needs to resend it.
- TCP ensures that the data reaches intended destination in the same order it was sent.
- TCP is connection oriented. TCP requires that connection between two remote points be established before sending actual data.
- TCP provides error-checking and recovery mechanism.
- TCP provides end-to-end communication.
- TCP provides flow control and quality of service.
- TCP operates in Client/Server point-to-point mode.
- TCP provides full duplex server, i.e. it can perform roles of both receiver and sender.

Header

The length of TCP header is minimum 20 bytes long and maximum 60 bytes.



- **Source Port (16-bits)** - It identifies source port of the application process on the sending device.
- **Destination Port (16-bits)** - It identifies destination port of the application process on the receiving device.
- **Sequence Number (32-bits)** - Sequence number of data bytes of a segment in a session.
- **Acknowledgement Number (32-bits)** - When ACK flag is set, this number contains the next sequence number of the data byte expected and works as acknowledgement of the previous data received.
- **Data Offset (4-bits)** - This field implies both, the size of TCP header (32-bit words) and the offset of data in current packet in the whole TCP segment.
- **Reserved (3-bits)** - Reserved for future use and all are set zero by default.
- **Flags (1-bit each)**
 - **NS** - Nonce Sum bit is used by Explicit Congestion Notification signaling process.
 - **CWR** - When a host receives packet with ECE bit set, it sets Congestion Windows Reduced to acknowledge that ECE received.
 - **ECE** -It has two meanings:
 - If SYN bit is clear to 0, then ECE means that the IP packet has its CE (congestion experience) bit set.
 - If SYN bit is set to 1, ECE means that the device is ECT capable.
 - **URG** - It indicates that Urgent Pointer field has significant data and should be processed.
 - **ACK** - It indicates that Acknowledgement field has significance. If ACK is cleared to 0, it indicates that packet does not contain any acknowledgement.
 - **PSH** - When set, it is a request to the receiving station to PUSH data (as soon as it comes) to the receiving application without buffering it.
 - **RST** - Reset flag has the following features:
 - It is used to refuse an incoming connection.
 - It is used to reject a segment.
 - It is used to restart a connection.
 - **SYN** - This flag is used to set up a connection between hosts.

- **FIN** - This flag is used to release a connection and no more data is exchanged thereafter. Because packets with SYN and FIN flags have sequence numbers, they are processed in correct order.
- **Windows Size** - This field is used for flow control between two stations and indicates the amount of buffer (in bytes) the receiver has allocated for a segment, i.e. how much data is the receiver expecting.
- **Checksum** - This field contains the checksum of Header, Data and Pseudo Headers.
- **Urgent Pointer** - It points to the urgent data byte if URG flag is set to 1.
- **Options** - It facilitates additional options which are not covered by the regular header. Option field is always described in 32-bit words. If this field contains data less than 32-bit, padding is used to cover the remaining bits to reach 32-bit boundary.

Addressing

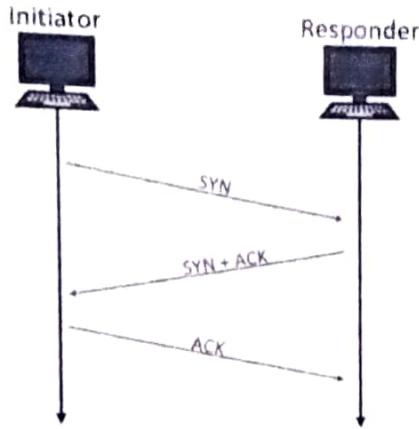
TCP communication between two remote hosts is done by means of port numbers (TSAPs).

Ports numbers can range from 0 – 65535 which are divided as:

- System Ports (0 – 1023)
- User Ports (1024 – 49151)
- Private/Dynamic Ports (49152 – 65535)

Connection Management

TCP communication works in Server/Client model. The client initiates the connection and the server either accepts or rejects it. Three-way handshaking is used for connection management.



Establishment

Client initiates the connection and sends the segment with a Sequence number. Server acknowledges it back with its own Sequence number and ACK of client's segment which is one more than client's Sequence number. Client after receiving ACK of its segment sends an acknowledgement of Server's response.

Release

Either of server and client can send TCP segment with FIN flag set to 1. When the receiving end responds it back by ACKnowledging FIN, that direction of TCP communication is closed and connection is released.

Bandwidth Management

TCP uses the concept of window size to accommodate the need of Bandwidth management. Window size tells the sender at the remote end, the number of data byte segments the receiver at this end can receive. TCP uses slow start phase by using window size 1 and increases the window size exponentially after each successful communication.

For example, the client uses windows size 2 and sends 2 bytes of data. When the acknowledgement of this segment received the windows size is doubled to 4 and next sent the

segment sent will be 4 data bytes long. When the acknowledgement of 4-byte data segment is received, the client sets windows size to 8 and so on.

If an acknowledgement is missed, i.e. data lost in transit network or it received NACK, then the window size is reduced to half and slow start phase starts again.

Error Control &and Flow Control

TCP uses port numbers to know what application process it needs to handover the data segment. Along with that, it uses sequence numbers to synchronize itself with the remote host. All data segments are sent and received with sequence numbers. The Sender knows which last data segment was received by the Receiver when it gets ACK. The Receiver knows about the last segment sent by the Sender by referring to the sequence number of recently received packet.

If the sequence number of a segment recently received does not match with the sequence number the receiver was expecting, then it is discarded and NACK is sent back. If two segments arrive with the same sequence number, the TCP timestamp value is compared to make a decision.

Multiplexing

The technique to combine two or more data streams in one session is called Multiplexing. When a TCP client initializes a connection with Server, it always refers to a well-defined port number which indicates the application process. The client itself uses a randomly generated port number from private port number pools.

Using TCP Multiplexing, a client can communicate with a number of different application process in a single session. For example, a client requests a web page which in turn contains different types of data (HTTP, SMTP, FTP etc.) the TCP session timeout is increased and the session is kept open for longer time so that the three-way handshake overhead can be avoided.

This enables the client system to receive multiple connection over single virtual connection. These virtual connections are not good for Servers if the timeout is too long.

Congestion Control

When large amount of data is fed to system which is not capable of handling it, congestion occurs. TCP controls congestion by means of Window mechanism. TCP sets a window size telling the other end how much data segment to send. TCP may use three algorithms for congestion control:

- Additive increase, Multiplicative Decrease
- Slow Start
- Timeout React

Timer Management

TCP uses different types of timer to control and management various tasks:

Keep-alive timer:

- This timer is used to check the integrity and validity of a connection.
- When keep-alive time expires, the host sends a probe to check if the connection still exists.

Retransmission timer:

- This timer maintains stateful session of data sent.
- If the acknowledgement of sent data does not receive within the Retransmission time, the data segment is sent again.

Persist timer:

- TCP session can be paused by either host by sending Window Size 0.
- To resume the session a host needs to send Window Size with some larger value.
- If this segment never reaches the other end, both ends may wait for each other for infinite time.

- When the Persist timer expires, the host re-sends its window size to let the other end know.
- Persist Timer helps avoid deadlocks in communication.

Timed-Wait:

- After releasing a connection, either of the hosts waits for a Timed-Wait time to terminate the connection completely.
- This is in order to make sure that the other end has received the acknowledgement of its connection termination request.
- Timed-out can be a maximum of 240 seconds (4 minutes).

Crash Recovery

TCP is very reliable protocol. It provides sequence number to each of byte sent in segment. It provides the feedback mechanism i.e. when a host receives a packet, it is bound to ACK that packet having the next sequence number expected (if it is not the last segment).

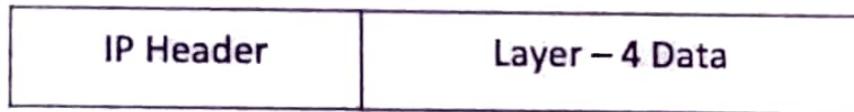
When a TCP Server crashes mid-way communication and re-starts its process it sends TPDU broadcast to all its hosts. The hosts can then send the last data segment which was never unacknowledged and carry onwards.

Internet Protocol Version 4 (IPv4)

Internet Protocol is one of the major protocols in the TCP/IP protocols suite. This protocol works at the network layer of the OSI model and at the Internet layer of the TCP/IP model. Thus this protocol has the responsibility of identifying hosts based upon their logical addresses and to route data among them over the underlying network.

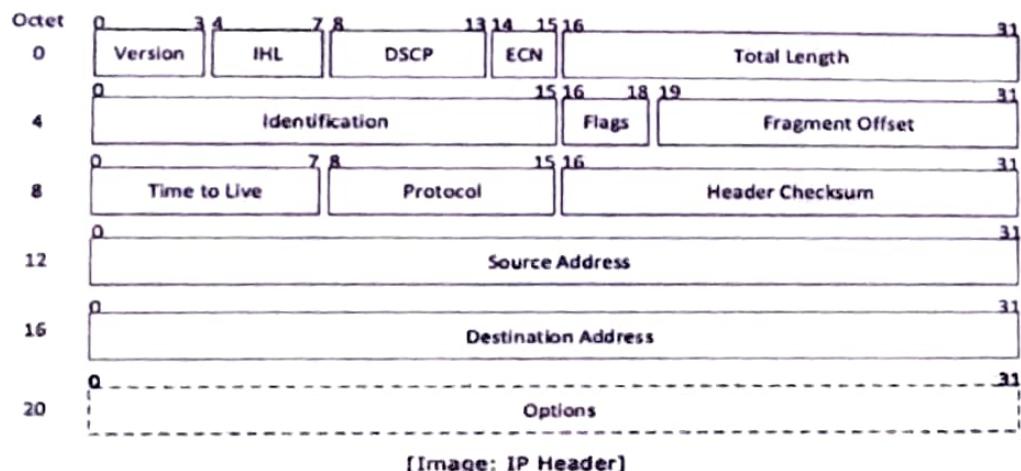
IP provides a mechanism to uniquely identify hosts by an IP addressing scheme. IP uses best effort delivery, i.e. it does not guarantee that packets would be delivered to the destined host, but it will do its best to reach the destination. Internet Protocol version 4 uses 32-bit logical address.

Internet Protocol being a layer-3 protocol (OSI) takes data Segments from layer-4 (Transport) and divides it into packets. IP packet encapsulates data unit received from above layer and add to its own header information.



(IP Encapsulation)

The encapsulated data is referred to as IP Payload. IP header contains all the necessary information to deliver the packet at the other end.



[Image: IP Header]

IP header includes many relevant information including Version Number, which, in this context, is 4. Other details are as follows:

- **Version:** Version no. of Internet Protocol used (e.g. IPv4).
- **IHL:** Internet Header Length; Length of entire IP header.
- **DSCP:** Differentiated Services Code Point; this is Type of Service.
- **ECN:** Explicit Congestion Notification; It carries information about the congestion seen in the route.
- **Total Length:** Length of entire IP Packet (including IP header and IP Payload).
- **Identification:** If IP packet is fragmented during the transmission, all the fragments contain same identification number. to identify original IP packet they belong to.
- **Flags:** As required by the network resources, if IP Packet is too large to handle, these 'flags' tells if they can be fragmented or not. In this 3-bit flag, the MSB is always set to '0'.
- **Fragment Offset:** This offset tells the exact position of the fragment in the original IP Packet.
- **Time to Live:** To avoid looping in the network, every packet is sent with some TTL value set, which tells the network how many routers (hops) this packet can cross. At each hop, its value is decremented by one and when the value reaches zero, the packet is discarded.
- **Protocol:** Tells the Network layer at the destination host, to which Protocol this packet belongs to, i.e. the next level Protocol. For example protocol number of ICMP is 1, TCP is 6 and UDP is 17.
- **Header Checksum:** This field is used to keep checksum value of entire header which is then used to check if the packet is received error-free.
- **Source Address:** 32-bit address of the Sender (or source) of the packet.
- **Destination Address:** 32-bit address of the Receiver (or destination) of the packet.
- **Options:** This is optional field, which is used if the value of IHL is greater than 5. These options may contain values for options such as Security, Record Route, Time Stamp, etc.

The User Datagram Protocol (UDP)

The User Datagram Protocol (UDP) is simplest Transport Layer communication protocol available of the TCP/IP protocol suite. It involves minimum amount of communication mechanism. UDP is said to be an unreliable transport protocol but it uses IP services which provides best effort delivery mechanism.

In UDP, the receiver does not generate an acknowledgement of packet received and in turn, the sender does not wait for any acknowledgement of packet sent. This shortcoming makes this protocol unreliable as well as easier on processing.

Requirement of UDP

A question may arise, why do we need an unreliable protocol to transport the data? We deploy UDP where the acknowledgement packets share significant amount of bandwidth along with the actual data. For example, in case of video streaming, thousands of packets are forwarded towards its users. Acknowledging all the packets is troublesome and may contain huge amount of bandwidth wastage. The best delivery mechanism of underlying IP protocol ensures best efforts to deliver its packets, but even if some packets in video streaming get lost, the impact is not calamitous and can be ignored easily. Loss of few packets in video and voice traffic sometimes goes unnoticed.

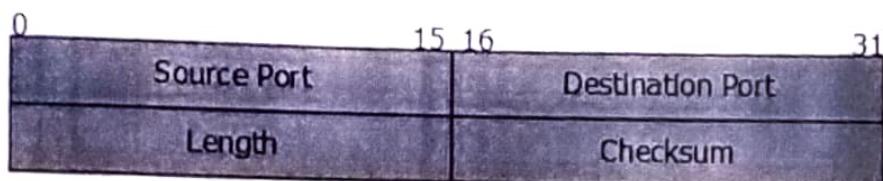
Features

- UDP is used when acknowledgement of data does not hold any significance.
- UDP is good protocol for data flowing in one direction.
- UDP is simple and suitable for query based communications.
- UDP is not connection oriented.
- UDP does not provide congestion control mechanism.
- UDP does not guarantee ordered delivery of data.
- UDP is stateless.

- UDP is suitable protocol for streaming applications such as VoIP, multimedia streaming.

UDP Header

UDP header is as simple as its function.



UDP header contains four main parameters:

- **Source Port** - This 16 bits information is used to identify the source port of the packet.
- **Destination Port** - This 16 bits information, is used identify application level service on destination machine.
- **Length** - Length field specifies the entire length of UDP packet (including header). It is 16-bits field and minimum value is 8-byte, i.e. the size of UDP header itself.
- **Checksum** - This field stores the checksum value generated by the sender before sending. IPv4 has this field as optional so when checksum field does not contain any value it is made 0 and all its bits are set to zero.

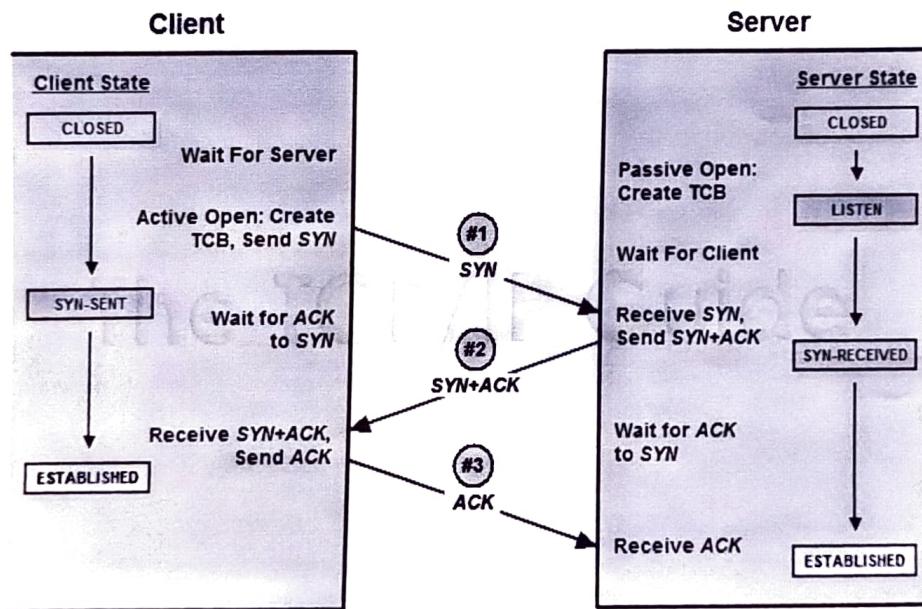
UDP application

Here are few applications where UDP is used to transmit data:

- Domain Name Services
- Simple Network Management Protocol
- Trivial File Transfer Protocol
- Routing Information Protocol
- Kerberos

Table 152: TCP "Three-Way Handshake" Connection Establishment Procedure

| Client | | | Server | | |
|-------------|---|---------------|---------------|--|---------------|
| Start State | Action | Move To State | Start State | Action | Move To State |
| CLOSED | The client cannot do anything until the server has performed a passive OPEN and is ready to accept a connection. (Well, it can try but nothing will be accomplished until the server is ready.) | — | CLOSED | The server performs a passive OPEN, creating a transmission control block (TCB) for the connection and readying itself for the receipt of a connection request (SYN) from a client. | LISTEN |
| CLOSED | Step #1 Transmit: The client performs an active OPEN, creating a transmission control block (TCB) for the connection and sending a SYN message to the server. | SYN-SENT | LISTEN | The server waits for contact from a client. | — |
| SYN-SENT | The client waits to receive an ACK to the SYN it has sent, as well as the server's SYN. | — | LISTEN | Step #1 Receive, Step #2 Transmit: The server receives the SYN from the client. It sends a single SYN+ACK message back to the client that contains an ACK for the client's SYN, and the server's own SYN. | SYN- RECEIVED |
| SYN-SENT | Step #2 Receive, Step #3 Transmit: The client receives from the server the SYN-ACK containing the ACK to the client's SYN, and the SYN from the server. It sends the server an ACK for the server's SYN. The client is now done with the connection establishment. | ESTABLISHED | SYN- RECEIVED | The server waits for an ACK to the SYN it sent previously. | — |
| ESTABLISHED | The client is waiting for the server to finish connection establishment so they can operate normally. | — | SYN- RECEIVED | Step #3 Receive: The server receives the ACK to its SYN and is now done with connection establishment. | ESTABLISHED |
| ESTABLISHED | The client is ready for normal data transfer operations. | — | ESTABLISHED | The server is ready for normal data transfer operations. | — |



Architecture of SONET consists signals, devices, and connections.

1. SONET Devices

Figure 1 shows a simple link using SONET devices. SONET transmission relies on three basic devices: STS multiplexers/demultiplexers, re-regenerators, add/drop multiplexers and terminals.

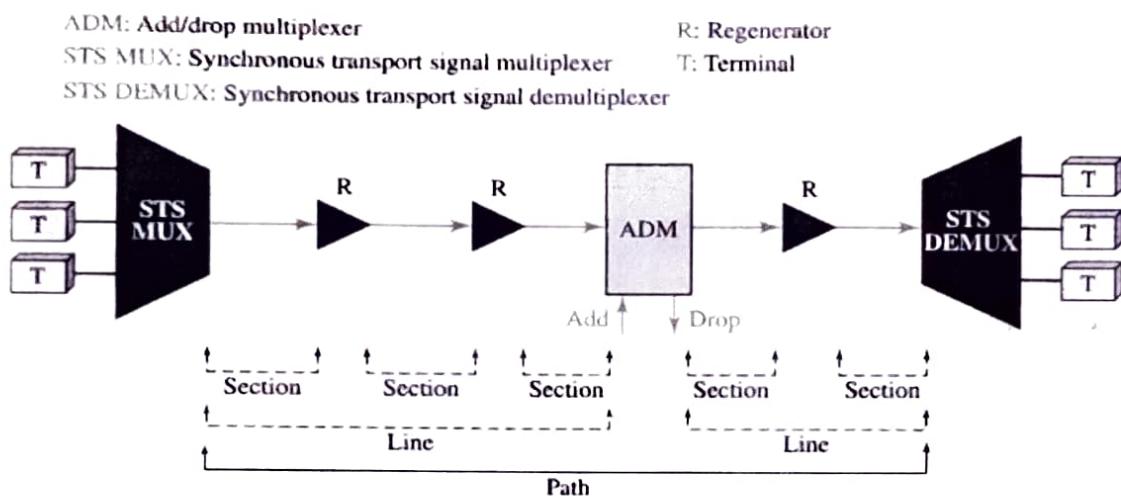


Fig.1

STS Multiplexer/Demultiplexer: STS multiplexers/demultiplexers mark the beginning points and endpoints of a SONET link. They provide the interface between an electrical tributary network and the optical network. An STS multiplexer multiplexes signals from multiple electrical sources and creates the corresponding OC signal. An STS demultiplexer demultiplexes an optical OC signal into corresponding electric signals.

Regenerator: Regenerators extend the length of the links. A regenerator is a repeater, that takes a received optical signal (OC-n), demodulates it into the corresponding electric signal (STS-n), regenerates the electric signal, and finally modulates the electric signal into its correspondent OC-n signal. A SONET regenerator replaces some of the existing overhead information with new information.

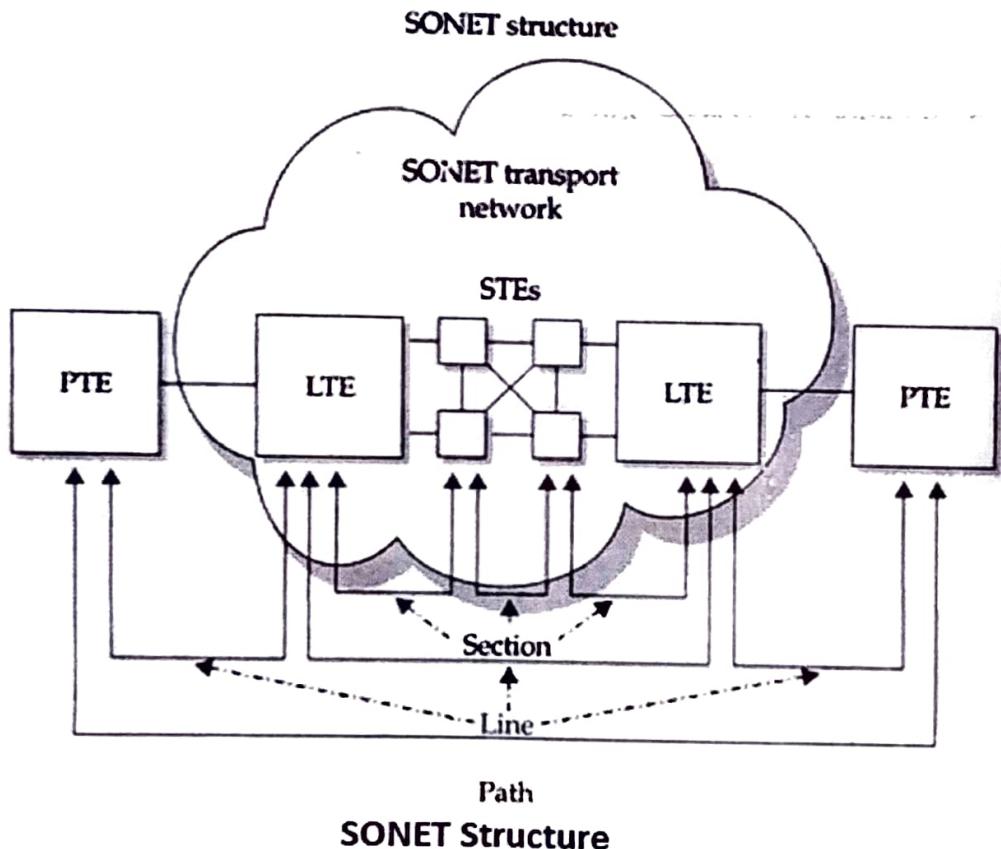
Add/drop Multiplexer: Add/drop multiplexers allow insertion and extraction of signals. An add/drop multiplexer (ADM) can add STSs coming from different sources into a given path or can remove a desired signal from a path and redirect it without demultiplexing the entire signal. Instead of relying on timing and bit positions, add/drop multiplexers use overhead information such as addresses and pointers to identify individual streams.

In the simple configuration shown by Figure 1, a number of incoming electronic signals are fed into an STS multiplexer, where they are combined into a single optical signal.

The optical signal is transmitted to a regenerator, where it is recreated without the noise it has picked up in transit. The regenerated signals from a number of sources are then fed into an add/drop multiplexer. The add/drop multiplexer reorganizes these signals, if necessary, and sends them out as directed by information in the data frames. These remultiplexed signals are sent to another regenerator and from there to the receiving STS demultiplexer, where they are returned to a format usable by the receiving links.

Terminals: A terminal is a device that uses the services of a SONET network. For example, in the Internet, a terminal can be a router that needs to send packets to another router at the other side of a SONET network.

2. Connections



Note: Refer Both the figures

Sections: A section is the optical link connecting two neighboring devices: multiplexer to multiplexer, multiplexer to regenerator, or regenerator to regenerator.

Lines: A line is the portion of the network between two multiplexers: STS multiplexer to add/ drop multiplexer, two add/drop multiplexers, or two STS multiplexers.

Paths: A path is the end-to-end portion of the network between two STS multiplexers. In a simple SONET of two STS multiplexers linked directly to each other, the section, line, and path are the same.

Path-terminating equipment (PTE): Typically the user interface at the CPE.

Line-terminating equipment (LTE): Typically a terminal, switch, add/drop multiplexer, or cross-connect.

Section-terminating equipment (STE): Primarily a regenerator.

3. Signals

SONET defines a hierarchy of electrical signaling levels called synchronous transport signals (STSs). Each STS level (STS-1 to STS-192) supports a certain data rate, specified in megabits per second. The corresponding optical signals are called optical carriers (OCs). SDH specifies a similar system called a synchronous transport module (STM).

STM is intended to be compatible with existing European hierarchies, such as E lines, and with STS levels. To this end, the lowest STM level, STM-1, is defined as 155.520 Mbps, which is exactly equal to STS-3.

| STS | OC | Rate (Mbps) | STM |
|---------|--------|-------------|--------|
| STS-1 | OC-1 | 51.840 | - |
| STS-3 | OC-3 | 155.520 | STM-1 |
| STS-9 | OC-9 | 466.560 | STM-3 |
| STS-12 | OC-12 | 622.080 | STM-4 |
| STS-18 | OC-18 | 933.120 | STM-6 |
| STS-24 | OC-24 | 1244.160 | STM-8 |
| STS-36 | OC-36 | 1866.230 | STM-12 |
| STS-48 | OC-48 | 2488.320 | STM-16 |
| STS-96 | OC-96 | 4976.640 | STM-32 |
| STS-192 | OC-192 | 9953.280 | STM-64 |

From the above table, We observe

- First, the lowest level in this hierarchy has a data rate of 51.840 Mbps, which is greater than that of the DS-3 service (44.736 Mbps). In fact, the STS-1 is designed to accommodate data rates equivalent to those of the DS-3. The difference in capacity is provided to handle the overhead needs of the optical system.
- Second, the STS-3 rate is exactly three times the STS-1 rate; and the STS-9 rate is exactly one-half the STS-18 rate. These relationships mean that 18 STS-1 channels

can be multiplexed into one STS-18, six STS-3 channels can be multiplexed into one STS-18, and so on.

Transmission Medium :: Guided Transmission

Twisted Pair Cable

A twisted pair cable is made of two plastic insulated copper wires twisted together to form a single media. Out of these two wires, only one carries actual signal and another is used for ground reference. The twists between wires are helpful in reducing noise (electromagnetic interference) and crosstalk.

There are two types of twisted pair cables:

- Shielded Twisted Pair (STP) Cable
- Unshielded Twisted Pair (UTP) Cable

STP cables comes with twisted wire pair covered in metal foil. This makes it more indifferent to noise and crosstalk.

UTP has seven categories, each suitable for specific use. In computer networks, Cat-5, Cat-5e, and Cat-6 cables are mostly used. UTP cables are connected by RJ45 connectors.

Coaxial Cable

Coaxial cable has two wires of copper. The core wire lies in the center and it is made of solid conductor. The core is enclosed in an insulating sheath. The second wire is wrapped around over the sheath and that too in turn encased by insulator sheath. This all is covered by plastic cover.

Because of its structure, the coax cable is capable of carrying high frequency signals than that of twisted pair cable. The wrapped structure provides it a good shield against noise and cross talk. Coaxial cables provide high bandwidth rates of up to 450 mbps.

There are three categories of coax cables namely, RG-59 (Cable TV), RG-58 (Thin Ethernet), and RG-11 (Thick Ethernet). RG stands for Radio Government. Cables are connected using BNC connector and BNC-T. BNC terminator is used to terminate the wire at the far ends.

Fiber Optics

Fiber Optic works on the properties of light. When light ray hits at critical angle it tends to refracts at 90 degree. This property has been used in fiber optic. The core of fiber optic cable is made of high quality glass or plastic. From one end of it light is emitted, it travels through it and at the other end light detector detects light stream and converts it to electric data.

Fiber Optic provides the highest mode of speed. It comes in two modes, one is single mode fiber and second is multimode fiber. Single mode fiber can carry a single ray of light whereas multimode is capable of carrying multiple beams of light.

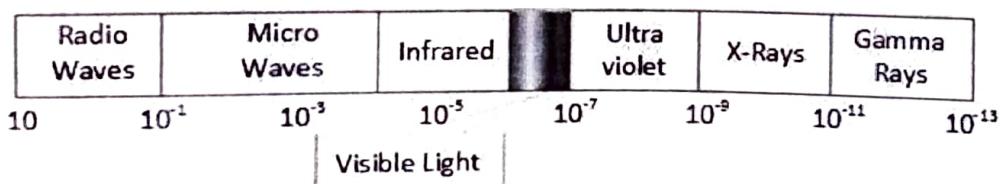
Fiber Optic also comes in unidirectional and bidirectional capabilities. To connect and access fiber optic special type of connectors are used. These can be Subscriber Channel (SC), Straight Tip (ST), or MT-RJ.

Unguided 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

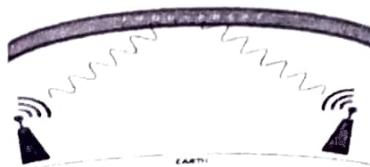
Radio frequency is easier to generate and because of its large wavelength it can penetrate through walls and structures alike. Radio waves can have wavelength from 1 mm – 100,000 km and have frequency ranging from 3 Hz (Extremely Low Frequency) to 300 GHz (Extremely High Frequency). Radio frequencies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



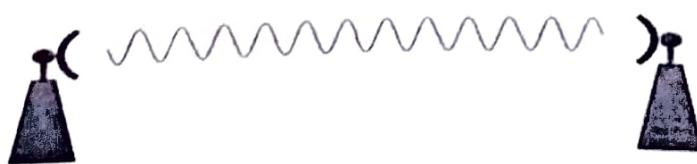
Radio waves of high frequencies are prone to be absorbed by rain and other obstacles. They use Ionosphere of earth atmosphere. High frequency radio waves such as HF and VHF bands are spread upwards. When they reach Ionosphere, they are refracted back to the earth.



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

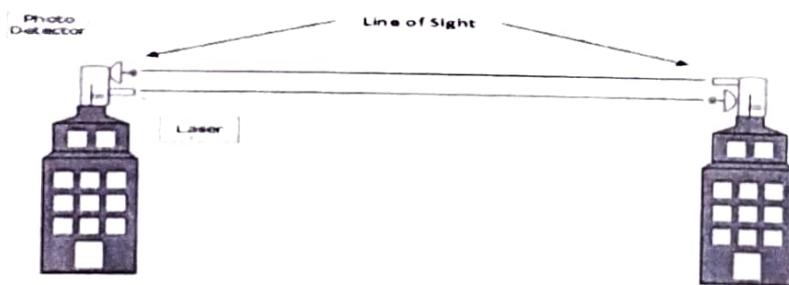
Infrared wave lies in between visible light spectrum and microwaves. It has wavelength of 700-nm to 1-mm and frequency ranges from 300-GHz to 430-THz.

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 signalling. 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.

Architecture of IEEE 802.11

IEEE 802.11:

IEEE 802.11 is a set of media access control (MAC) and physical layer (PHY) specifications for implementing wireless local area network(WLAN) computer communication in the 900 MHz and 2.4, 3.6, 5, and 60 GHz frequency bands

The IEEE developed an international standard for WLANs. The 802.11 standard focuses on the bottom two layers of the OSI model, the physical layer (PHY) and data link layer (DLL).

The objective of the IEEE 802.11 standard was to define a medium access control (MAC) sublayer, MAC management protocols and services, and three PHYs for wireless connectivity of fixed, portable, and moving devices within a local area.

The three physical layers are an IR base band PHY, an FHSS radio in the 2.4 GHz band, and a DSSS radio in the 2.4 GHz.

IEEE 802.11 Architecture:

The architecture of the IEEE 802.11 WLAN is designed to support a network where most decision making is distributed to mobile stations. This type of architecture has several advantages. It is tolerant of faults in all of the WLAN equipment and eliminates possible bottlenecks a centralized architecture would introduce. The architecture is flexible and can easily support both small, transient networks and large, semipermanent or permanent networks. In addition, the architecture and protocols offer significant power saving and prolong the battery life of mobile equipment without losing network connectivity

Two network architectures are defined in the IEEE 802.11 standard:

- **Infrastructure network:** An infrastructure network is the network architecture for providing communication between wireless clients and wired network resources. The transition of data from the wireless to wired medium occurs via an AP. An AP and its associated wireless clients define the coverage area. Together all the devices form a basic service set (refer figure 1).

- **Point-to-point (ad-hoc) network:** An ad-hoc network is the architecture that is used to support mutual communication between wireless clients. Typically, an ad-hoc network is created spontaneously and does not support access to wired networks. An ad-hoc network does not require an AP.

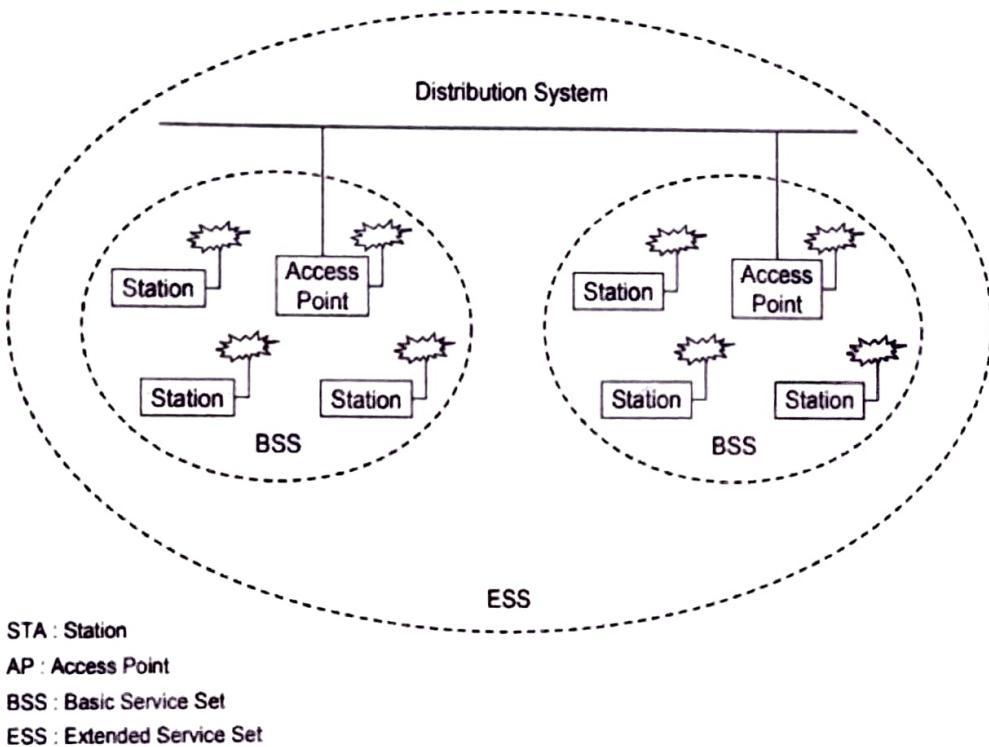


Fig1: BSS and ESS configuration of IEEE 802.11 WLAN

IEEE 802.11 supports three basic topologies for WLANs, the independent basic service set (IBSS), the basic service set, and the extended service set (ESS). The MAC layer supports implementations of IBSS, basic service set, and ESS configurations.

Independent basic service set: The IBSS configuration is referred to as an independent configuration or an ad-hoc network. An IBSS configuration is analogous to a peer-to-peer office network in which no single node is required to act as a server. IBSS WLANs include a number of nodes or wireless stations that communicate directly with one another on an ad-hoc, peer-to-peer basis. Generally, IBSS implementations cover a limited area and are not connected to any large network. An IBSS is typically a short-lived network, with a small number of stations, that is created for a particular purpose.

Basic service set: The basic service set (BSS) defines the set of APs that serve the same physical service for a single WLAN cell or channel. Communication between stations occurs in two steps: 1) data flows from station 1 to AP1 and then from AP1 to AP2 and then from AP2 to AP3 and finally AP3 to station 4 (refer to Figure 2). An AP performs a bridging function and connects multiple WLAN cells or channels, and connects WLAN cells to a wired existing LAN.

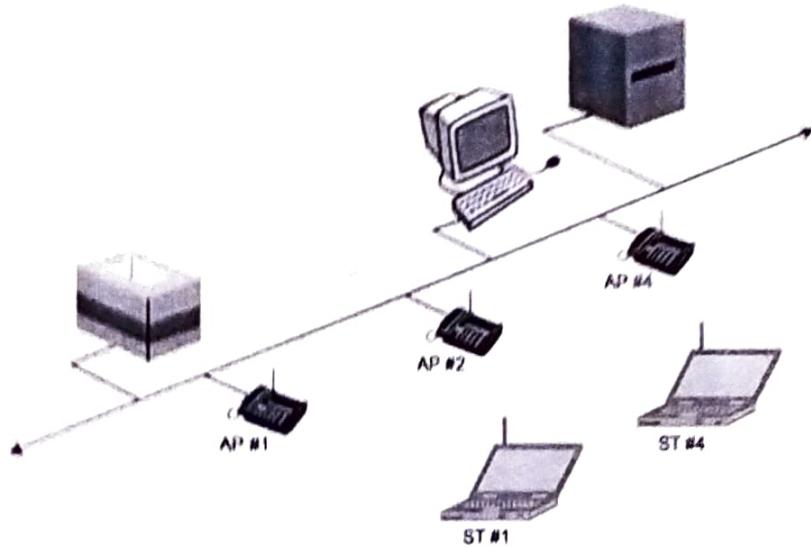


Fig.2 Access point-based topology

Extended service set: The ESS configuration consists of multiple basic service set cells that can be linked by either wired or wireless backbones called a distributed system. IEEE 802.11 supports ESS configurations in which multiple cells use the same channel to boost aggregate throughput to network. The equipment outside of the ESS, the ESS and all of its mobile stations appear to be a single MAC layer network where all stations are physically stationary. Thus, the ESS hides the mobility of the mobile stations from everything outside the ESS.

Digital Transmission

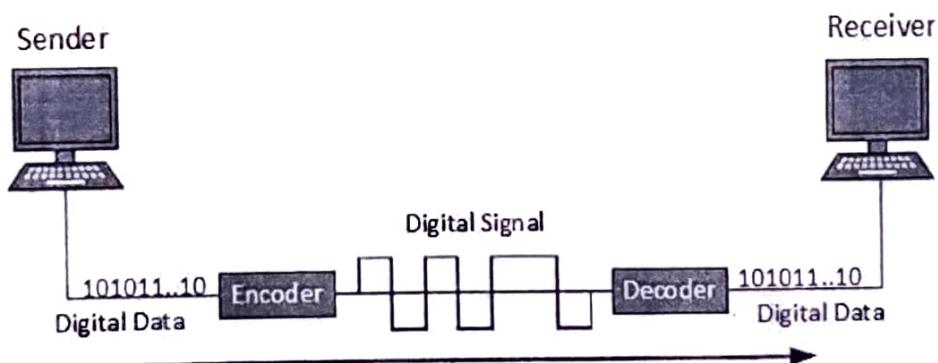
Data or information can be stored in two ways, analog and digital. For a computer to use the data, it must be in discrete digital form. Similar to data, signals can also be in analog and digital form. To transmit data digitally, it needs to be first converted to digital form.

Digital-to-Digital Conversion

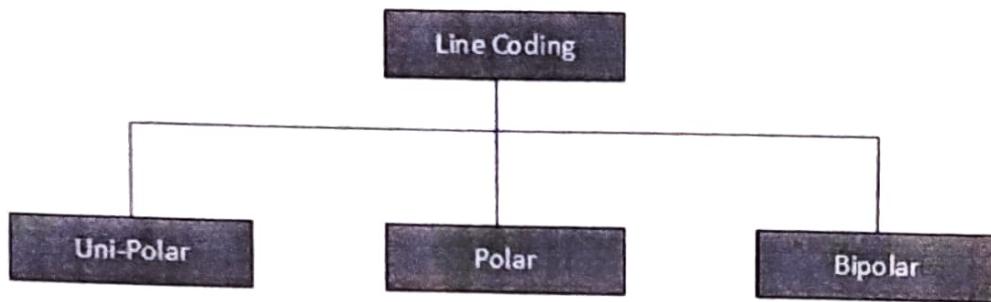
This section explains how to convert digital data into digital signals. It can be done in two ways, line coding and block coding. For all communications, line coding is necessary whereas block coding is optional.

Line Coding

The process for converting digital data into digital signal is said to be Line Coding. Digital data is found in binary format. It is represented (stored) internally as series of 1s and 0s.

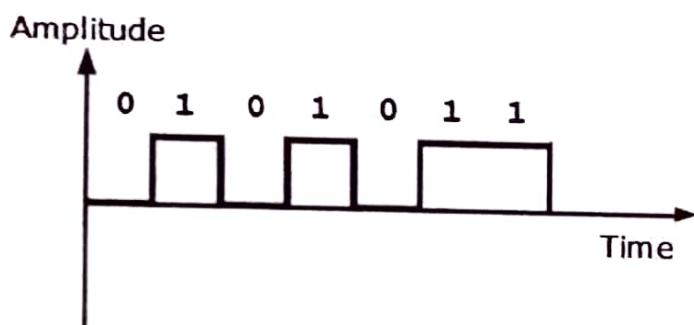


Digital signal is denoted by discreet signal, which represents digital data. There are three types of line coding schemes available:



Uni-polar Encoding

Unipolar encoding schemes use single voltage level to represent data. In this case, to represent binary 1, high voltage is transmitted and to represent 0, no voltage is transmitted. It is also called Unipolar-Non-return-to-zero, because there is no rest condition i.e. it either represents 1 or 0.



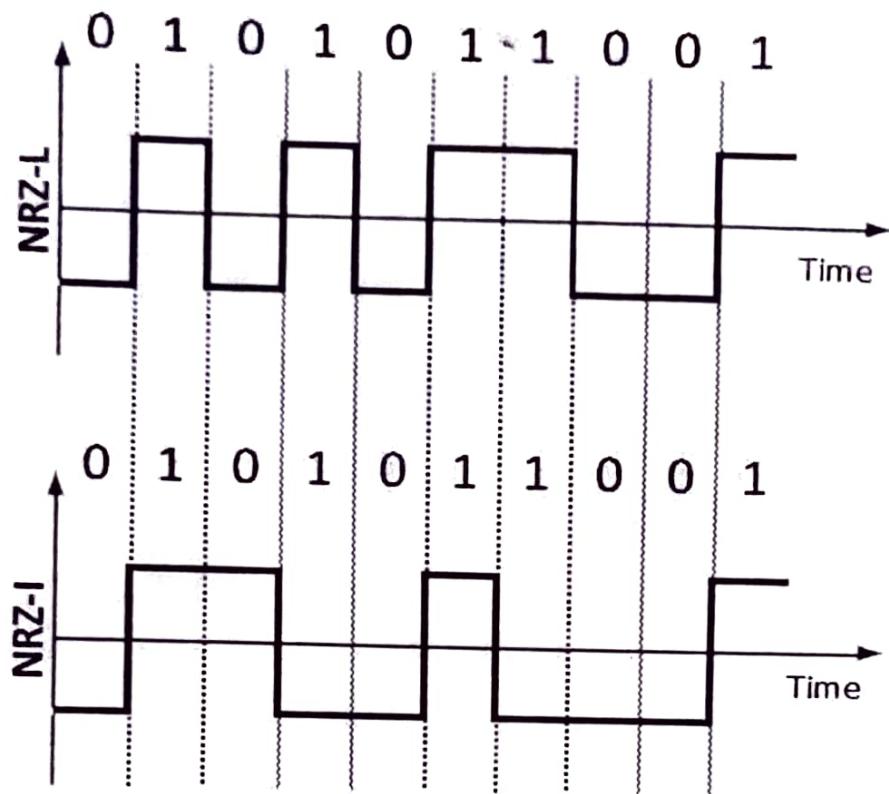
Polar Encoding

Polar encoding scheme uses multiple voltage levels to represent binary values. Polar encodings is available in four types:

- Polar Non-Return to Zero (Polar NRZ)

It uses two different voltage levels to represent binary values. Generally, positive voltage represents 1 and negative value represents 0. It is also NRZ because there is no rest condition.

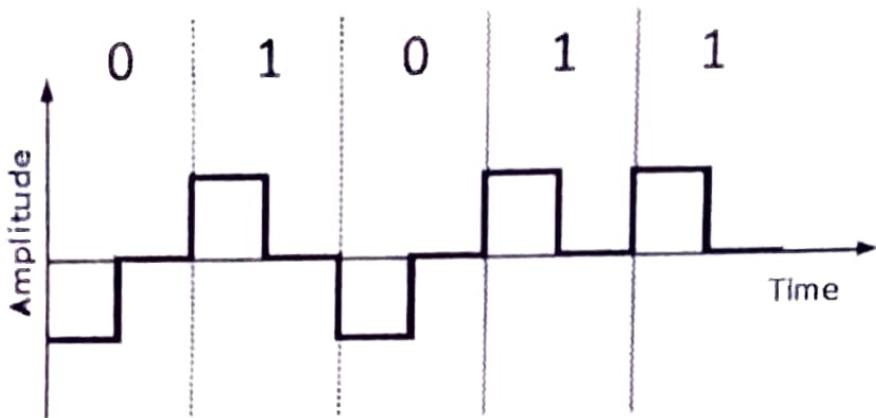
NRZ scheme has two variants: NRZ-L and NRZ-I.



NRZ-L changes voltage level at when a different bit is encountered whereas NRZ-I changes voltage when a 1 is encountered.

- Return to Zero (RZ)

Problem with NRZ is that the receiver cannot conclude when a bit ended and when the next bit is started, in case when sender and receiver's clock are not synchronized.



RZ uses three voltage levels, positive voltage to represent 1, negative voltage to represent 0 and zero voltage for none. Signals change during bits not between bits.

- **Manchester**

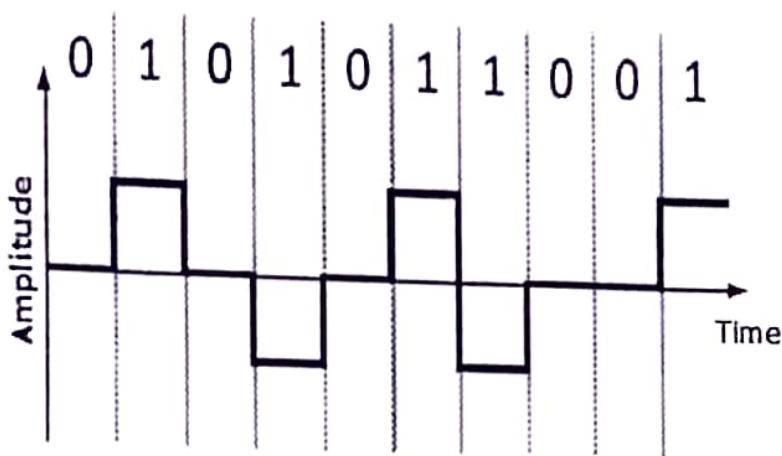
This encoding scheme is a combination of RZ and NRZ-L. Bit time is divided into two halves. It transits in the middle of the bit and changes phase when a different bit is encountered.

- **Differential Manchester**

This encoding scheme is a combination of RZ and NRZ-I. It also transit at the middle of the bit but changes phase only when 1 is encountered.

Bipolar Encoding

Bipolar encoding uses three voltage levels, positive, negative and zero. Zero voltage represents binary 0 and bit 1 is represented by altering positive and negative voltages.



Block Coding

To ensure accuracy of the received data frame redundant bits are used. For example, in even-parity, one parity bit is added to make the count of 1s in the frame even. This way the original number of bits is increased. It is called Block Coding.

Block coding is represented by slash notation, mB/nB . Means, m-bit block is substituted with n-bit block where $n > m$. Block coding involves three steps:

- Division,
- Substitution
- Combination.

After block coding is done, it is line coded for transmission.

Analog-to-Digital Conversion

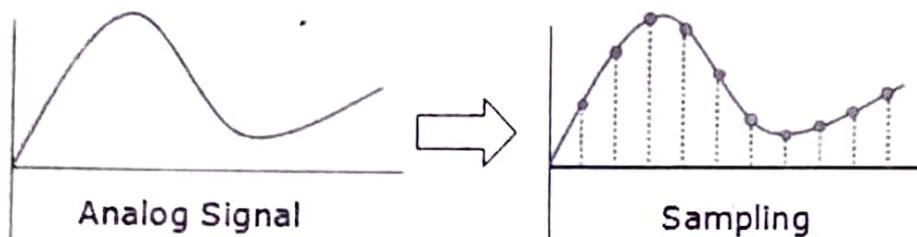
Microphones create analog voice and camera creates analog videos, which are treated as analog data. To transmit this analog data over digital signals, we need analog to digital conversion.

Analog data is a continuous stream of data in the wave form whereas digital data is discrete. To convert analog wave into digital data, we use Pulse Code Modulation (PCM).

PCM is one of the most commonly used method to convert analog data into digital form. It involves three steps:

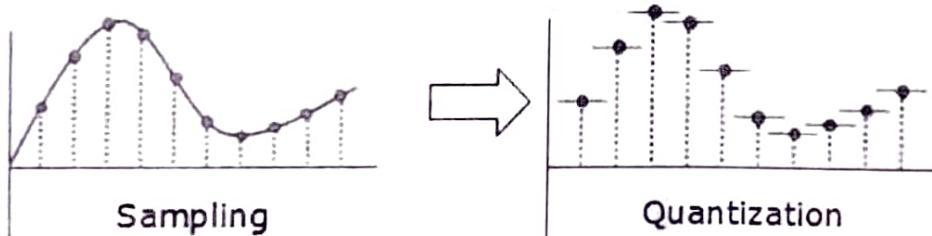
- Sampling
- Quantization
- Encoding.

Sampling



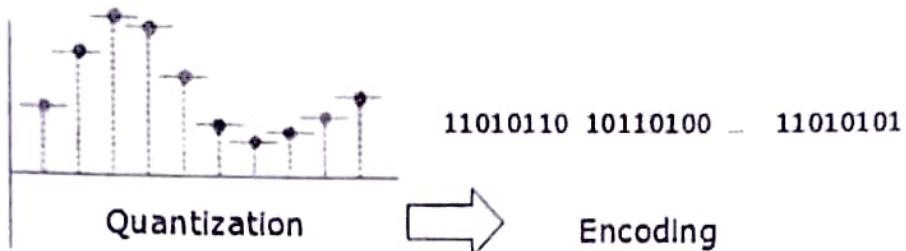
The analog signal is sampled every T interval. Most important factor in sampling is the rate at which analog signal is sampled. According to Nyquist Theorem, the sampling rate must be at least two times of the highest frequency of the signal.

Quantization



Sampling yields discrete form of continuous analog signal. Every discrete pattern shows the amplitude of the analog signal at that instance. The quantization is done between the maximum amplitude value and the minimum amplitude value. Quantization is approximation of the instantaneous analog value.

Encoding

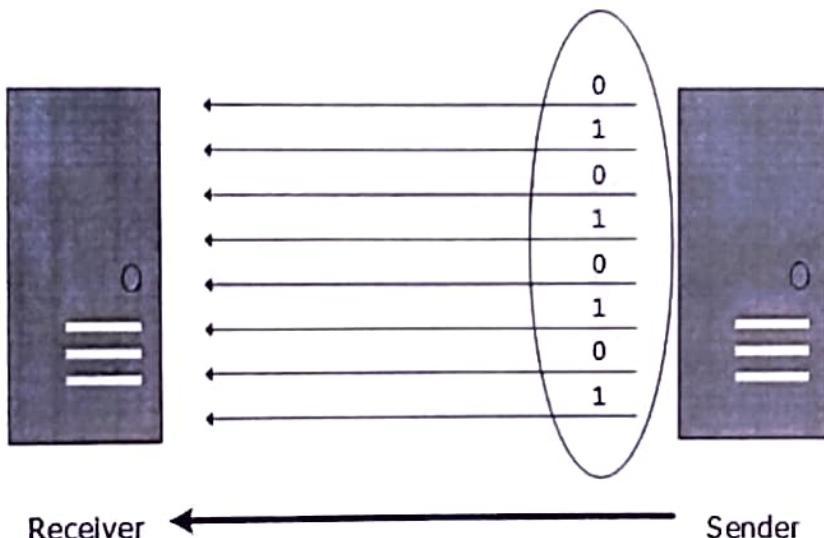


In encoding, each approximated value is then converted into binary format.

Transmission Modes

The transmission mode decides how data is transmitted between two computers. The binary data in the form of 1s and 0s can be sent in two different modes: Parallel and Serial.

Parallel Transmission

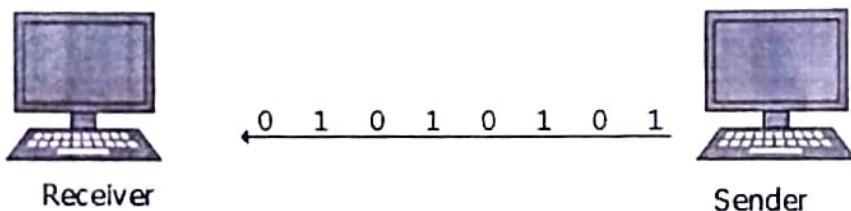


The binary bits are organized into groups of fixed length. Both sender and receiver are connected in parallel with the equal number of data lines. Both computers distinguish between high order and low order data lines. The sender sends all the bits at once on all lines. Because the data lines are equal to the number of bits in a group or data frame, a complete group of

bits (data frame) is sent in one go. Advantage of Parallel transmission is high speed and disadvantage is the cost of wires, as it is equal to the number of bits sent in parallel.

Serial Transmission

In serial transmission, bits are sent one after another in a queue manner. Serial transmission requires only one communication channel.



Serial transmission can be either asynchronous or synchronous.

Asynchronous Serial Transmission

It is named so because there's no importance of timing. Data-bits have specific pattern and they help receiver recognize the start and end data bits. For example, a 0 is prefixed on every data byte and one or more 1s are added at the end.

Two continuous data-frames (bytes) may have a gap between them.

Synchronous Serial Transmission

Timing in synchronous transmission has importance as there is no mechanism followed to recognize start and end data bits. There is no pattern or prefix/suffix method. Data bits are sent in burst mode without maintaining gap between bytes (8-bits). Single burst of data bits may contain a number of bytes. Therefore, timing becomes very important.

It is up to the receiver to recognize and separate bits into bytes. The advantage of synchronous transmission is high speed, and it has no overhead of extra header and footer bits as in asynchronous transmission.

Analog Transmission

To send the digital data over an analog media, it needs to be converted into analog signal. There can be two cases according to data formatting.

Bandpass: The filters are used to filter and pass frequencies of interest. A bandpass is a band of frequencies which can pass the filter.

Low-pass: Low-pass is a filter that passes low frequencies signals.

When digital data is converted into a bandpass analog signal, it is called digital-to-analog conversion. When low-pass analog signal is converted into bandpass analog signal, it is called analog-to-analog conversion.

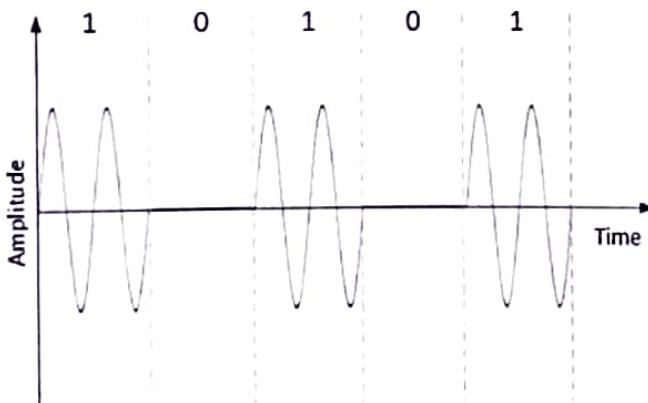
Digital-to-Analog Conversion

When data from one computer is sent to another via some analog carrier, it is first converted into analog signals. Analog signals are modified to reflect digital data.

An analog signal is characterized by its amplitude, frequency, and phase. There are three kinds of digital-to-analog conversions:

- **Amplitude Shift Keying**

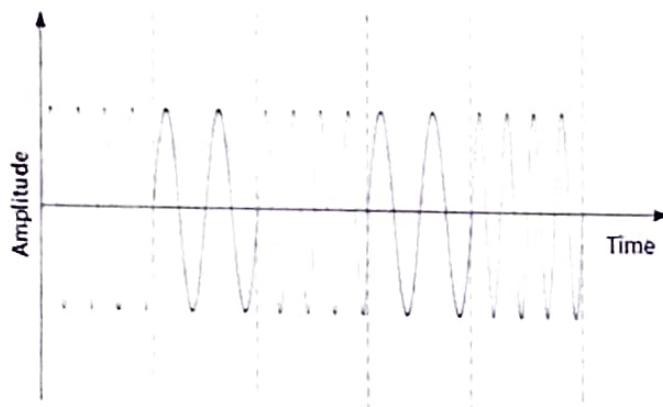
In this conversion technique, the amplitude of analog carrier signal is modified to reflect binary data.



When binary data represents digit 1, the amplitude is held; otherwise it is set to 0. Both frequency and phase remain same as in the original carrier signal.

- **Frequency Shift Keying**

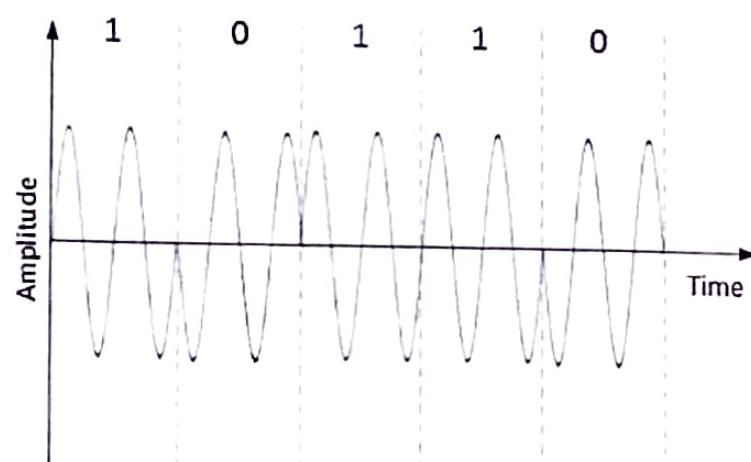
In this conversion technique, the frequency of the analog carrier signal is modified to reflect binary data.



This technique uses two frequencies, f_1 and f_2 . One of them, for example f_1 , is chosen to represent binary digit 1 and the other one is used to represent binary digit 0. Both amplitude and phase of the carrier wave are kept intact.

- **Phase Shift Keying**

In this conversion scheme, the phase of the original carrier signal is altered to reflect the binary data.



When a new binary symbol is encountered, the phase of the signal is altered. Amplitude and frequency of the original carrier signal is kept intact.

- **Quadrature Phase Shift Keying**

QPSK alters the phase to reflect two binary digits at once. This is done in two different phases. The main stream of binary data is divided equally into two sub-streams. The serial data is converted in to parallel in both sub-streams and then each stream is converted to digital signal using NRZ technique. Later, both the digital signals are merged together.

Multiplexing

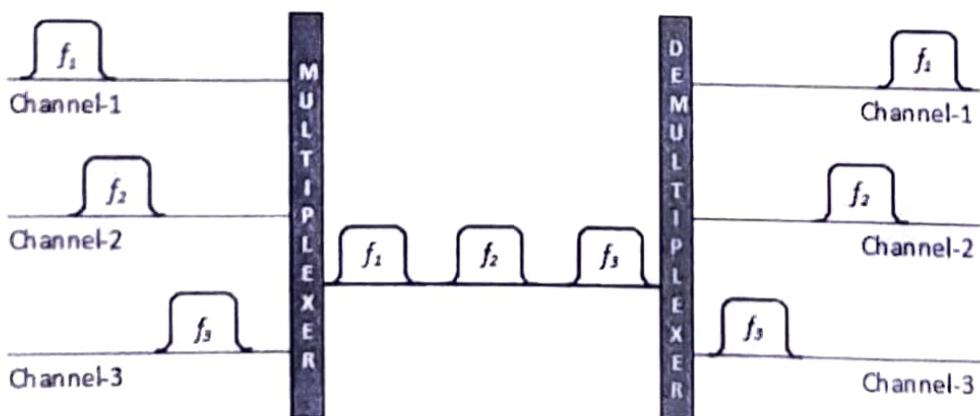
Multiplexing is a technique by which different analog and digital streams of transmission can be simultaneously processed over a shared link. Multiplexing divides the high capacity medium into low capacity logical medium which is then shared by different streams.

Communication is possible over the air (radio frequency), using a physical media (cable), and light (optical fiber). All mediums are capable of multiplexing.

When multiple senders try to send over a single medium, a device called Multiplexer divides the physical channel and allocates one to each. On the other end of communication, a De-multiplexer receives data from a single medium, identifies each, and sends to different receivers.

Frequency Division Multiplexing

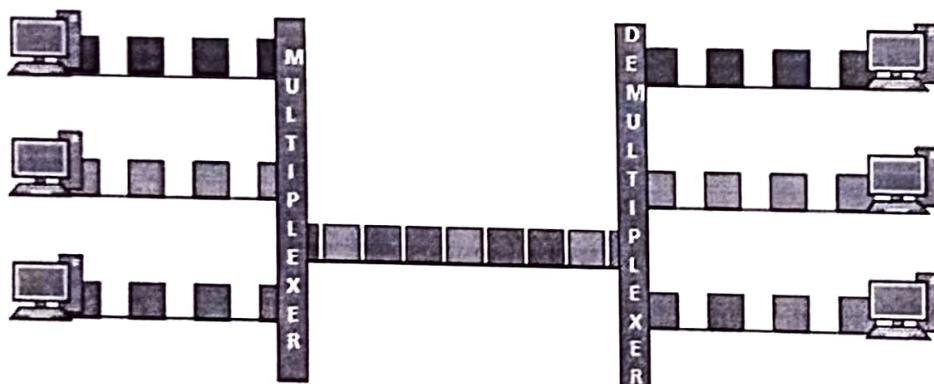
When the carrier is frequency, FDM is used. FDM is an analog technology. FDM divides the spectrum or carrier bandwidth in logical channels and allocates one user to each channel. Each user can use the channel frequency independently and has exclusive access of it. All channels are divided in such a way that they do not overlap with each other. Channels are separated by guard bands. Guard band is a frequency which is not used by either channel.



Time Division Multiplexing

TDM is applied primarily on digital signals but can be applied on analog signals as well. In TDM the shared channel is divided among its user by means of time slot. Each user can transmit data within the provided time slot only. Digital signals are divided in frames, equivalent to time slot i.e. frame of an optimal size which can be transmitted in given time slot.

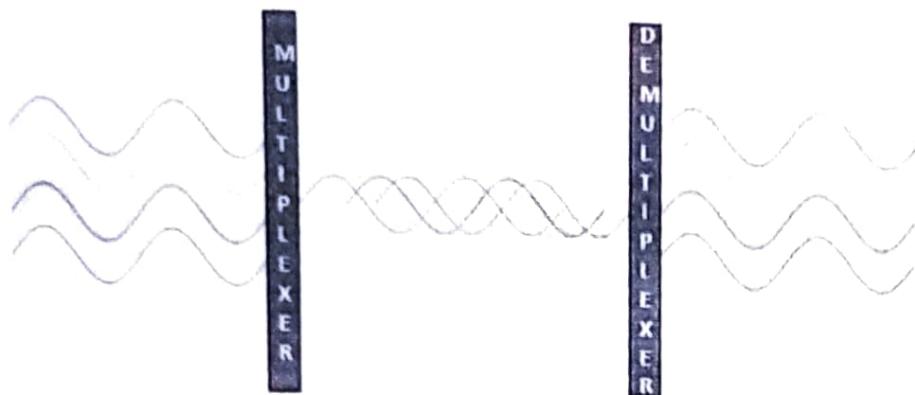
TDM works in synchronized mode. Both ends, i.e. Multiplexer and De-multiplexer are timely synchronized and both switch to next channel simultaneously.



When channel A transmits its frame at one end, the De-multiplexer provides media to channel A on the other end. As soon as the channel A's time slot expires, this side switches to channel B. On the other end, the De-multiplexer works in a synchronized manner and provides media to channel B. Signals from different channels travel the path in interleaved manner.

Wavelength Division Multiplexing

Light has different wavelength (colors). In fiber optic mode, multiple optical carrier signals are multiplexed into an optical fiber by using different wavelengths. This is an analog multiplexing technique and is done conceptually in the same manner as FDM but uses light as signals.



Further, on each wavelength time division multiplexing can be incorporated to accommodate more data signals.