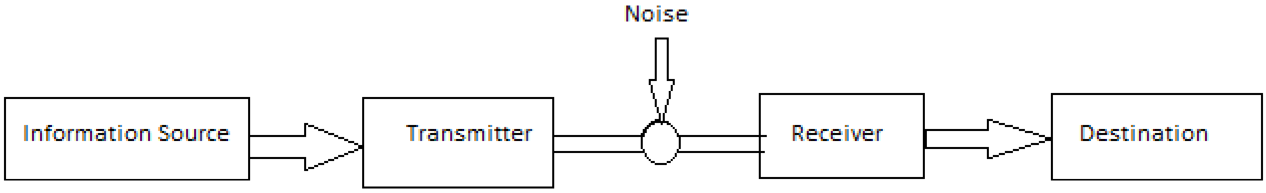
1. Draw a generic block diagram of a communication system and explain its working.

The block diagram of a generic communication system is given in **Fig(1)**



Fig(1) Block diagram of a generic communication system

A generic communication system consists of five elements/blocks/ components are:-

i. Information source (Input transducer)
ii. Transmitter
iii. Transmission medium/ Channel
iv. Receiver
v. Destination

i. Information source (Input transducer)
- Generates information to be transmitted(produces message)
- Can be analog or digital

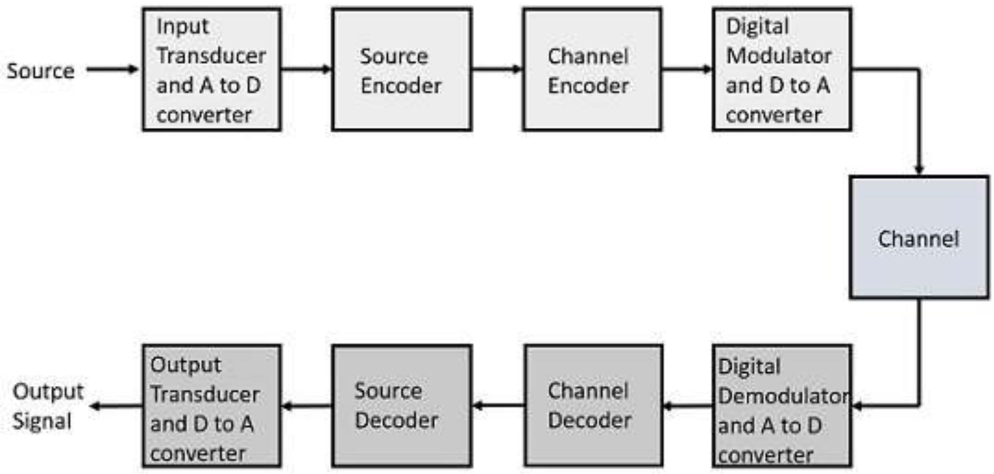
ii. Transmitter
- Receives a signal from the information source
- Processes the signal in such a way that it is compatible with the transmission medium
- Compatibility ensures low loss/ attenuation and low distortion of the signal(noise)
- Processes could be – Modulation, Amplification, Filtering, Multiplexing

iii. Transmission medium/ Channel
- Provides the path for the propagation of a signal from transmission to receiver
- Path could be
  - Guided/ Wired transmission medium: twisted pair, copper wire, coaxial cable, optical fiber
  - Unguided: Terrestrial microwave, Satellite, Radiowave, Infrared
- Characteristics of transmission medium are-
  - Bandwidth or Frequency response
  - Structure : cylindrical or not
  - Noise – attenuation, bandwidth loss

iv. Receiver
- Receives signal from the transmission medium
- Processes the signal to make it compatible for the destination
- Processes could be demodulation

v. Destination
- The final element where the information is delivered.

2. Draw a block diagram of a digital communication system and explaint its working.



Basic Elements of a Digital Communication System

- **Source:** The source can be an **analog** signal. **Example**: A Sound signal
- **Input Transducer:** This is a transducer which takes a physical input and converts it to an electrical signal (**Example**: microphone). This block also consists of an **analog to digital** converter where a digital signal is needed for further processes.
- A digital signal is generally represented by a binary sequence.
- **Source Encoder:** The source encoder compresses the data into minimum number of bits. This process helps in effective utilization of the bandwidth. It removes the redundant bits (unnecessary excess bits, i.e., zeroes).
- **Channel Encoder:** The channel encoder, does the coding for error correction. During the transmission of the signal, due to the noise in the channel, the signal may get altered and hence to avoid this, the channel encoder adds some redundant bits to the transmitted data. These are the error correcting bits.
- **Digital Modulator:** The signal to be transmitted is modulated here by a carrier. The signal is also converted to analog from the digital sequence, in order to make it travel through the channel or medium.
- **Channel:** The channel or a medium, allows the analog signal to transmit from the transmitter end to the receiver end.
- **Digital Demodulator:** This is the first step at the receiver end. The received signal is demodulated as well as converted again from analog to digital. The signal gets reconstructed here.
- **Channel Decoder:** The channel decoder, after detecting the sequence, does some error corrections. The distortions which might occur during the transmission, are corrected by adding some redundant bits. This addition of bits helps in the complete recovery of the original signal.
- **Source Decoder:** The resultant signal is once again digitized by sampling and quantizing so that the pure digital output is obtained without the loss of information. The source decoder recreates the source output.
- **Output Transducer:** This is the last block which converts the signal into the original physical form, which was at the input of the transmitter. It converts the electrical signal into physical output (**Example**: loud speaker).
- **Output Signal:** This is the output which is produced after the whole process. **Example** − The sound signal received.
- This unit has dealt with the introduction, the digitization of signals, the advantages and the elements of digital communications. In the coming chapters, we will learn about the concepts of Digital communications, in detail.

3. Compare between Analog and Digital communication system.

SN	Attributes	Analog CS	Digital CS	Remarks
1	Information generated by source	Analog	Digital or analog	In Digital CS If source is analog, it will be digitized
2	Source coding	Not performed	Performed to make source represent efficient	
3.	Channel coding	Not performed	Performed	DC for error detection and correction
4	Modulation	Analog modulation such as AM, FM, PM, DM	Digital modulation such as ASK, FSK, PSK, QAM	For digital modulation bot binary and m-Aray modulation
5	Security (Encryption/ Decryption)	Not possible	Possible for security	For secure communication
6	Amplification	Amplification is done to bust the signal strength	No amplification is required uses required regeneration	
7	Impact of Noise	During amplification noise also amplify	Due to regeneration, noise is completely removed	
8	Jitter	There is no jitter	Jitter is main limiting factor impact in regeneration	
9	Multiplexing	FDM	TDM, CDMA	
10	Bandwidth required	Less bandwidth if analog signal converts to digital using waveform coding using PCM. If any other coding techniques/ digital coding used the BW required can be lower than analog signal compression techniques		
11	Power required	Higher	Less	

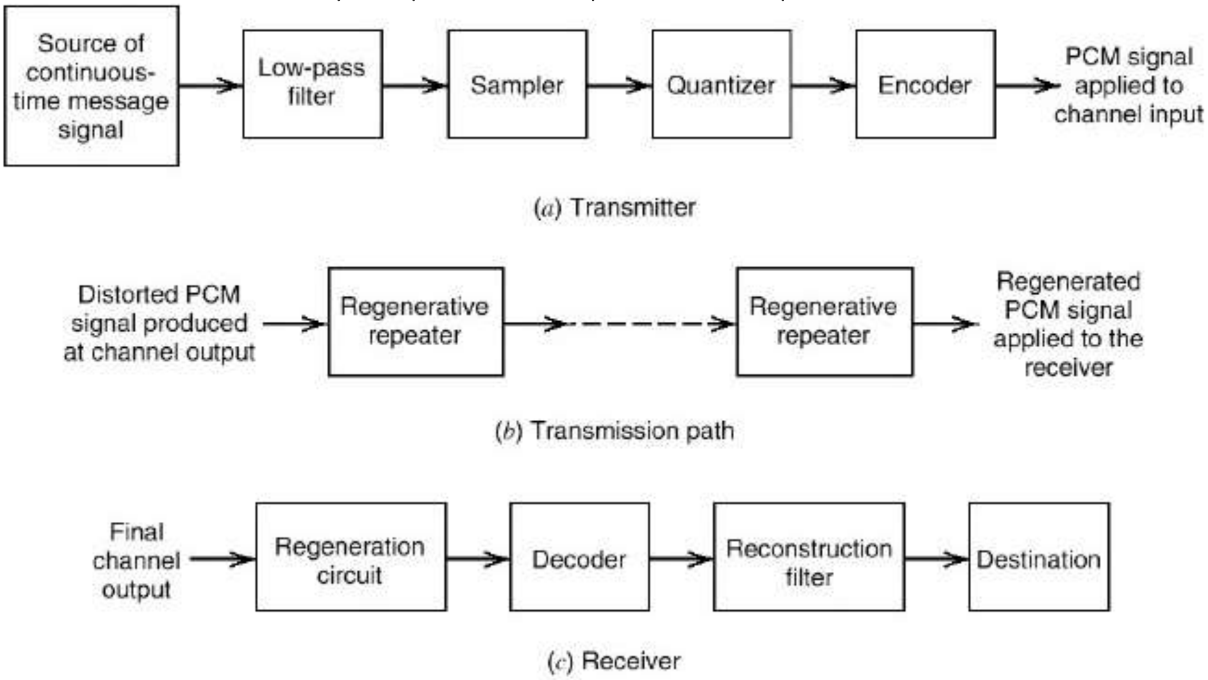
12	Distance covered	Effect of noise will cause effect in long distance	Using regeneration digital communication can cover very large scale, only limiting by jitter
13	Cost	Costly	Cheaper
14	Size	Larger size	Compact in small chips
15	Delaying	More delaying	Not delaying

4. What is frequency?

- Spectrum or frequency is a scarce natural resources i.e. very limited
- It is a parameter generated by generator

5. Explain PCM as a waveform coding techniques.

To get a pulse code modulated waveform from an analog waveform at the transmitter end (source) of a communications circuit, the amplitude of the analog signal samples at regular time intervals. The sampling rate or number of samples per second is several times the maximum frequency. The message signal converted into binary form will be usually in the number of levels which is always to a power of 2. This process is called quantization.



Basic Elements of PCM System

At the receiver end, a pulse code demodulator decodes the binary signal back into pulses with same quantum levels as those in the modulator. By further processes we can restore the original analog waveform.

Low pass Filter

The LPF before the smoothing filter makes sure only the information signal within the bandwidth of interest is captured. Due to the multiplication of sinusoidal signals, integer multiples of the carrier frequency (besides the one centered at 0 Hz) are generated. These high frequency components are not needed and the LPF removes those. Your signal is constructed by the smoothing filter that follows the LPF. The smoothing filter is also a type of low pass filter. Depending on its depth (i.e. the memory), you can adjust the output response (i.e. more depth means less high frequency components are allowed through)

Sampling

Sampling is a process of measuring the amplitude of a continuous-time signal at discrete instants, converts the continuous signal into a discrete signal. For example, conversion of a sound wave to a sequence of samples. The Sample is a value or set of values at a point in time or it can be spaced. Sampler extract samples of a continuous signal, it is a subsystem ideal sampler produces samples which are equivalent to the instantaneous value of the continuous signal at the specified various points. The Sampling process generates flat- top Pulse Amplitude Modulated (PAM) signal.

Sampling frequency, Fs is the number of average samples per second also known as Sampling rate. According to the Nyquist Theorem sampling rate should be at least 2 times the upper cutoff frequency. Sampling frequency, Fs>=2*fmax to avoid Aliasing Effect. If the sampling frequency is very higher than the Nyquist rate it become Oversampling, theoretically a bandwidth limited signal can be reconstructed if sampled at above the Nyquist rate. If the sampling frequency is less than the Nyquist rate it will become Undersampling.

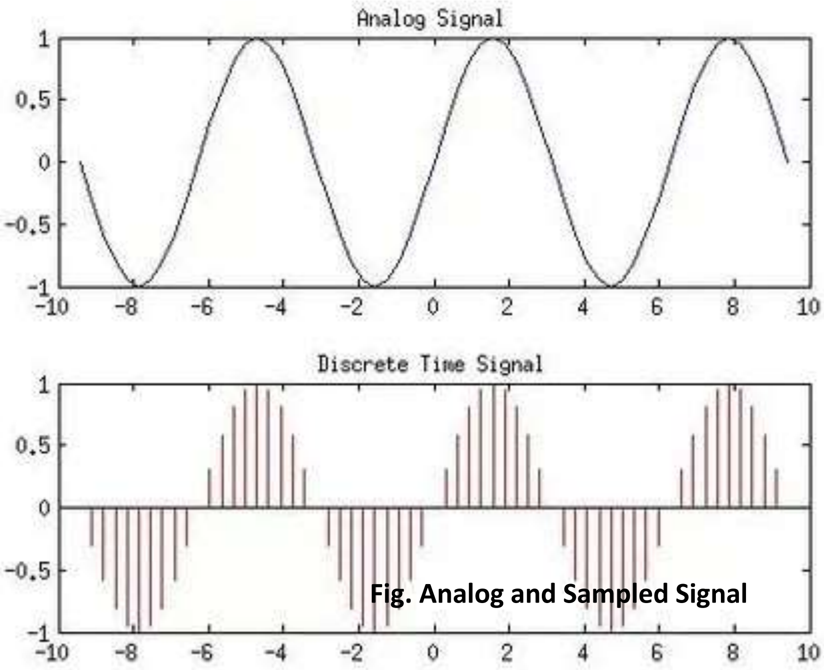


Fig. Analog and Sampled Signal

Basically two types of techniques are used for the sampling process. Those are 1. Natural Sampling and 2. Flat- top Sampling.

Quantization

In quantization, an analog sample with an amplitude that converted into a digital sample with an amplitude that takes one of a specific defined set of quantization values. Quantization is done by dividing the range of possible values of the analog samples into some different levels, and assigning the center value of each level to any sample in quantization interval. Quantization approximates the analog sample values with the nearest quantization values. So almost all the quantized samples will differ from the original samples by a small amount. That amount is called as quantization error. The result of this quantization error is we will hear hissing noise when play a random signal. Converting analog samples into binary numbers that is 0 and 1.

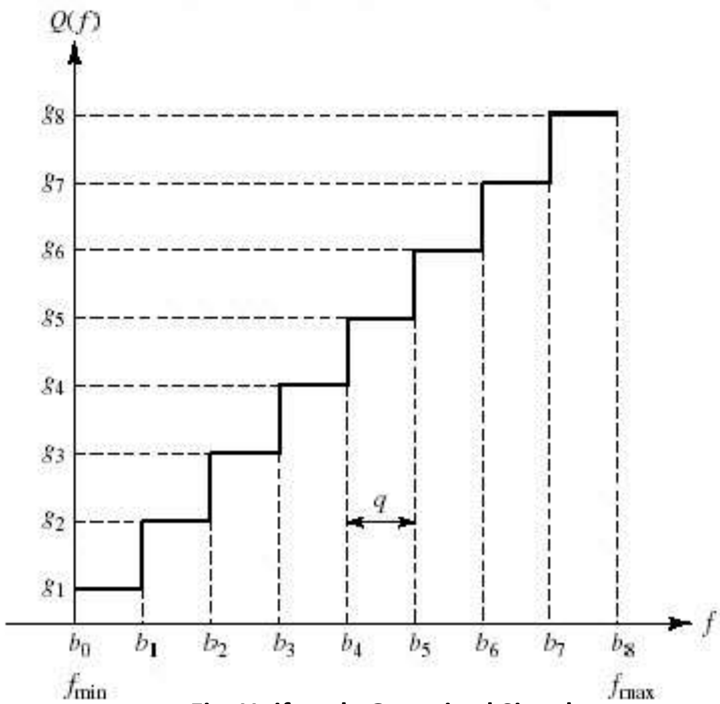


Fig. Uniformly Quantized Signal

In most of the cases we will use uniform quantizers. Uniform quantization is applicable when the sample values are in a finite range (Fmin, Fmax). The total data range is divided into 2n levels, let it be L intervals. They will have an equal length Q. Q is known as Quantization interval or quantization step size. In uniform quantization there will be no quantization error.

As we know, L=2n, then Step size Q = (Fmax – Fmin) / L

Interval i is mapped to the middle value. We will store or send only index value of quantized value.

An Index value of quantized value Qi (F) = [F – Fmin / Q]

Quantized value Q (F) = Qi (F) Q + Q / 2 + Fmin

But there are some problems raised in uniform quantization those are

- Only optimal for uniformly distributed signal.
- Real audio signals are more concentrated near zeros.
- The Human ear is more sensitive to quantization errors at small values.

The solution for this problem is using Non- uniform quantization. In this Process quantization interval is smaller near zero.

Coding

The encoder encodes the quantized samples. Each quantized sample is encoded into an 8-bit code word by using A-law in the encoding process.

- Bit 1 is the most significant bit (MSB), it represents the polarity of the sample. "1" represents positive polarity and "0" represents negative polarity.

- Bit 2,3 and 4 will defines the location of sample value. These three bits together form linear curve for low level negative or positive samples.
- Bit 5,6,7 and 8 are the least significant bits (LSB) it represents one of the segments quantized value. Each segment is divided into 16 quantum levels.

PCM is two types Differential Pulse Code Modulation (DPCM) and Adaptive Differential Pulse Code Modulation (ADPCM).

In DPCM only the difference between a sample and the previous value is encoded. The difference will be much smaller than the total sample value so we need some bits for getting same accuracy as in ordinary PCM. So that the required bit rate will also reduce. For example, in 5 bit code 1 bit is for polarity and the remaining 4 bits for 16 quantum levels.

ADPCM is achieved by adapting the quantizing levels to analog signal characteristics. We can estimate the values with preceding sample values. Error estimation is done as same as in DPCM. In 32Kbps ADPCM method difference between predicted value and sample value is coded with 4 bits, so that we'll get 15 quantum levels. In this method data rate is half of the conventional PCM.

6. Compare PCM, DPCM, ADPCM, DM and ADM.

SN	Attributes	PCM	DPCM	DM	ADM
1	**No of Bits**	4, 8, or 16	Bits can be more than one but less than PCM	Only one bit	Only one bit is used to encode one sample
2	**No of levels, step size**	Depend on no. of bits, level size is fixed.	Fixed no of levels is used	Step size is fixed and cannot be varied	According to signal variation, step size varies (Adapted)
3	**Quantization error and distortion**	Depends on no. of levels used	Slope overload distortion and quantization noise is present	Slope overload distortion and granular noise is present	Quantization error is present but other errors are absent
4	**Bandwidth of transmission channel**	Highest BW is required since no of bits are high	BW required is lesser than PCM	Lowest BW is required	Lowest BW is required
5	**Feedback**	No feedback in transmitter or receiver	Feedback exists	Feedback exists in transmitter	Feedback exists
6	**Complexity of notation**	Complex	Simple	Simple	Simple
7	**SNR**	Good	Fair	Poor	Better than DM
8	**Area of application**	Audio and video telephony	Speech and Video	Speech and Images	Speech and Images
9	**Sampling rate**	8	8	64-128	48-64
10	**Bits/sample**	7-8	4-8	1	1
11	**Bit rate**	56-64	32-48	64-128	46-64
12	**Storage**	Higher	Lesser	Lesser	Lesser
13	**Cost**	Higher	Lesser	Lesser	Lesser

7. What are the desirable characteristics of a good Line Code?

Line coding is the process of converting digital data to digital signals.
The data may be in the form of text, numbers, graphical images, audio, or video, are stored in computer memory as sequences of bits.
Line coding converts a sequence of bits to a digital signal. At the sender, digital data are encoded into a digital signal; at the receiver, the digital data are recreated by decoding the digital signal.

Characteristics of Line coding

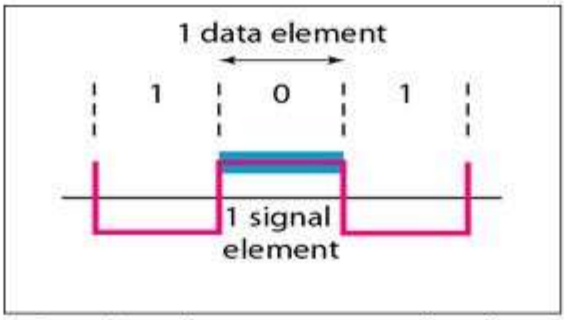
The different characteristics of Line Coding Technique are as follows:

1. Signal Element versus Data Element:

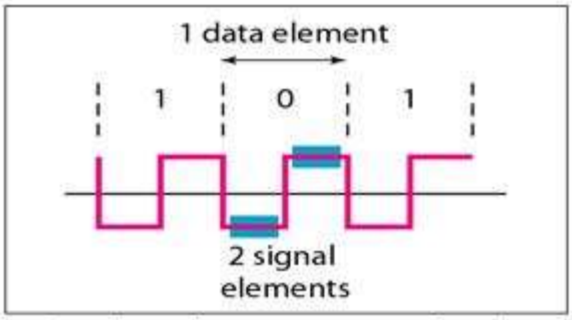
A data element is the smallest entity that can represent a piece of information. This is the bit. In digital data communications, a signal element carries data elements. A signal element is the shortest unit (time wise) of a digital signal. In other words, data elements are what we need to send; signal elements are what we can send. Data elements are being carried; signal elements are the carriers.

We define a ratio r which is the number of data elements carried by each signal element. The shows several situations with different values of r.

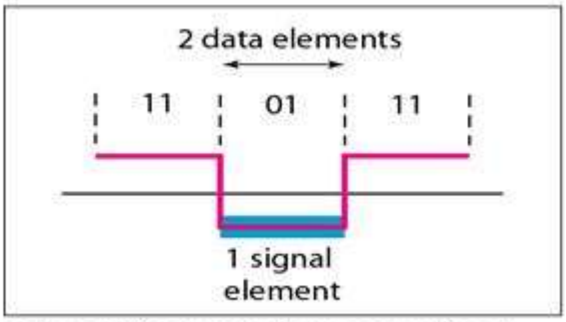
In part a of the figure, one data element is carried by one signal element (r = 1). In part b of the figure, we need two signal elements (two transitions) to carry each data element (r =1/2). In part c of the figure, a signal element carries two data elements (r = 2). In part d, a group of 4 bits is being carried by a group of three signal elements (r = 4/3). For every line coding scheme r value should be defined.



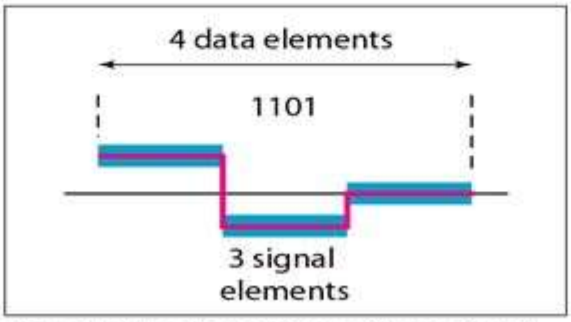
a. One data element per one signal element (r = 1)



b. One data element per two signal elements $\left(r = \frac{1}{2}\right)$



c. Two data elements per one signal element (r = 2)



d. Four data elements per three signal elements $\left(r = \frac{4}{3}\right)$

2. Data Rate versus Signal Rate:

The data rate defines the number of data elements (bits) sent in 1s. The unit is bits per second (bps). The signal rate is the number of signal elements sent in 1s. The unit is the baud. The data rate is sometimes called the bit rate; the signal rate is sometimes called the pulse rate, the modulation rate, or the baud rate.

One goal in data communications is to increase the data rate while decreasing the signal rate. Increasing the data rate increases the speed of transmission; decreasing the signal rate decreases the bandwidth requirement.

3. Bandwidth:

Digital signal that carries information is non-periodic. The bandwidth of a non-periodic signal is continuous with an infinite range. However, most digital signals we encounter in real life have a bandwidth with finite values. In other words, the bandwidth is theoretically infinite, but many of the components have such a small amplitude that they can be ignored. The effective bandwidth is finite.

4. Baseline Wandering:

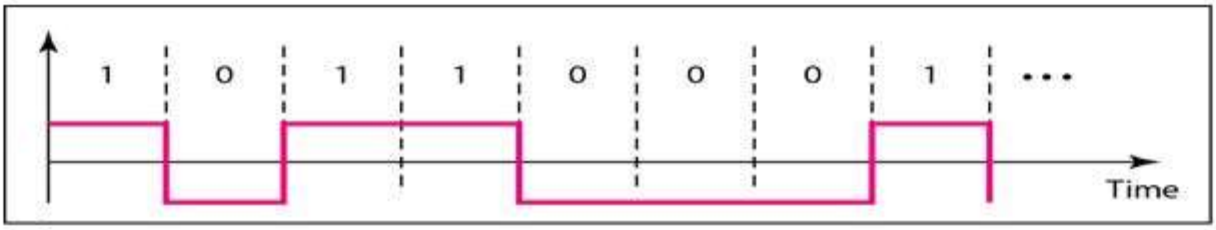
In decoding a digital signal, the receiver calculates a running average of the received signal power. This average is called the baseline. The incoming signal power is evaluated against this baseline to determine the value of the data element. A long string of Os or 1s can cause a drift in the baseline (baseline wandering) and make it difficult for the receiver to decode correctly. A good line coding scheme needs to prevent baseline wandering.

5.DC Components:

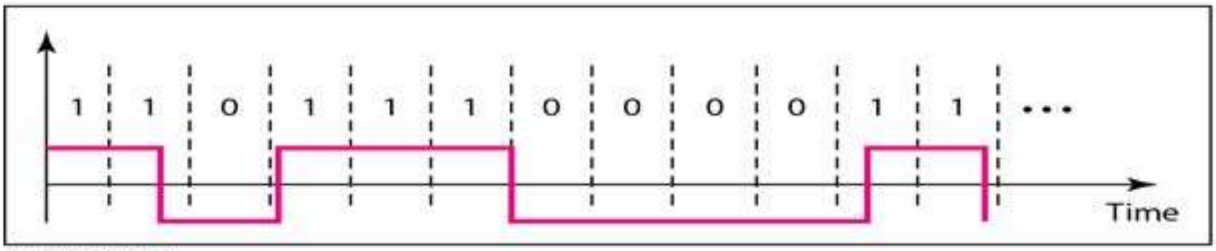
When the voltage level in a digital signal is constant for a while, the spectrum creates very low frequencies (results of Fourier analysis). These frequencies around zero, called DC (direct-current) components, present problems for a system that cannot pass low frequencies or a system that uses electrical coupling (via a transformer).

6. Self-synchronization:

To correctly interpret the signals received from the sender, the receiver's bit intervals must correspond exactly to the sender's bit intervals. If the receiver clock is faster or slower, the bit intervals are not matched and the receiver might misinterpret the signals. The following figure represents the synchronization problem.



a. Sent



b. Received

7. Built-in Error Detection:
It is desirable to have a built-in error-detecting capability in the generated code to detect some of or all the errors that occurred during transmission. Some encoding schemes that we will discuss have this capability to some extent.

8. Immunity to Noise and Interference:
Another desirable code characteristic is a code that is immune to noise and other interferences.

9. Complexity:
A complex scheme is more costly to implement than a simple one. For example, a scheme that uses four signal levels is more difficult to interpret than one that uses only two levels.

8. Types of line code

Types of Line Coding

There are 3 types of Line Coding

- Unipolar
- Polar
- Bi-polar

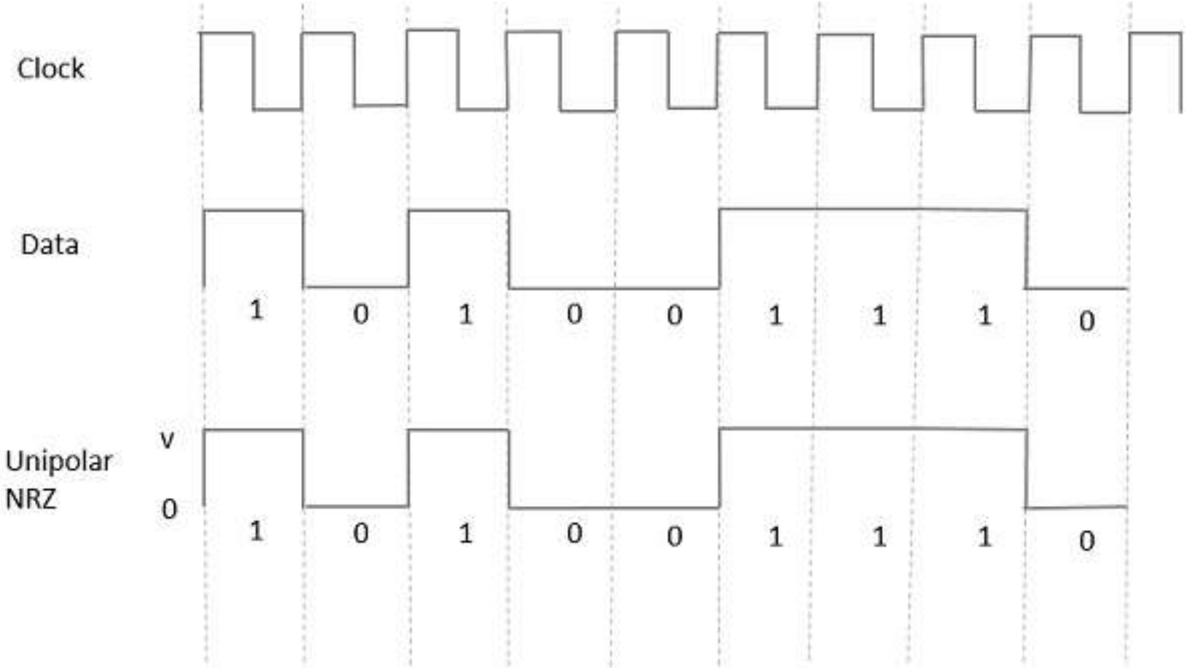
Unipolar Signaling

- Unipolar signaling is also called as **On-Off Keying** or simply **OOK**.
- The presence of pulse represents a **1** and the absence of pulse represents a **0**.
- There are two variations in Unipolar signaling −
  - Non Return to Zero (NRZ)
  - Return to Zero (RZ)

Unipolar Non-Return to Zero (NRZ)

In this type of unipolar signaling, a High in data is represented by a positive pulse called as **Mark**, which has a duration $\mathbf{T_0}$ equal to the symbol bit duration. A Low in data input has no pulse.

The following figure clearly depicts this.



Advantages

The advantages of Unipolar NRZ are −

- It is simple.
- A lesser bandwidth is required.

Disadvantages

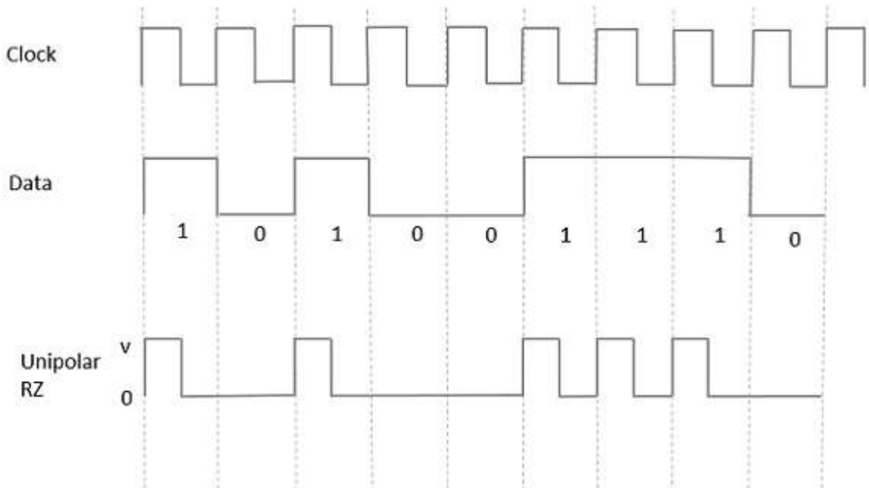
The disadvantages of Unipolar NRZ are −

- No error correction done.
- Presence of low frequency components may cause the signal droop.
- No clock is present.
- Loss of synchronization is likely to occur (especially for long strings of **1s** and **0s**).

Unipolar Return to Zero (RZ)

In this type of unipolar signaling, a High in data, though represented by a **Mark pulse**, its duration T_0 is less than the symbol bit duration. Half of the bit duration remains high but it immediately returns to zero and shows the absence of pulse during the remaining half of the bit duration.

It is clearly understood with the help of the following figure.



Advantages

The advantages of Unipolar RZ are −

- It is simple.
- The spectral line present at the symbol rate can be used as a clock.

Disadvantages

The disadvantages of Unipolar RZ are −

- No error correction.
- Occupies twice the bandwidth as unipolar NRZ.
- The signal droop is caused at the places where signal is non-zero at 0 Hz.

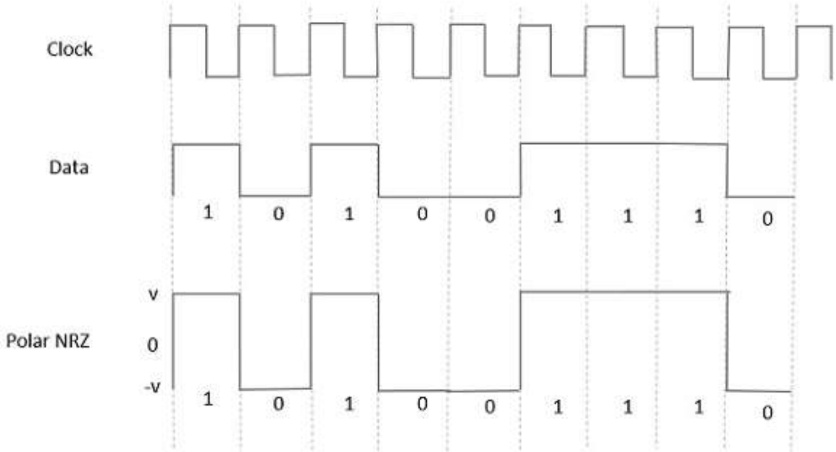
Polar Signaling

There are two methods of Polar Signaling. They are −

- Polar NRZ
- Polar RZ

Polar NRZ

In this type of Polar signaling, a High in data is represented by a positive pulse, while a Low in data is represented by a negative pulse. The following figure depicts this well.



Advantages

The advantages of Polar NRZ are −

- It is simple.
- No low-frequency components are present.

Disadvantages

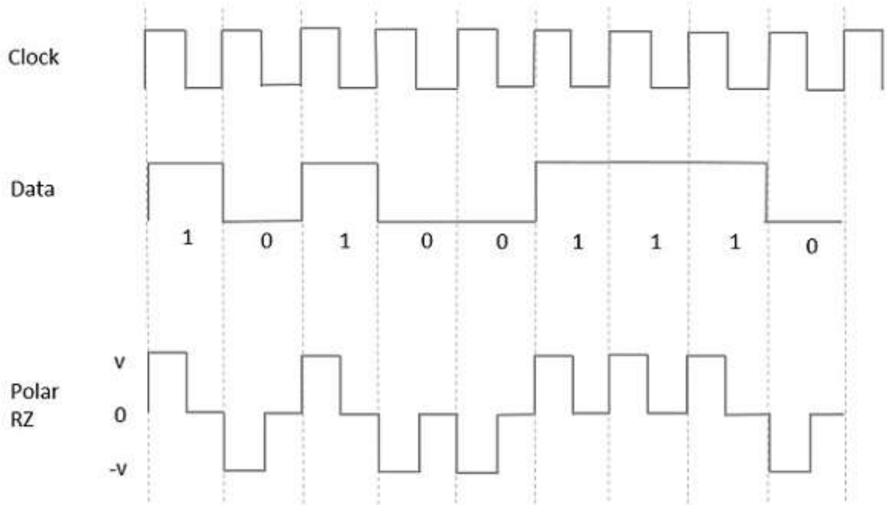
The disadvantages of Polar NRZ are −

- No error correction.
- No clock is present.
- The signal droop is caused at the places where the signal is non-zero at **0 Hz**.

Polar RZ

In this type of Polar signaling, a High in data, though represented by a **Mark pulse**, its duration T_0 is less than the symbol bit duration. Half of the bit duration remains high but it immediately returns to zero and shows the absence of pulse during the remaining half of the bit duration.

However, for a Low input, a negative pulse represents the data, and the zero level remains same for the other half of the bit duration. The following figure depicts this clearly.



Advantages

The advantages of Polar RZ are −

- It is simple.
- No low-frequency components are present.

Disadvantages

The disadvantages of Polar RZ are −

- No error correction.
- No clock is present.
- Occupies twice the bandwidth of Polar NRZ.
- The signal droop is caused at places where the signal is non-zero at **0 Hz**.

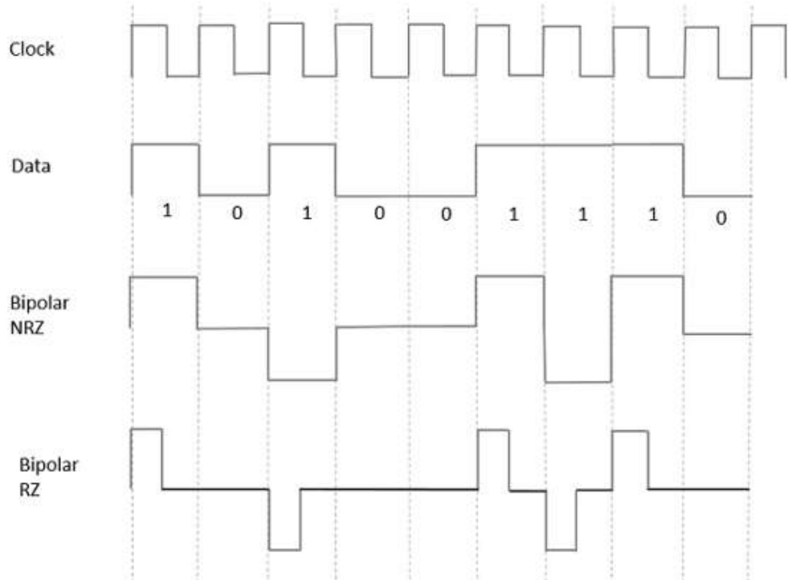
Bipolar Signaling

This is an encoding technique which has three voltage levels namely **+, -** and **0**. Such a signal is called as **duo-binary signal**. An example of this type is **Alternate Mark Inversion (AMI)**. For a **1**, the voltage level gets a transition from + to – or from – to +, having alternate **1s**to be of equal polarity. A **0** will have a zero voltage level.

Even in this method, we have two types.

- Bipolar NRZ
- Bipolar RZ

From the models so far discussed, we have learnt the difference between NRZ and RZ. It just goes in the same way here too. The following figure clearly depicts this.



The above figure has both the Bipolar NRZ and RZ waveforms. The pulse duration and symbol bit duration are equal in NRZ type, while the pulse duration is half of the symbol bit duration in RZ type.

Advantages

Following are the advantages −

- It is simple.
- No low-frequency components are present.
- Occupies low bandwidth than unipolar and polar NRZ schemes.
- This technique is suitable for transmission over AC coupled lines, as signal drooping doesn’t occur here.
- A single error detection capability is present in this.

Disadvantages

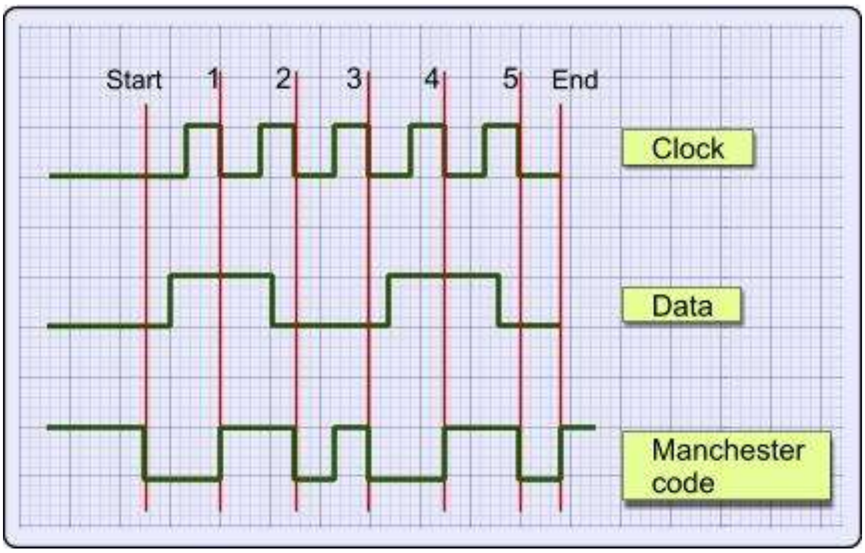
Following are the disadvantages −

- No clock is present.
- Long strings of data causes loss of synchronization.

Manchester encoding

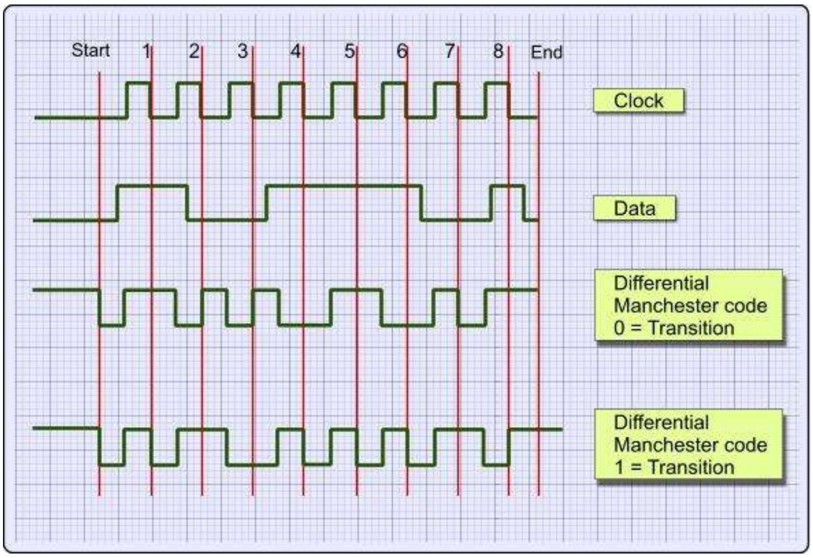
In data transmission, Manchester encoding is a form of digital encoding in which data bits are represented by transitions from one logical state to the other. This is different from the more common method of encoding, in which a bit is represented by either a high state such as +5 volts or a low state such as 0 volts.

When the Manchester code is used, the length of each data bit is set by default. This makes the signal self-clocking. The state of a bit is determined according to the direction of the transition. In some systems, the transition from low to high represents logic 1, and the transition from high to low



represents logic 0. In other systems, the transition from low to high represents logic 0, and the transition from high to low represents logic 1.

The chief advantage of Manchester encoding is the fact that the signal synchronizes itself. This minimizes the error rate and optimizes reliability. The main disadvantage is the fact that a Manchester-encoded signal requires that more bits be transmitted than those in the original signal.

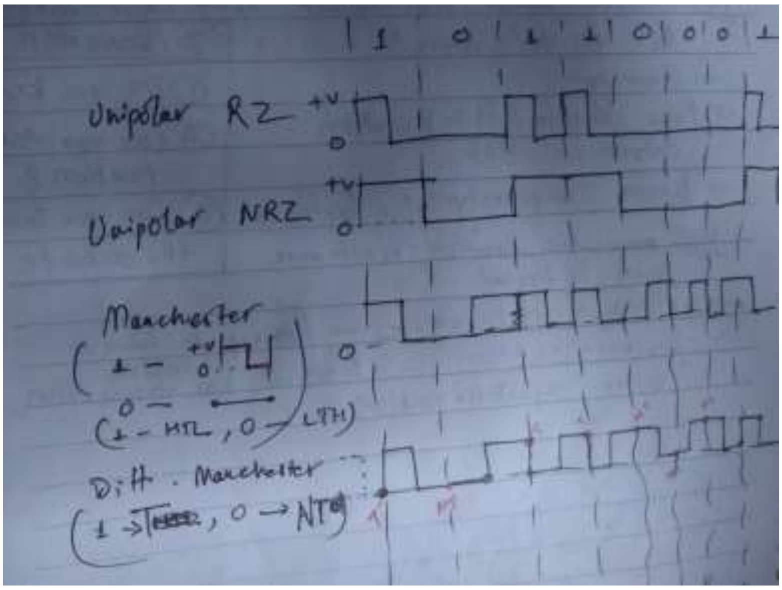


The Differential Manchester Coding

The Differential Manchester Code is a variation of the Manchester code. After searching around the internet for a while to find more info about this method, i sadly discovered that although there are many sites explaining the Differential Manchester Code, most of them do not make it very clear. So i had to seek in my books for more info. I then discovered that it is not that hard to understand the differential coding algorithm after all.

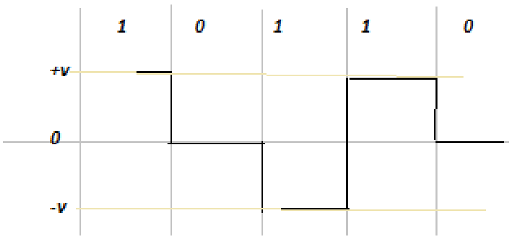
The major difference between the Differential Manchester Code, is that the receiver does not need to know the polarity of the signal. The polarity can be figured from the line transitions. In the above example, and NOT from the code line polarity. This is something that makes this method kinda difficult to understand. **It does not matter whether a logical 1 or 0 is received, but only whether the polarity is the same or is different from the previous value.**

9. **For a given bit sequence 10110001, draw the waveforms for the following line codes. NRZ, RZ, Differential Manchester, Manchester, AMI.**



10. **Draw the waveform of a line code which satisfies the following requirements**
    (a) **Has a built in error detection capability**
    (b) **Requires lesser bandwidth**
    (c) **Has zero average DC value.**

The line code that satisfies the requirements is AMI (Alternate Mark Inversion) and waveform is

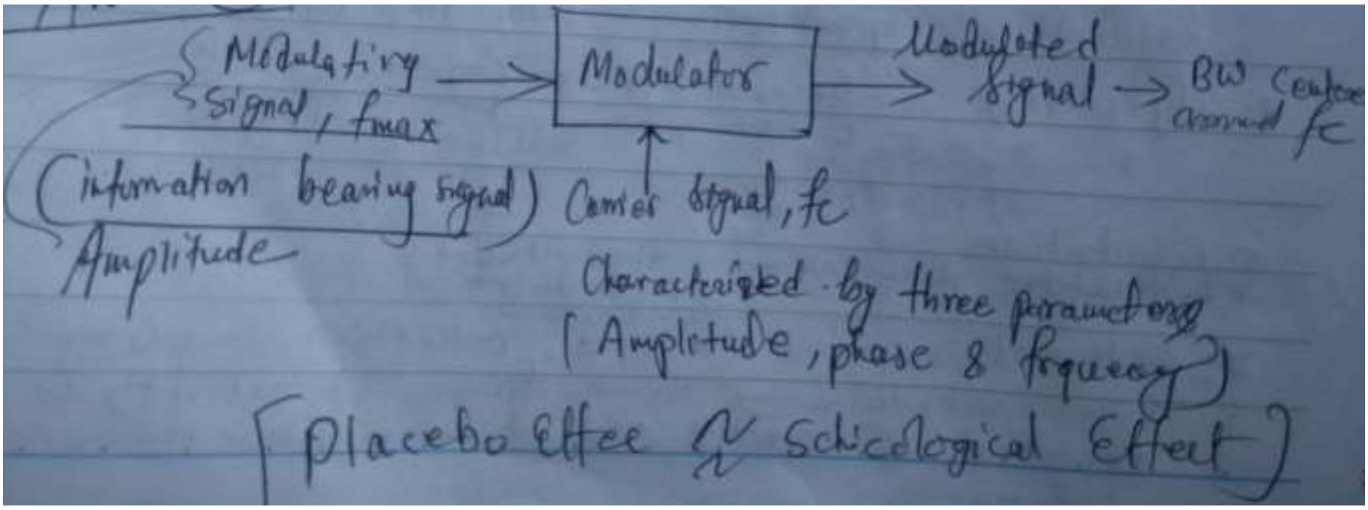


bit sequence are => 10110

Modulation

11. What do you mean by modulation in electronic communication system?

Modulation is the process of varying one or more properties of a periodic waveform, called the carrier signal, with a modulating signal that typically contains information to be transmitted.



12. Why modulation required, Importance of modulation, Characteristics of modulation.

- Reduction in the height of antenna
- Avoids mixing of signals
- Increases the range of communication
- Multiplexing is possible
- Improves quality of reception

1. Reduction in the height of antenna

For the transmission of radio signals, the antenna height must be multiple of λ/4 ,where λ is the wavelength .

$\lambda = c/f$

where c : is the velocity of light

f: is the frequency of the signal to be transmitted

The minimum antenna height required to transmit a baseband signal of f = 10 kHz is calculated as follows :

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^3} = 7500\ meters\ i.e.\ 7.5\ km$$

The antenna of this height is practically impossible to install .

Now, let us consider a modulated signal at f = 1 MHz . The minimum antenna height is given by,

$$Minimum\ antenna\ height = \frac{\lambda}{4} = \frac{c}{4f} = \frac{3 \times 10^8}{4 \times 10 \times 10^6} = 75\ meters$$

This antenna can be easily installed practically . Thus, modulation reduces the height of the antenna .

2. Avoids mixing of signals

If the baseband sound signals are transmitted without using the modulation by more than one transmitter, then all the signals will be in the same frequency range i.e. 0 to 20 kHz . Therefore, all the signals get mixed together and a receiver can not separate them from each other .

Hence, if each baseband sound signal is used to modulate a different carrier then they will occupy different slots in the frequency domain (different channels). Thus, modulation avoids mixing of signals .

3. Increase the Range of Communication

The frequency of baseband signal is low, and the low frequency signals can not travel long distance when they are transmitted . They get heavily attenuated .

The attenuation reduces with increase in frequency of the transmitted signal, and they travel longer distance .

The modulation process increases the frequency of the signal to be transmitted . Therefore, it increases the range of communication.

4. Multiplexing is possible

Multiplexing is a process in which two or more signals can be transmitted over the same communication channel simultaneously .

This is possible only with modulation.

The multiplexing allows the same channel to be used by many signals . Hence, many TV channels can use the same frequency range, without getting mixed with each other or different frequency signals can be transmitted at the same time .

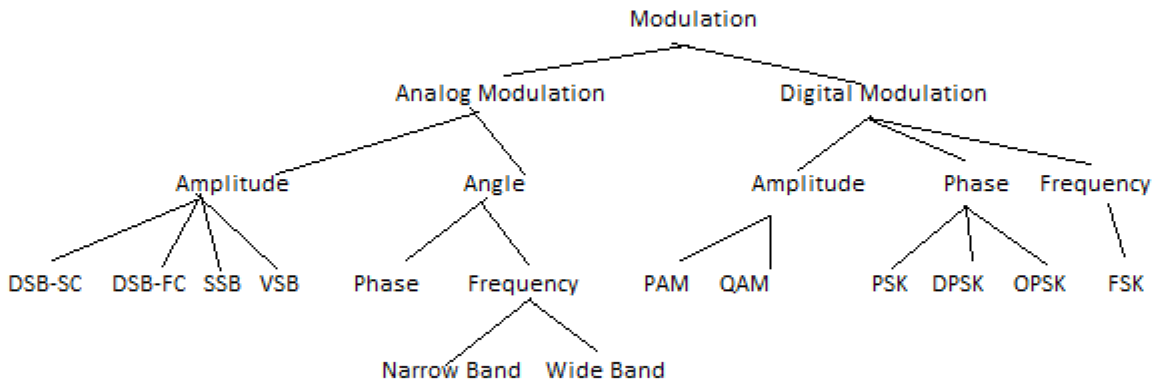
5. Improves Quality of Reception

With frequency modulation (FM) and the digital communication techniques such as PCM, the effect of noise is reduced to a great extent . This improves quality of reception

13. Why modulation is used in communication?

- In modulation technique, the message signal frequency is raised to a range so that it is more useful for transmission. The following points describe modulation's importance in communication system.
- In signal transmission, the signals from various sources are transmitted through a common channel simultaneously by using multiplexers. If these signals are transmitted simultaneously with certain bandwidth, they cause interference. To overcome this, speech signals are modulated to various carrier frequencies in order for the receiver to tune them to desired bandwidth of his own choice within the range of transmission.
- Another technical reason is antenna size; the antenna size is inversely proportional to the frequency of the radiated signal. The order of the antenna aperture size is at least one by tenth of the wavelength of the signal. Its size is not practicable if the signal is 5 KHz; therefore, raising frequency by modulating process will certainly reduce the height of the antenna.
- Modulation is important to transfer the signals over large distances, since it is not possible to send low-frequency signals for longer distances.
- Similarly, modulation is also important to allocate more channels for users and to increase noise immunity.

14. What are the different type of modulation techniques? Give a summary of Characteristics of modulation techniques.

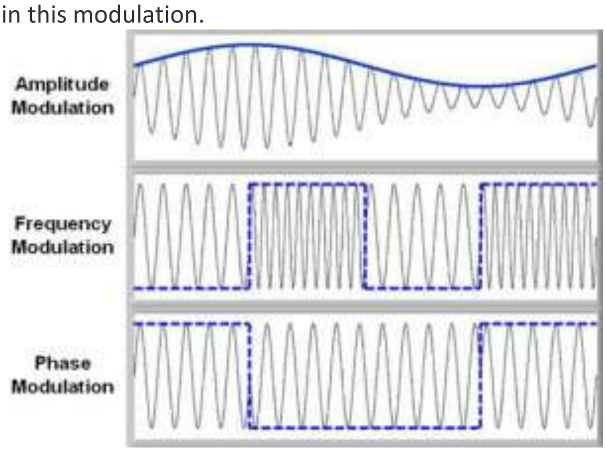


1) Analog Modulation

i. Amplitude modulation (AM) – DSB-SC, DSB-FC, SSB, VSB
ii. Frequency modulation (FM)
iii. Phase modulation (PM)

In **amplitude modulation**, the amplitude of the carrier wave is varied in proportion to the message signal, and the other factors like frequency and phase remain constant. The modulated signal is shown in the below figure, and its spectrum consists of lower frequency band, upper frequency band and carrier frequency components. This type of modulation requires greater band width, more power. Filtering is very difficult in this modulation.

Frequency modulation (FM) varies the frequency of the carrier in proportion to the message or data signal while maintaining other parameters constant. The advantage of FM over AM is the greater suppression of noise at the expense of bandwidth in FM. It is used in applications like radio, radar, telemetry seismic prospecting, and so on. The efficiency and bandwidths depend on modulation index and maximum modulating frequency.



In **phase modulation**, the carrier phase is varied in accordance with the data signal. In this type of modulation, when the phase is changed it also affects the frequency, so this modulation also comes under frequency modulation.

Analog modulation (AM, FM and PM) is more sensitive to noise. If noise enters into a system, it persists and gets carried till the end receiver. Therefore, this drawback can be overcome by the digital modulation technique.

i. DSB-SC – VHF, Air traffic control radios, Stereo transmission, Analog TV
ii. DSB-FC – Radio Transmission
iii. SSB – Satellite - BW is sensitive
iv. VSB – Video transmission e.g. Television – simpler than SSB, BW is more
v. FM – Audio transmission

2) Digital Modulation

For a better quality and efficient communication, digital modulation technique is employed. The main advantages of the digital modulation over analog modulation include permissible power, available bandwidth and high noise immunity. In digital modulation, a message signal is converted from analog to digital message, and then modulated by using a carrier wave.

The carrier wave is keyed or switched on and off to create pulses such that the signal is modulated. Similar to the analog, here the parameters like amplitude, frequency and phase variation of the carrier wave decides the type of digital modulation.

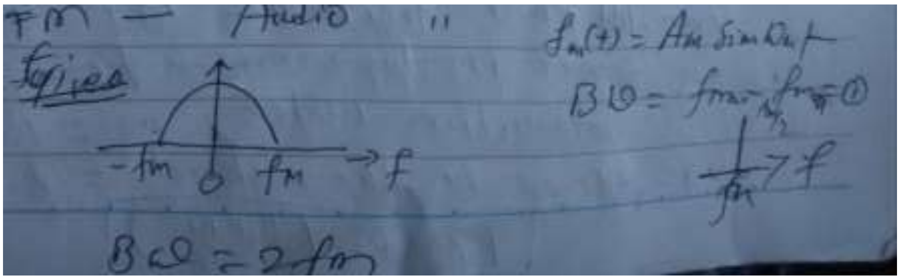
Digital modulation is of several types depending on the type of signal and application used such as Amplitude Shift Keying, Frequency Shift Keying, Phase Shift Keying, Differential Phase Shift Keying, Quadrature Phase Shift Keying, Minimum Shift Keying, Gaussian Minimum Shift Keying, Orthogonal Frequency Division Multiplexing, etc., as shown in the figure.

- ✓ **Amplitude shift keying (ASK)** changes the amplitude of the carrier wave based on the base band signal or message signal, which is in digital format. It is used for low-band requirements and is sensitive to noise.
- ✓ In **frequency shift keying (FSK),** the frequency of the carrier wave is varied for each symbol in the digital data. It needs larger bandwidths as shown in the figure.
- ✓ Similarly, the **phase shift keying (PSK)** changes the phase of the carrier for each symbol and it is less sensitive to noise.

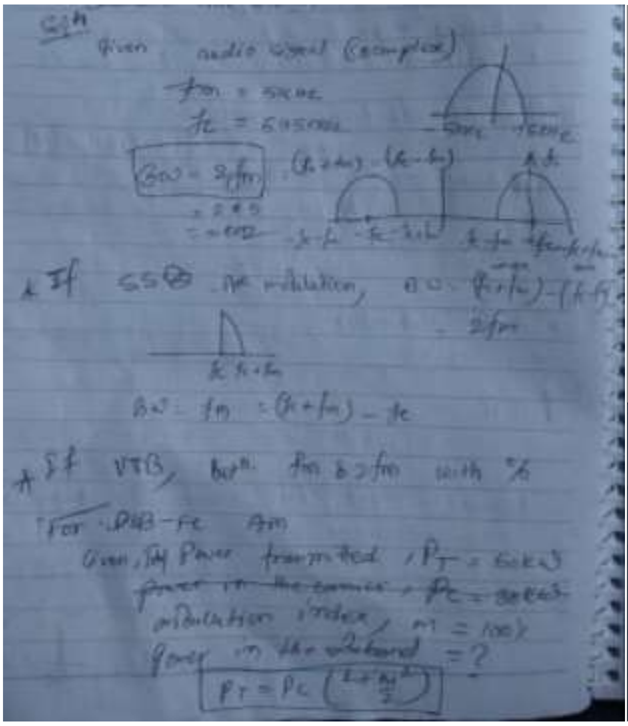
15. Given AM system

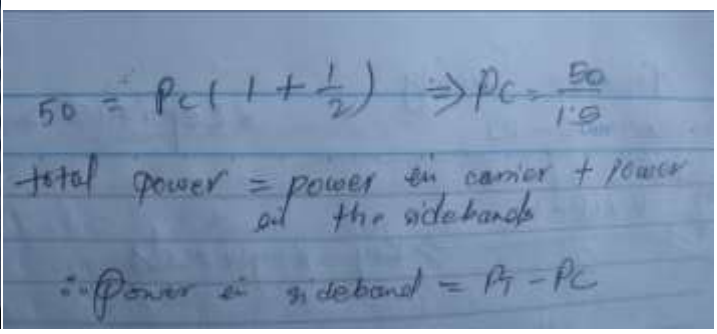
(a) Calculate the BW

(b) Calculate the power (carrier, sideband)



Q. You are transmitting audio signal of maximum frequency, fm=5KHz using a carrier frequency 535 KHz and using DSB-FC AM modulation- calculate the bandwidth of the modulated signal





16. For a given FM system, calculate the BW.

17. Compare AM and FM systems.

Attributes	AM	FM
Transmission	In AM modulation, amplitude of the signal is varied, and frequency and phase are kept constant.	In FM modulation, frequency of the signal is varied, and amplitude and phase are kept constant.
No of sidebands	AM has two sidebands	FM has infinite number of sidebands
Power uses	The carrier of AM comprises of most of the transmitted power, which contains no information.	All transmitted power in FM is useful, and there is no wastage of power unlike AM.
Modulation index range	Modulation index in AM varies from 0 to 1.	Modulation index in FM is always greater than one.
Noisy	AM is more noisy since the AM receivers do not have amplitude limiters.	Noise in FM can be reduced by employing amplitude limiters to remove the amplitude variations caused by noise.
Bandwidth	AM has narrow channel bandwidth which is 2fmfm.	The bandwidth in FM is much higher, up to 10 times as that of AM.
No of signals	In AM if two or more signals received at same frequency, then both will be demodulated, this can lead to interference.	In FM if two or more signal received at same frequency, the receiver will capture the stronger signal and eliminate the weaker signal.
Operations	AM broadcast operates in the medium frequency (MF) and high frequency (HF).	FM broadcast operates in the upper VHF and UHF range, where noise effects are minimal.
Complexity	The design of AM transmitter and receiver is not complex for the modulation and demodulation purpose.	The design of FM transmitter and receiver is relatively complex for the modulation and demodulation purpose.
Cost	AM transmission and reception equipments are not that expensive since the circuitry is relatively simple.	FM transmission and reception equipment is expensive as the circuitry is complex.
Uses	Mainly talk radio and news programming	Music radio and public radio
Frequency range	AM radio ranges from 535 to 1705 KHz (OR) Up to 1200 bits per second.	FM radio ranges in a higher spectrum from 88 to 108 MHz. (OR) 1200 to 2400 bits per second.

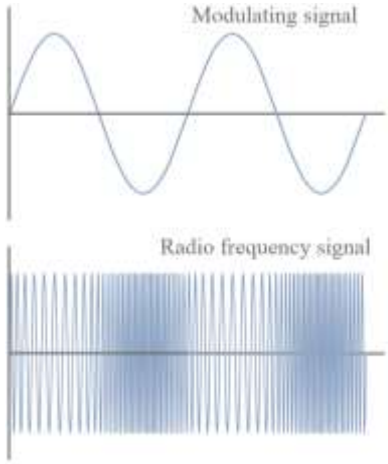
18. Compare different types of Digital Modulation techniques.

Compare ASK , PSK & FSK systems

Parameters	ASK	FSK	PSK
Variable characteristics	Amplitude	Frequency	Phase
Bandwidth	Is proportional to signal rate (B =(1+d)S),d is due to modulation & filtering ,lies between 0 & 1.	B=(1+d)×S+2Δf	B=(1+d)×S
Noise immunity	low	High	High
Complexity	Simple	Moderately complex	Very complex
Error probability	High	Low	Low
Performance in presence of noise	Poor	Better than ASK	Better than FSK
Bit rate	Suitable upto 100 bits/sec	Suitable upto about 1200 bits/sec	Suitable for high bit rates

19. What is FM? Illustrate with waveform.

To generate a frequency modulated signal, the frequency of the radio carrier is changed in line with the amplitude of the incoming audio signal.



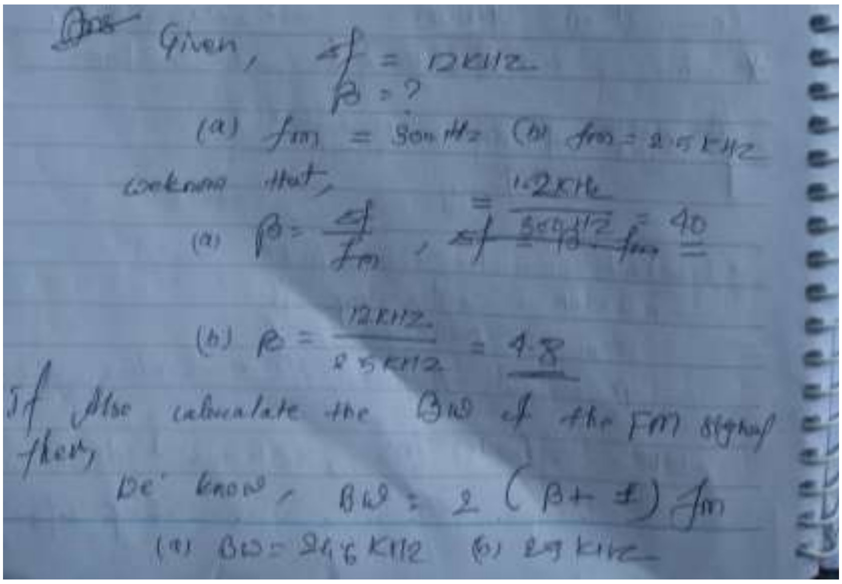
When the audio signal is modulated onto the radio frequency carrier, the new radio frequency signal moves up and down in frequency. The amount by which the signal moves up and down is important. It is known as the deviation and is normally quoted as the number of kilohertz deviation. As an example the signal may have a deviation of plus and minus 3 kHz, i.e. ±3 kHz. In this case the carrier is made to move up and down by 3 kHz.

Broadcast stations in the VHF portion of the frequency spectrum between 88.5 and 108 MHz use large values of deviation, typically ±75 kHz. This is known as wide-band FM (WBFM). These signals are capable of supporting high quality transmissions, but occupy a large amount of bandwidth. Usually 200 kHz is allowed for each wide-band FM transmission. For communications purposes less bandwidth is used. Narrow band FM (NBFM) often uses deviation figures of around ±3 kHz.

It is narrow band FM that is typically used for two-way radio communication applications. Having a narrower band it is not able to provide the high quality of the wideband transmissions, but this is not needed for applications such as mobile radio communication.

20. A cell phone transmitter has a maximum frequency deviation of 12 kHz, calculate the modulation index. If it operates a maximum deviation with a voice frequency of

(a) 300 Hz (b) 2.5 KHz



Ans Given, $\Delta f = 12KHz$

$\beta = ?$

(a) $f_m = 300 Hz$ (b) $f_m = 2.5 KHz$

We know that,

(a) $\beta = \frac{\Delta f}{f_m}$, $= \frac{12KHz}{300Hz} = 40$

(b) $\beta = \frac{12KHz}{2.5KHz} = 4.8$

If Also calculate the BW of the FM signal then,

We know, $BW = 2(\beta + 1) f_m$

(a) BW = 24.6 KHz (b) 29 KHz

21. When do we go for M-aray modulation?

- The word binary represents two-bits. M simply represents a digit that corresponds to the number of conditions, levels, or combinations possible for a given number of binary variables.
- This is the type of digital modulation technique used for data transmission in which instead of one-bit, two or more bits are transmitted at a time. As a single signal is used for multiple bit transmission, the channel bandwidth is reduced.

M-ary Equation

If a digital signal is given under four conditions, such as voltage levels, frequencies, phases and amplitude, then M = 4.

The number of bits necessary to produce a given number of conditions is expressed mathematically as

$$N = \log_2 M$$

Where,

N is the number of bits necessary.

M is the number of conditions, levels, or combinations possible with N bits.

The above equation can be re-arranged as −

$$2^N = M$$

For example, with two bits, 2^2 = 4 conditions are possible.

22. Measure of system performance in AM and DM.

For Analog, System performance parameters – SNR

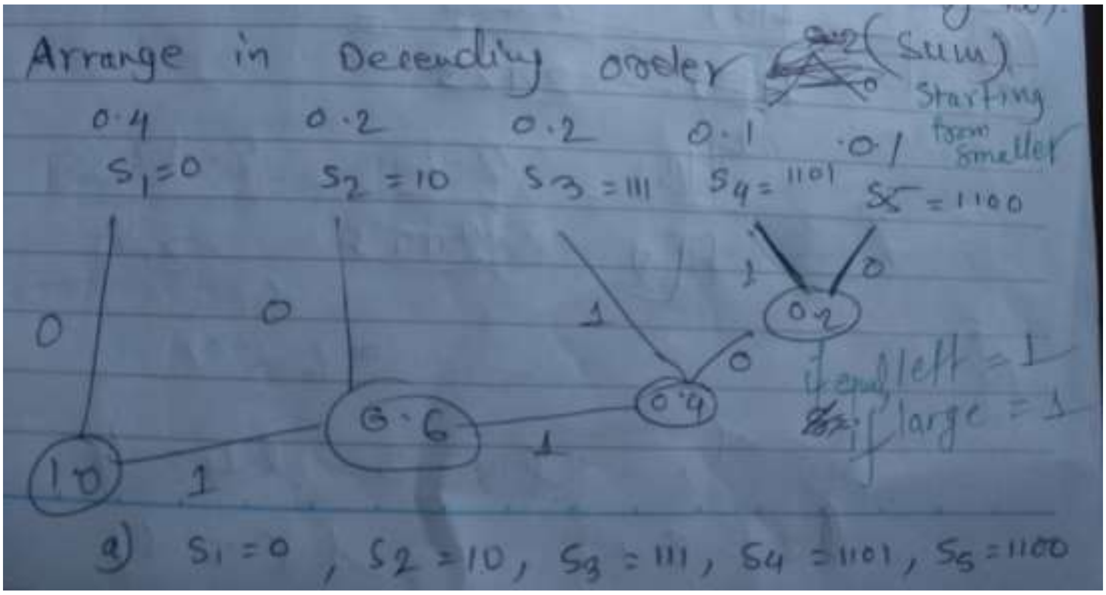
For Digital, System performance parameters – BER

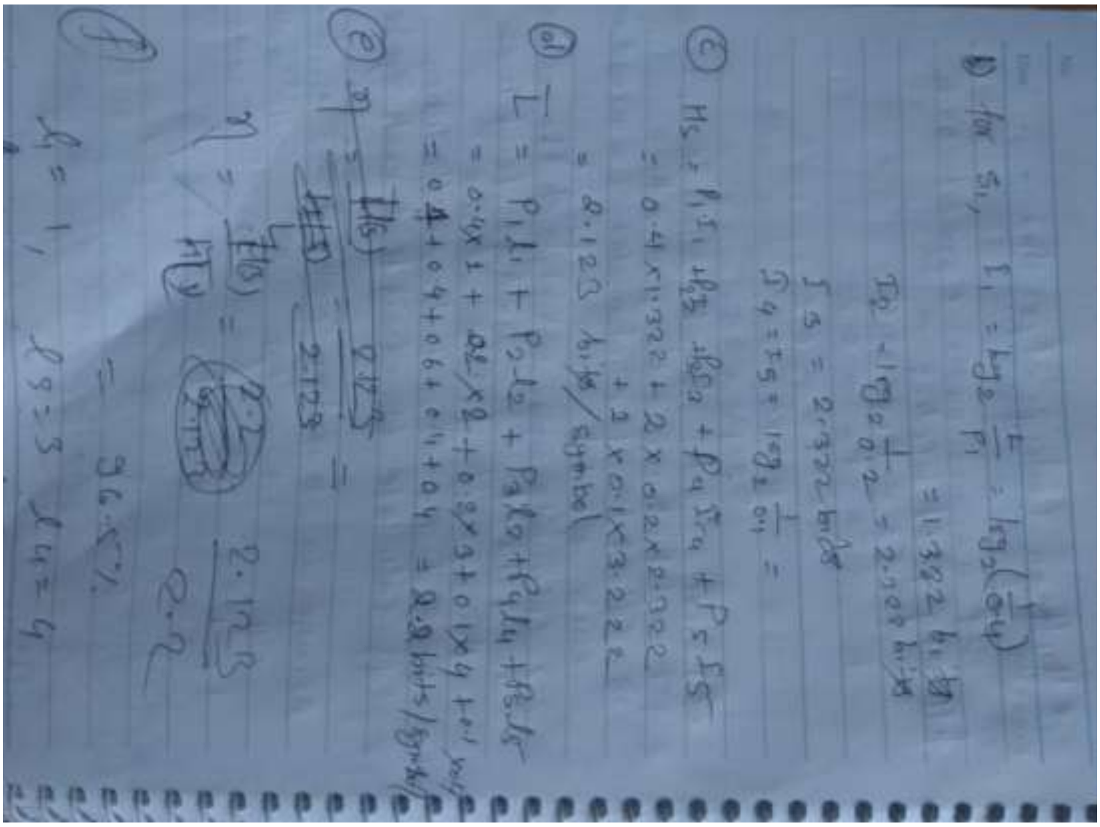
23. Huffman Coding Algorithm

If a source emits 5 symbols with following probabilities

S1-P1 = 0.4, S2-P2=0.2, S3-P3=0.2, S4-P4=0.1, S5-P5=0.1

a) Find the code for all symbols S1 to S5 using Huffman Coding algorithm
b) Calculate the information contained I each symbol
c) Calculate the Entrophy of the source.
d) Calculate the average length of the symbol for the source.
e) Calculate the coding efficiency
f) Calculate the length of each symbol





24. Category of channels

a) **Noiseless channel** – determine by Nyquist's theorem

$C = 2B \log_2 M$

Suppose the highest frequency component, in hertz, for a given analog signal is *f*max. According to the Nyquist Theorem, the sampling rate must be at least 2*f*max, or twice the highest analog frequency component. The sampling in an analog-to-digital converter is actuated by a pulse generator (clock). If the sampling rate is less than 2*f*max, some of the highest frequency components in the analog input signal will not be correctly represented in the digitized output. When such a digital signal is converted back to analog form by a digital-to-analog converter, false frequency components appear that were not in the original analog signal. This undesirable condition is a form of distortion called aliasing.

b) **Noisy channel** – determine by Shannon's Channel capacity theorem

Shannon's Theorem gives an upper bound to the capacity of a link, in bits per second (bps), as a function of the available bandwidth and the signal-to-noise ratio of the link.

The Theorem can be stated as:

$$C = B \log_2 (1 + S/N)$$

Where C is the achievable channel capacity, B is the bandwidth of the line, S is the average signal power and N is the average noise power.The signal-to-noise ratio (S/N) is usually expressed in decibels (dB)

Error Control Coding

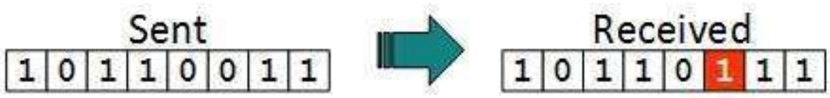
25. What is error?

Error is in digital communication system in which 1 is transmitted, 0 is received or 0 is transmitted, 1 is received.

Types of Errors

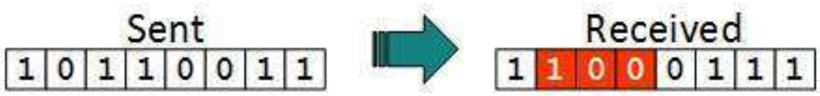
There may be three types of errors:

- 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 than1 consecutive bits corrupted.



26. What is error detection?

It allows a receiver to check whether received data has been corrupted during transmission. It can, for example, request a retransmission. Techniques are:- Parity check, CRC, Checksum

27. What is error correction?

This type of error control allows a receiver to reconstruct the original information when it has been corrupted during transmission. Techniques are:- Stop-and-wait ARQ, Sliding window- Go-Back-N, Selective Repeat

28. How can we detect error?

Parity check, CRC, Checksum

29. How can we correct error?

Stop-and-wait ARQ, Sliding window- Go-Back-N, Selective Repeat

- Automatic repeat request *(ARQ) or* **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** (FCC) - 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.

30. What are the advantages of error control channel coding?

- **Reduce the cost of communications systems**: Transmitter power is expensive, especially on satellite transponders. Coding can reduce the satellite's power needs because messages received at close to the thermal noise level can still be recovered correctly.
- **Eliminate interference**: As the electromagnetic spectrum becomes more crowded with man-made signals, error-control coding will mitigate the effects of unintentional interference.

31. What is a block code?

The main idea behind block codes is to provide the user or recipient of such codes inputs with the help of which the user can address any possible errors in the code without needing to contact the source of the code. In telecommunications, the principle is to encode a message in such a way so that the recipient of the message is able to correct a limited number of errors so as to have minimum acceptability of the message. This action prevents the possibility of retransmission of the message, which wastes time and resources.

Block coding helps in error detection and re-transmission of the signal. It is normally referred to as mB/nB coding as it replaces each m-bit data group with an n-bit data group (where n>m). Thus, its adds extra bits (redundancy bits) which helps in synchronization at receiver's and sender's end and also providing some kind of error detecting capability.

32. What is a convolution code?

A type of channel coding that adds patterns of redundancy to the data in order to improve the signal-to-noise ratio (SNR) for more accurate decoding at the receiving end.

- **One advantage of convolutional coding** is that it is not designed for a fixed block size, but can be adapted quickly to variable block lengths. However, it is not used for continuous streams, since the evaluation must reach the end of the block before making corrections. In practice, block sizes can vary from a few bits to many thousands of bits, each block length associated with a different processing delay.
- **Convolutional coding schemes need a matched pair of coders and decoders.** Although implementations differ slightly, the algorithms of the coder and decoder must be the same. Each must contain a specific knowledge of the most likely transitions that could occur. It is this underlying assumption that allows a best estimate as to the intended message. Their error correcting abilities differ, and some designs are known to be better than others. Coding schemes that use more complexity in forming their coded outputs tend to be better at correcting errors.

33. What is a repetition code?

- ✓ The **repetition code** is one of the most basic error-correcting codes. In order to transmit a message over a noisy channel that may corrupt the transmission in a few places, the idea of the repetition code is to just repeat the message several times. The hope is that the channel corrupts only a minority of these repetitions. This way the receiver will notice that a transmission error occurred since the received data stream is not the repetition of a single message, and moreover, the receiver can recover the original message by looking at the received message in the data stream that occurs most often.
- ✓ Because of the bad error correcting performance and the low ratio between information symbols and actually transmitted symbols, other error correction codes are preferred in most cases. The chief attraction of the repetition code is the ease of implementation.

In the case of a binary repetition code, there exist two code words - all ones and all zeros - which have a length of **n**. Therefore, the minimum Hamming distance of the code equals its length **n**. This gives the repetition code an error correcting capacity **of {(n-1)/2**} (i.e. it will correct up to **{(n-1)/2}** errors in any code word).

34. What is Hamming distance between two code word?

The Hamming distance between two code words is the number of bits in which two code words differ.

The Hamming distance between:

- "karolin" and "kathrin" is 3.
- "karolin" and "kerstin" is 3.
- **1011101** and **1001001** is 2.
- **2173896** and **2233796** is 3.

The minimum Hamming distance for a code is the smallest Hamming distance between all pairs of words in the code. E.g. in above example, the minimum hamming distance is 2 .

35. What is the maximum hamming distance of a code?

The maximum Hamming distance for a code is the maximum Hamming distance between all pairs of words in the code. E.g. in above example, the maximum hamming distance is 3.

36. What is the weight of a code word?

- In error-correcting coding, the minimum Hamming weight, commonly referred to as the **minimum weight** w_{min} of a code is the weight of the lowest-weight non-zero code word.
- The weight w of a code word is the number of 1s in the word.
  For example the word 11001010 has a weight of 4.
- In a linear block code the minimum weight is also the minimum Hamming distance (d_{min}) and defines the error correction capability of the code. If $w_{min} = n$, then $d_{min} = n$ and the code will correct up to $d_{min}/2$ errors

37. Which error your-control coding method (ARQ or FEC) would you choose in the following situations and why?

(a) For a highly reliable channel.

(b) For a long distance channel. Highly reliable and unreliable

(c) For a unreliable channel

38. Given

$$G = \begin{bmatrix} 1 & 1 & 0 & 1 & 0 & 0 & 0 \\ 0 & 1 & 1 & 0 & 1 & 0 & 0 \\ 1 & 1 & 1 & 0 & 0 & 1 & 0 \\ 1 & 0 & 1 & 0 & 0 & 0 & 1 \end{bmatrix}$$

(a) Find all the valid code words of the (7,4) block code.

(b) Calculate its error detection capacity.

(c) Calculate its error correction capacity.

Ans: we know, $C = mG$

$C_0 = m_0 G$

$C_1 = m_1 G$

$\vdots$

$C_{15} = m_{15} G$

Now,

$$C_0 = [0000] \begin{bmatrix} 1 & 1 & 0 & 1 & 0 & 0 & 0 \\ 0 & 1 & 1 & 0 & 1 & 0 & 0 \\ 1 & 1 & 1 & 0 & 0 & 1 & 0 \\ 1 & 0 & 1 & 0 & 0 & 0 & 1 \end{bmatrix} = [0000000]$$

$C_1 = [0001][\quad " \quad] = [1010001]$

$\vdots$

$C_{15} = [1111][\quad " \quad] = [1111111]$

$C_{10} = [1010][\quad " \quad] = [0011010]$

$C_7 = [0111][\quad " \quad] = [0010111]$

odd 1 ⇒ 1

even 1 ⇒ 0

$m_{13} = [1101][\quad " \quad] = [\cancel{0101001}]$
$= [0001101]$

w ⇒ non-zero weight

$C_0 = [0000000]$ — weight (W)

$C_1 = [$ — $w_0 =$

w_1

$\vdots$

$C_{15} = [\quad]$ — w_{15}

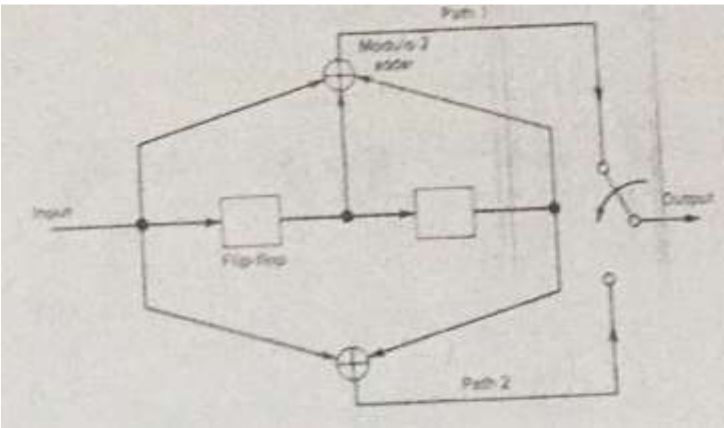
$d_{min} = 3$

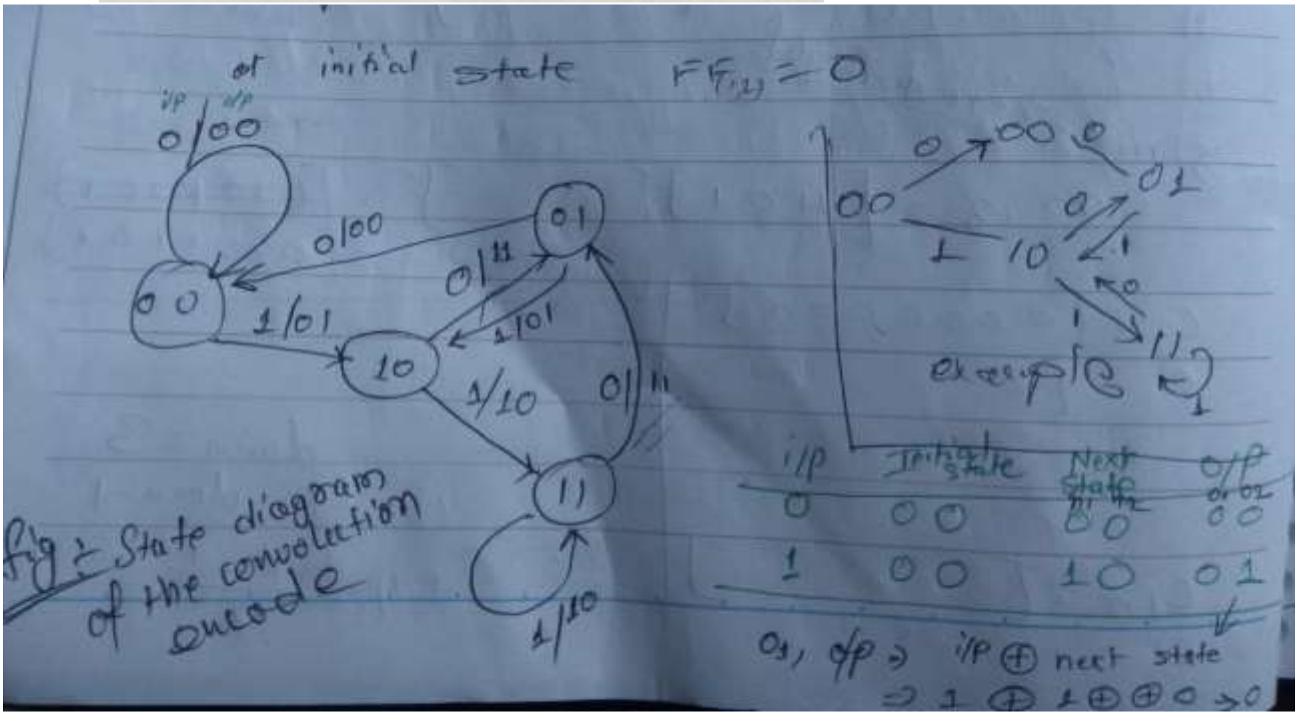
(b) $t_d = d_{min} - 1$

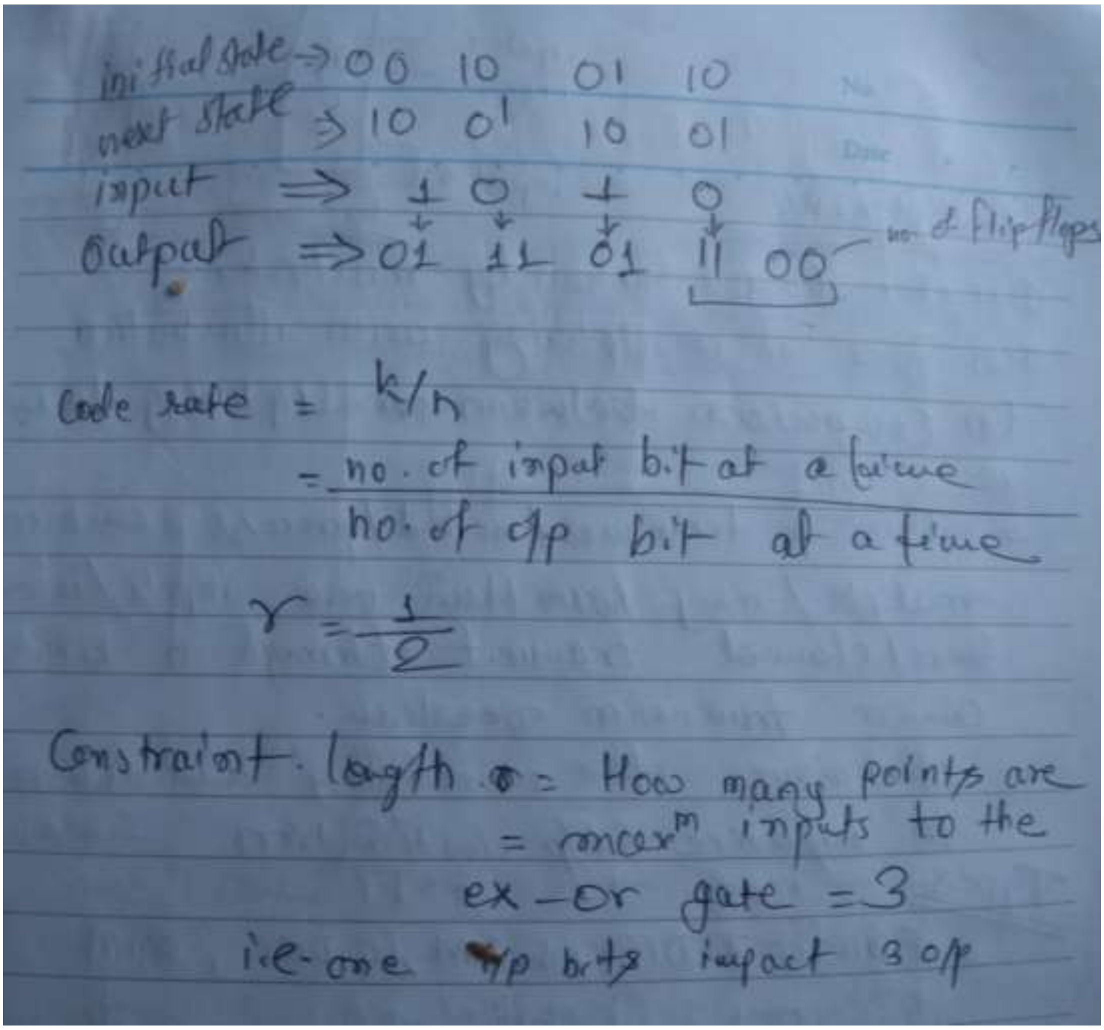
(c) $t_c = \frac{d_{min} - 1}{2}$

39. Convolution Code:

Q. for a given convolution encoder and the input message, find out the output and draw its state diagram. Also calculate it code rate and constraints length.






initial state → 00 10 01 10
next state → 10 01 10 01
input → 1 0 1 0
Output → 01 11 01 11 00 no. of flip flops

Code rate = k/n

$= \frac{\text{no. of input bit at a time}}{\text{no. of o/p bit at a time}}$

$r = \frac{1}{2}$

Constraint length = How many points are = concern inputs to the ex-or gate = 3
i.e. one i/p bits impact 3 o/p

40. Compare convolution code and block code.

- In block codes, information bits are followed by parity bits. In convolution codes, information bits are spread along the sequence.
- Block codes are memoryless whereas Convolution codes have memory.
- Convolution codes use small codewords in comparison to block codes, both achieving the same quality.
- Block code code information in blocks while convolutional codes convolve information bit sequence
- Convolutions codes encode much longer inputs at once, and hope to take advantage of this by spreading error correcting information over a long area. This makes it much easier to protect against e.g. burst errors.
- Unfortunately this also makes it really hard to study them and derive mathematical properties like possible with block codes.
- In practice, convolution codes often work better than block codes. In theory, it's really hard to show that this is the case.

Multiplexing

41. What do you mean by multiplexing?

Multiplexing (or MUX) is a way of sending multiple signals or streams of information over a communications link at the same time in the form of a single, complex signal; the receiver recovers the separate signals, a process called Demultiplexing (or DEMUX).

Networks use multiplexing for two reasons:

i. To make it possible for any network device to talk to any other network device without having to dedicate a connection for each pair. This requires shared media;
ii. To make a scarce or expensive resource stretch further -- e.g., to send many signals down each cable or fiber strand running between major metropolitan areas, or across one satellite uplink.

42. Types of multiplexing with illustrations.

There are 5 main types of multiplexing.

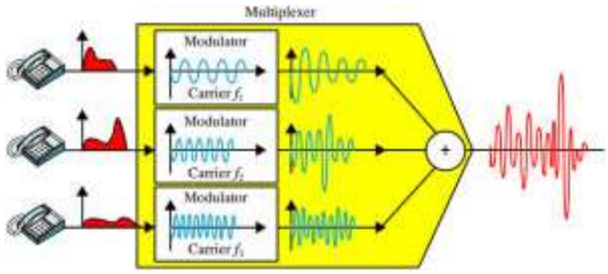
- Frequency division multiplexing (FDM)
- Time division multiplexing
- Wavelength division multiplexing

- Code Division Multiplexing
- Space Division Multiplexing

Frequency division multiplexing (FDM)

This is the oldest multiplexing technique and is used in many fields of communication, including broadcast television and radio, cable tv, and cell phones.

- It is also one of the simplest multiplexing techniques.



FDM is the assignment of non-overlapping frequency ranges to each "user" of a medium. A user might be a television station that transmits its television channel through the airwaves (the medium) into homes and businesses. A user might also be the cell phone transmitting signals over the medium on which you are talking.

To allow multiple users to share a single medium, FDM assigns each user a separate channel.

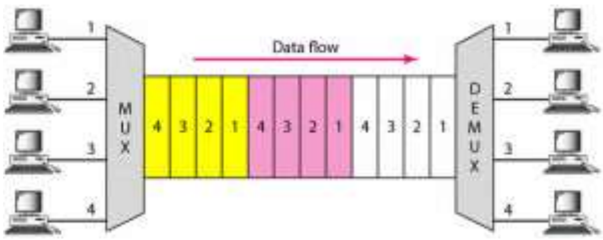
- *Channel*: an assigned set of frequencies that is used to transmit the user's signal.

The way cellphone towers perform frequency division multiplexing, is by dividing the bandwidth that is available to them into multiple channels. Therefore, the telephone connection of one user is assigned one set of frequencies.

The device that accepts input from one or more users is called the *multiplexor*. The device attached to the receiving end of the medium that splits off each signal to deliver it to the appropriate receiver is called the second multiplexor, or *demultiplexor*.

To keep one signal from interfering with another signal, a set of unused frequencies called a *guard band* is usually inserted between the two signal to provide a form of insulation.

Time division multiplexing (TDM)

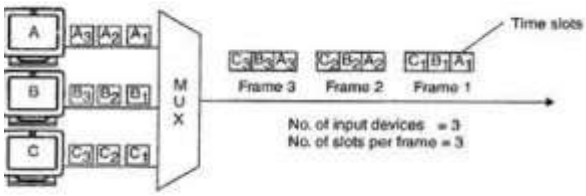


Time division multiplexing only allows one user at a time to transmit, and the sharing og the medium is accomplished by dividing available transmission time among users. Here, a user uses the entire bandwidth of channel, but only for a brief moment.

The way time division multiplexing works is, if there are 50 cellphones. The cell tower will ask all 50 cellphones one by one if they have something to transmit. If they don't a reply of 'No' is forwarded. If they do a 'message' is forwarded. This gives each device a turn at transmitting its data over a high-speed line. Suppose two users, A and B, wish to transmit data over a shared medium to a distant computer. We can create a rather simple time division multiplexing scheme by allowing user A to transmit during the first second, then user B during the following second, followed again by user A during the third second, and so on.

Time division multiplexing has split into two roughly parallel but separate technologies:

- Synchronous time division multiplexing
- Statistical time division multiplexing

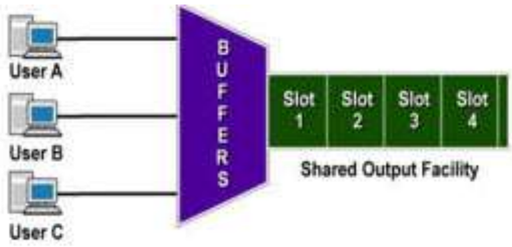


Synchronous time division multiplexing (Sync TDM)

This gives each incoming source signal a turn to be transmitted, proceeding through the sources in round-robin fashion. Given n inputs, a synchronous time division multiplexor accepts one piece of data, such as a byte, from the first device, transmits it over a high-speed link, accepts one byte from the second device, transmits it over the high-speed link, and continues this process until a byte is accepted from the nth device. After the nth device's first byte is transmitted, the multiplexor returns to the first device and continues in round-robin fashion. Alternately, rather than accepting a byte at a time from each source, the multiplexor may accept single bits as the unit input from each device.

If a device has nothing to transmit à the multiplexor must still allocate a slot for that device in the high-speed output stream; but that time slot will, in essence, be empty. Because each time slot is statically fixed in synchronous time division multiplexing, the multiplexor cannot take advantage of the empty slot and reassign busy devices to it.

Statistical time division multiplexing (Stat TDM)



Stat TDM transmits data only from active users and does not transmit empty time slots. To transmit data only from active users, the multiplexor creates a more complex frame that contains data only from those input sources that have something to send.
If only two of the four stations are transmitting, how does the demultiplexor on the receiving end recognize the correct recipients of the data? Some type of address must be included with each byte of data to identify who sent the data and for whom it is intended. The address can be as simple as a binary number that uniquely identifies the station that is transmitting.

Wavelength division multiplexing (WDM)

Wavelength division was the result of a single fiber-optic line transmitting billions of bits per second no longer being sufficient. This inability of a single fiber-optic line to meet users' needs is called *fiber exhaust*.
WDM multiplexes multiple data streams onto a single fiber-optic line. It is, in essence, a frequency division multiplexing technique that assigns input sources to separate sets of frequencies. Wave division multiplexing uses different wavelength (frequency) lasers to transmit multiple signals at the same time over a single medium. The wavelength of each differently colored laser is called the *lambda*. Thus, WDM supports multiple lambdas.
The technique assigns a uniquely colored laser to each input source and combines the multiple optical signals of the input sources so that they can be amplified as a group and transported over a single fiber. It is interesting to note that, because of the properties of the signals and glass fiber, plus the nature of light itself, each signal carried on the fiber can be transmitted at a different rate from the other signals. This means that a single fiber-optic line can support simultaneous transmission speeds such as 51.84 Mbps, 155.52 Mbps, 622.08 Mbps, and 2.488 Gbps.

- Wavelength division multiplexing is also scalable. As the demands grow, it is possible to add additional wavelengths, or lambdas, onto the fiber, thus further multiplying the capacity.

Code division multiplexing (CDM)

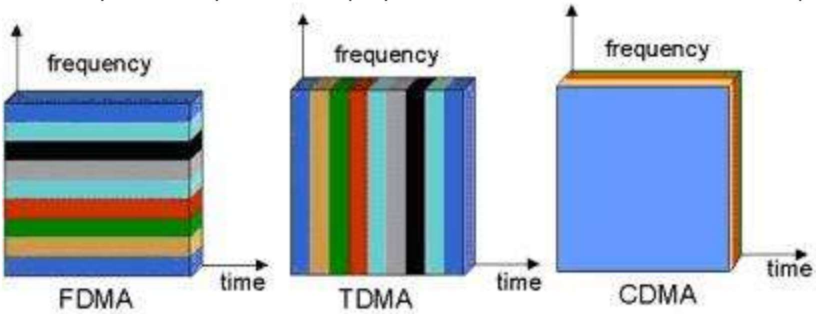
A relatively new technology that has been used extensively by both the military and cellular telephone companies. Whereas other multiplexing techniques differentiate one user from another by either assigning frequency ranges or interleaving bit sequences in time, code division multiplexing allows multiple users to share a common set of frequencies by assigning a unique digital code to each user.
Example of cellphone companies with CDMA phones: Sprint and Verizon
Discrete Multitone (DMT) is a multiplexing technique commonly found in DSL systems. DSL, as we have already seen, is a technology that allows a high-speed data signal to traverse a standard copper-based telephone line. We have also seen that the highest transmission speed we can achieve with a standard dial-up telephone line is 56 kbps.

Space Division Multiplexing (SDM)

Potential applications are in area where you require to transmit a huge quantity of data on short to mid ranges. For instance, in datacenters, or to connect different sections of a distribution network inside a city.

As SDM usually requires special fibers, it is not well suited for long-range applications, at least for short term, as it would require to replace a lot of infrastructures.

The potential advantage of SDM over simply using a bunch of fibers is it could potentially reduce costs and needed energy, if all parts of the network (amplifiers, routers, multiplexers, etc.) also support SDM. However, this cost reduction only apply in the case you already need to deploy a new fiber; otherwise the cost of deploying the new fiber would not justify it.



43. Comparison between multiplexing techniques.

Attributes	FDM	TDM	WDM
Definition	FDM is a transmission technique in which multiple data signals are combined for simultaneous transmission via a shared communication medium.	TDM is a transmission technique that allows multiple users to send signals over a common channel by allocating fixed time slot for each user.	WDM is a transmission technique that modulates numerous data streams, optical carrier signals of varying wavelengths into a single light beams through a single optical fiber.
Functionality	FDM divides the bandwidth into smaller frequency ranges an transmitter transmit data simultaneously through a common channel within their frequency range.	TDM allocates a fixed time slot for each user to send signals through a common channel. User gets the entire bandwidth within that time slot.	WDM combines multiple light beams from several channels and combine them to a single light beam and sends through a fiber optic strand similar to FDM.
Type of Signals	FDM uses analog signals.	TDM uses digital and analog signals.	WDM uses optical signals.

44. With the appropriate block diagram of a GSM system, list its important features.

1. Mobile Station (MS):

A mobile station communicates across the air interface with a base station transceiver in the same cell in which the mobile subscriber unit is located. The MS communicates the information with the user and modifies it to the transmission protocols if the air-interface to communicate with the BSS. The user's voice information is interfaced with the MS through a microphone and speaker for the speech, keypad, and display for short messaging, and the cable connection for other data terminals. The MS has two elements. The Mobile Equipment (ME) refers to the physical device, which comprises of transceiver, digital signal processors, and the antenna. The second element of the MS is the GSM is the Subscriber Identity Module (SIM). The SIM card is unique to the GSM system. It has a memory of 32 KB.

2. Base Station Subsystem (BSS):

A base station subsystem consists of a base station controller and one or more base transceiver station. Each Base Transceiver Station defines a single cell. A cell can have a radius of between 100m to 35km, depending on the environment. A Base Station Controller may be connected with a BTS. It may control multiple BTS units and hence multiple cells. There are two main architectural elements in the BSS – the Base Transceiver Subsystem (BTS) and the Base Station Controller (BSC). The interface that connects a BTS to a BSC is called the A-bis interface. The interface between the BSC and the MSC is called the A interface, which is standardised within GSM.

3. Network and switching subsystem (NSS)

The NSS is responsible for the network operation. It provides the link between the cellular network and the Public switched telecommunicates Networks (PSTN or ISDN or Data Networks). The NSS controls handoffs between cells in different BSSs, authenticates user and validates their accounts, and includes functions for enabling worldwide roaming of mobile subscribers.

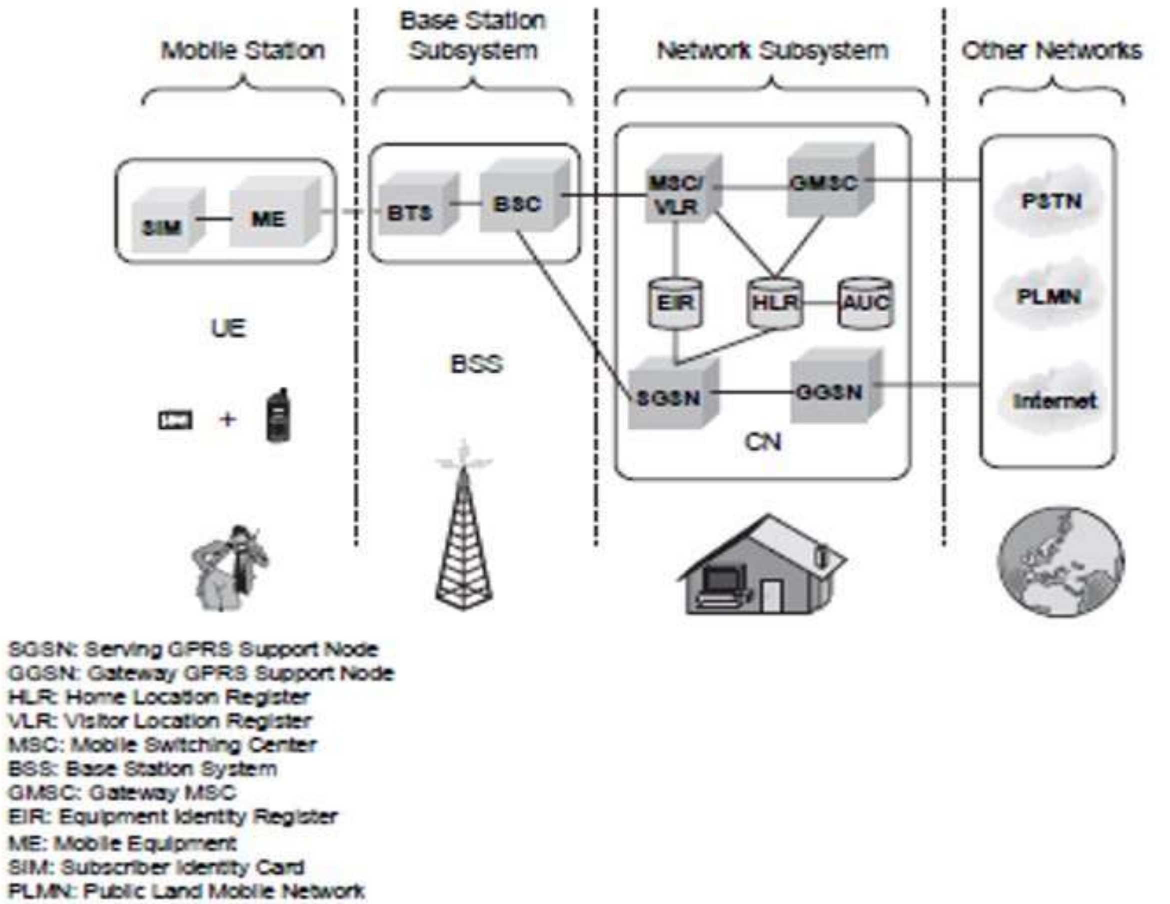


Figure **architecture in GSM.**

User Equipment (UE)—These are the users .Number of users are controlled by one BTS

- The mobile stations (MS) communicate with the base station subsystem over the radio the radio interface.
- The BSS called as radio the subsystem, provides and manages the radio transmission path between the mobile stations and the Mobile Switching Centre(MSC).It also manages radio interface between the mobile stations and other subsystems of GSM.
- Each BSS comprises many Base Station Controllers(BSC) that connect the mobile station to the network and switching subsystem (NSS) through the mobile switching center/
- The NSS controls the switching functions of the GSM system.It allows the mobile switching center to communicate with networks like PSTN, ISDN, CSPDN, PSPDN and other data networks.
- The operation support system (OSS) allows the operation and mantanance of the GSM system. It allows the system engineers to diagnose,troubleshoot and observe the parameters of the GSM systems.The OSS subsystem interacts with the other subsystems and is provided for the GSM operating company staff that provides service facilities for the network.

Base station(BSS)-- The following stations subsystem comprises of two parts:

i. Base Transceiver Station (BTS).
ii. Base Station Controller(BSC).

The BSS consists many BSC that connect to a single MSC. Each BSC controls upto several hundred BTS.

Base Transreciver Station(BTS)-BTS

It has radio transreciever that define a cell and are capable of handelling radio link protocols with MS.

Functions of BTS are

- Handelling radio link protocols
- Providing FD communication to MS.
- Interliving and de- interliving.

Base station controller(BSC) IT manages radio resources for one or more BTS.It controls several hundred BTS al are connected to single MSC.

Functions of BTS are

- To control BTS.

- Radio resource management
- Handoff management and control
- Radio channel setup and frequency hoping

Network subsystem(NSS)

1) 1.It handels the switching of GSM calls between external networks and indoor BSC
2) 2.It includes three different data bases for mobility management as
   (a) A .HLR (Home Location Register)
   (b) B .VLR (Visitor Location Register)
   (c) C. AUC (Authentication center)

Mobile switching center (MSC)—

- It connects fix networks like ISDN ,PSTN etc.
- Following are the functions of MSC
  - Call setup, supervision and relies
  - COLLECTION OF BILLING INFORMATION
  - Call handelling / routing
  - Management of signelling protocol
  - Record of VLR and HLR

HLR (Home Location Register) - Call roamiming and call routing capabilities of GSM are handeled.It stores all the adminstartive information of sub scriber registered in the networks.IT maintance unique international mobile subscriber identity.(IMSI).
VLR (Visitor Location Register) - It is a temporary data base.It stores the IMSC number and costomer information for each roaming customer visiting specific MSC.
Authentication center - It is protected database .It maintance authentication keys and algorithms.It contain s aregister called as Equipment Identity Register.
Operation subsystem(OSS) - IT manages all mobile equipment in the system 1)management for charging and billing procedure 2)To maintain all hardware and network operations
Interfaces used for GSM network : (ref fig 2)
1)UM Interface –Used to communicate between BTS with MS
2)Abis Interface— Used to communicate BSC TO BTS
3)A Interface-- Used to communicate BSC and MSC
4) Singling protocol (SS 7)- Used to communicate MSC with other network .

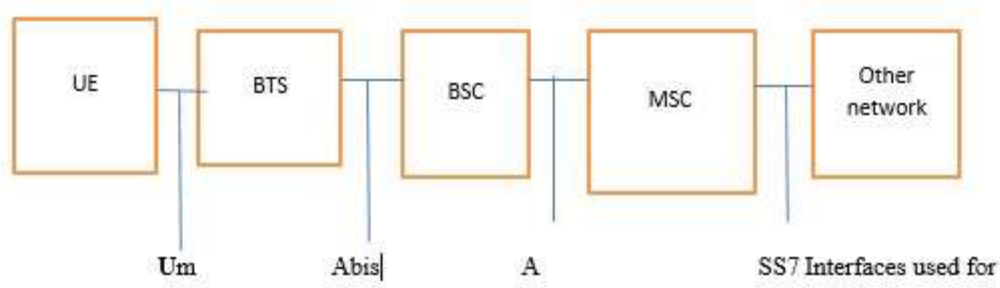


Fig 2 GSM network Interfaces

45. What are the advantages of a satellite communication system? Draw a simplified diagram of a satellite communication system and explain its operation.

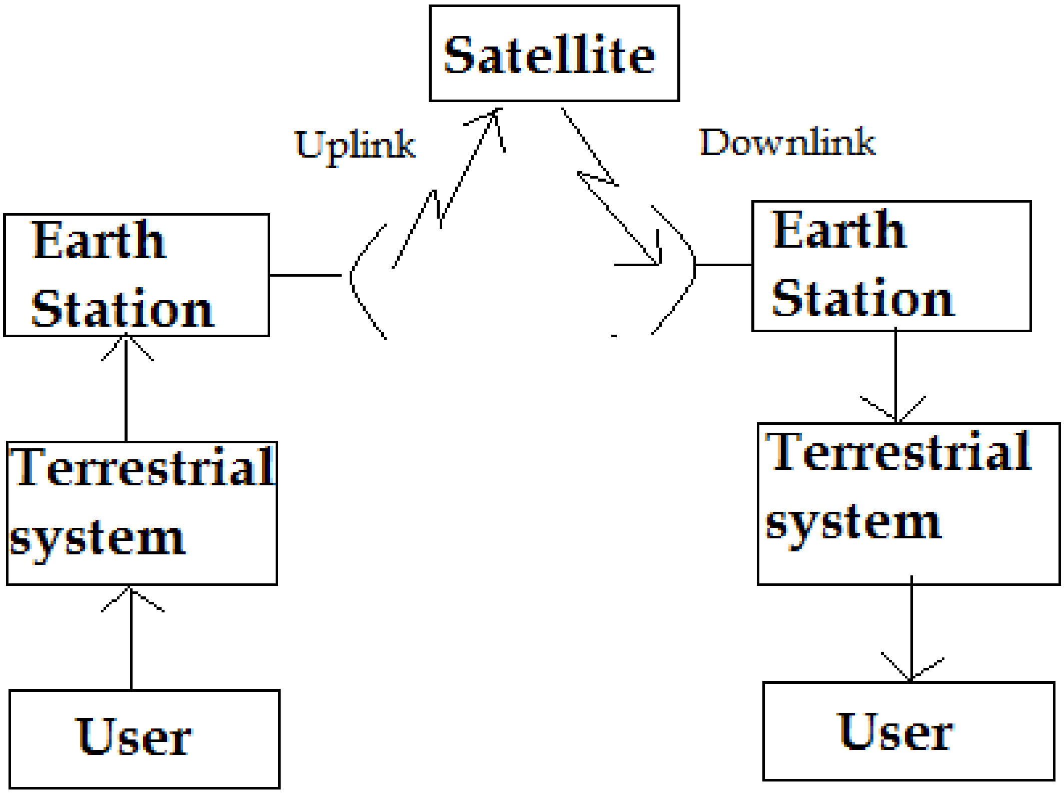
Advantages:

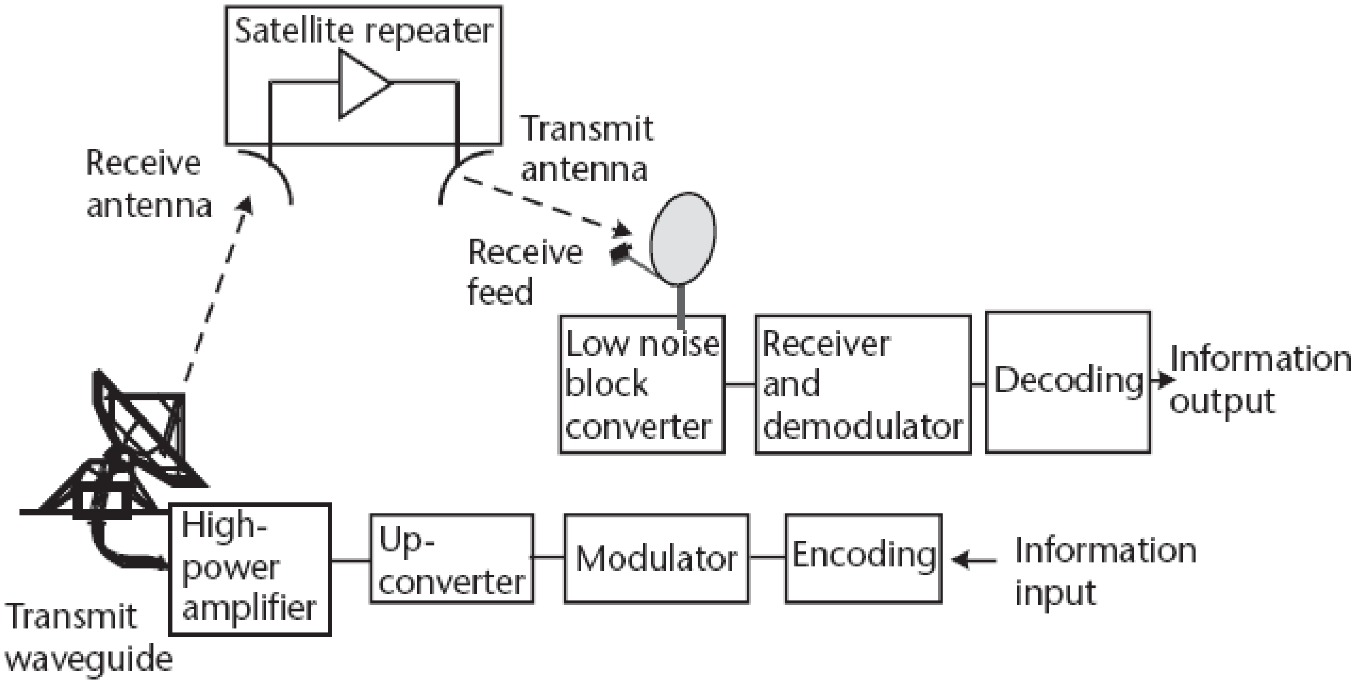
- ✓ Can reach over large geographical area. A single satellite can provide coverage to over 30% of Earth's surface.
- ▪ -With just 3 geosynchronous satellite we can cover the entire earth.
- ✓ Point to Multi point communication is possible.
- ✓ Only solution for isolated areas.
- ✓ Ideal for broadcast applications.
- ✓ No need for the local loop
- ✓ Wide bandwidths are available.
- ✓ Transmission cost and quality of signal is independent of distances.
- ✓ During critical condition earth stations can be removed and relocated easily.

Disadvantages:

- ✓ Satellite has been constructed for years. Moreover satellite design and development requires higher cost.

- ✓ Satellite once launched, requires to be monitored and controlled on regular periods so that it remains in the orbit.
- ✓ Satellite has life which is about 12-15 years. Due to this fact, another launch need to be planned before it becomes un-operational.
- ✓ Redundant components are used in the network design. This incur more cost in the installation.
- ✓ In the case of LEO/MEO, large number of satellites are needed to cover radius of earth. Moreover satellite visibility from earth is for very short duration which requires fast satellite to satellite handover. This makes system very complex.





- **Encoder:** ϖ The video/audio encoding system is fully complaint to DVB/MPEG-standards. ϖ One encoder is used for only one video/audio channel. ϖ For more channels more encoders are used. ϖ The output of the encoders are fed to multiplexer unit for multiple programs

- **Up Converter:** Up converter is a part to convert signal up for transmission. Basically, mixer part for frequency upward conversion is called UP CONVERTER. When input signal combines LO signal, RF signal is generated as much as input signal with LO signal.
  The up-conversion is required to raise the frequency of the signal in desired band: C-band, Extended C-band or Ku-band before transmission. ϖ The input to up converter is 70 MHz (output of modulator) and output of Up-converter is fed to HPA
- **High Power Amplifier:** The high power amplifier is used for the final power amplification of the digital RF signal in C-band/ Ku band that is fed to the antenna.
- **C-Band vs. KU-Band:** C-Band suitable in heavy rainfall areas, need larger bandwidth but; KU-Band is suitable in small area bcoz of smaller disc size, need smaller bandwidth and cheaper than C-Band.

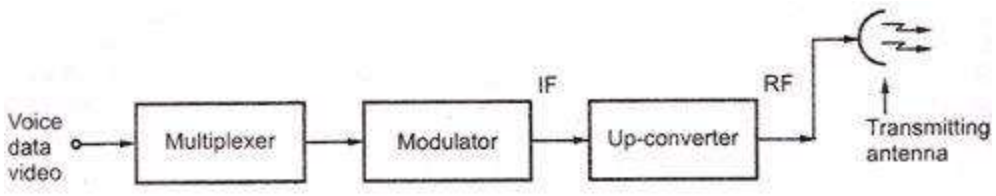
46. Terrestrial Microwave

Microwave transmission in the atmosphere can only take place when there is a direct line of sight between the sender's and receiver's antenna. This is why microwave transmission towers are speckled with antennas pointing in many directions -- they actually point at different microwave transmission towers. The absorption of microwaves in the atmosphere also means that there is very little interference between different microwave towers.

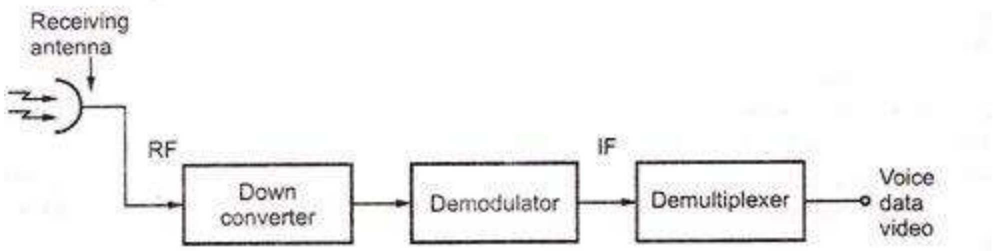
Advantages of Microwave Transmissions

Radio, including microwaves, is a form of energy transmission. Energy transmission at frequencies and wavelengths that are defined as microwaves tend to be absorbed by water molecules. This is why a microwave oven works. For microwave transmission, the water molecules in the atmosphere absorb the transmitted energy. The effect required for transmission is comparatively low for the amount of data transmitted because of the short distances afforded by the line-of-sight requirement. This is also true for satellites. A satellite can transmit at a relatively low effect, since there is nothing between it and the antenna.

47. Draw a generalized block diagram of microwave communication system and explain its components. Also calculate the free space loss.



(a) Microwave transmitter



(b) Microwave receiver

The voice, video, or data channels are combined by a technique known as multiplexing to produce a BB signal. This signal is frequency modulated to an IF and then up converted (heterodyned) to the RF for transmission through the atmosphere. The reverse process occurs at the receiver. The microwave transmission frequencies are within the approximate range 2 to 24 GHz.

The frequency bands used for digital microwave radio are recommended by the CCIR. Each recommendation clearly defines the frequency range, the number of channels that can be used within that range, the channel spacing the bit rate and the polarization possibilities.

48. Compare terrestrial microwave communication system with satellite system.

- ✓ Coverage area of a satellite based system is greater than that of a terrestrial based wireless communication system. A GEO satellite with one single antenna can cover about 1/4th of the earth.
- ✓ Satellite communications link will have more degradations compare to terrestrial communication link but quality of transmission is usually quite good.

- ✓ In a satellite link delay from earth to satellite to earth is about 240ms while in terrestrial link it will be far less. But transmission cost in a satellite system is independent of the distance within the area of coverage of the satellite antenna, while in terrestrial system it varies based on the distance.
- ✓ In a satellite based system satellite EIRP and bandwidth is very vital parameters which need to be carefully designed at the initial stage of both satellite and earth station point of view.
- ✓ Very high bandwidths and very high data rates are achievable in a satellite based communication system.
- ✓ In case of satellite based systems all the earth stations/VSATs can receive their own transmissions and hence transmitted power should be carefully decided based on the RF link budget. But both transmitting and receiving frequencies are different and hence will not create much problem. Transmit reject filter should be good enough to overcome this problem.
- ✓ Satellite communications only work when there is a line of sight from the communications satellite. So does terrestrial microwave communications. Both require parabolic antennas. This is because apart from the limited frequency bands used by satellite communications, terrestrial and satellite microwave communications are actually using the same technology, and the only difference is the distance between sender and receiver.

49. With a neat diagram, explain the working of a cellular GSM system.
50. WiMax
51. Design of an optical fiber communication system