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2 Hours

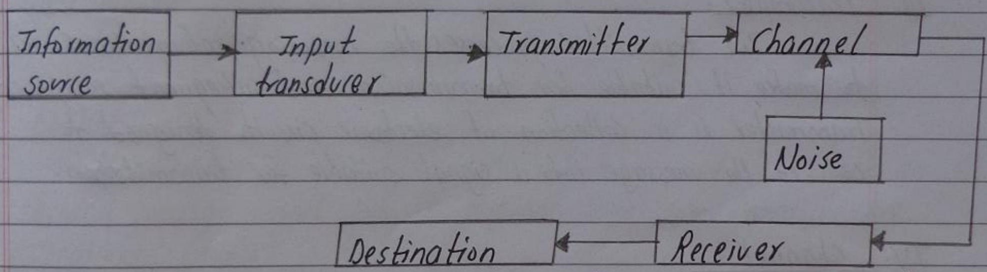
Date 1

1. Introduction : 1

1. Communication process : 2075/s

The goal of communication is to convey information. The communication process is divided into three basic components. The purpose of a communication system is to transmit message intelligence of a source to a user. Thus, communication may be defined as a process for information exchange. Communication refers to sending and receiving and processing the information by electrical means.

1.1 Elements of communication system :



The essential components of a communication system are information source, input transducer, transmitter, communication channel, receiver and destination.

i) Information source :

A communication system serves to communicate a message or information. This message or information originates in the information source. In general, there can be various messages in the form of words, group of words, code, symbols, sound signal, pictures etc. However, out of these messages, only the desired message is selected and conveyed or communicated. i.e, the function of information source is to produce required message which is need to be transmitted.

II. Input transducer:

The message from the information source may or may not be electrical form in nature. In case when the message produced by the information source is not electrical in nature, an input transducer is required to convert it into a time varying electrical signal.

For eg: in case of radio broadcasting a microphone converts the information which is in the form of sound waves into corresponding electrical signal.

III. Transmitter:

The transmitter processes the incoming information so as to make it suitable for transmission and subsequent reception. Transmitter is a collection of electronic circuits designed to convert the message into a signal suitable for transmission.

IV. Channel:

Channel means the medium through which the message travels from the transmitter to the receiver. It can be said that the function of the channel is to provide a physical connection between the transmitter and the receiver.

V. Noise:

Noise may interfere with signal at any point in the communication system but its impact is most noticeable when it occurs in the channel or at the input of the receiver. When the noise is severe, it can make the information useless. Here, interference and distortion can also occur.

vi Receiver :

once the signal has passed through the communication channel, it must be effectively captured by the receiver. The goal of the receiver is to capture and reconstruct the signal before it passed through transmitter. (the A/D converter, modulator, demodulator, encoder). This is done by passing the received signal through another circuit containing the following components;

- Noise Filter
- D/A Converter
- Decoder
- Demodulator
- Signal amplifier

vii Destination :

Destination is the final stage which is used to convert an electrical message signal into its original form. Some example are,

- speakers (audio)
- motors (movement)
- Lighting (visual)

For example, in radio broadcasting, the destination is a speaker which operates as a transducer i.e, it converts the electrical signal in the form of original sound signal.

1.2 Information source :

Analog Communication sources:

produce continuously varying message signals. microphone is a good example of analog source.

Digital Communication source : Eg: typewriter produce a finite set of possible messages. There are finite number of characters (messages) that can be emitted by this source.

1.3. Communication channel : 2016/S

The channel is a physical medium used to pass the signal from transmitter to receiver. Depending upon the medium, the channels can be subdivided into;

(i) wireline channel :

Extremely used in telephony, computer networks, as a link between transmitter and antenna. Eg: Twisted pair cable, co-axial cable, waveguides, optical fiber. Different media have different bandwidth ranging from few KHZ to hundred GHZ

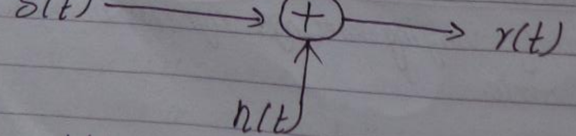
(ii) wireless channel :

Electromagnetic wave in the range of few KHZ to hundred GHZ is used as a signal carrier. Eg: Air, vacuum, sea water.

A channel can be modeled in the following manners;

(i) channel with additive noise only:

only noise is added to the signal and all other parameters of the channel is assumed to be constant.

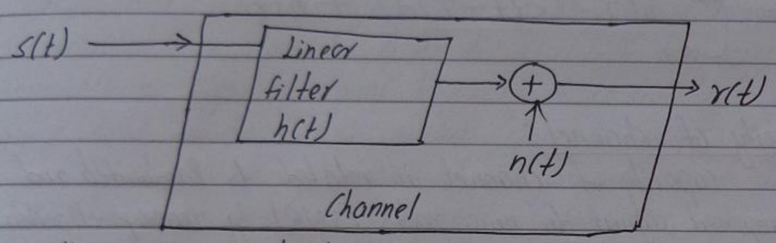


This model is suitable for simple wireline communication channel operating at relatively low frequency.

$$r(t) = s(t) + n(t)$$

(ii) channel with linear time invariant filter:

In most cases, filters are introduced at the inputs and outputs of the receiver and transmitter to limit the bandwidth and to avoid interference from or to other channels. In such cases, the above model is suitable for analysis. such model is suitable for more complex wireline channel.



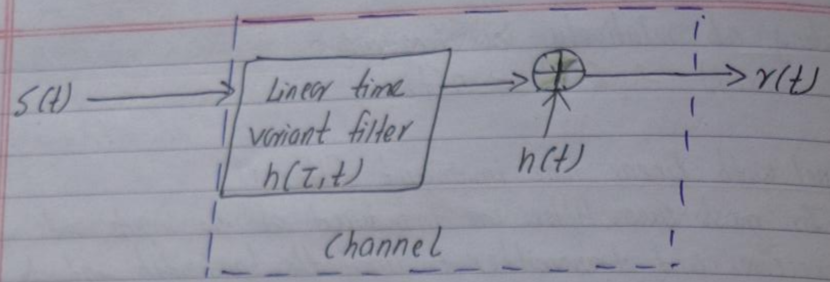
$$r(t) = s(t) * h(t) + n(t)$$

$$\therefore r(t) = \int_{-\infty}^{\infty} h(\tau) s(t-\tau)\, d\tau + n(t)$$

III Channel with linear time variant filter:

wireless channel experiences multipath fadding that randomly reduces or increases signal level at the receiver. The effect can be characterized mathematically by linear time time variant filter with impulse response $h(\tau, t)$. This response is the impulse response at instance 't' due to an impulse applied at $(t, -\tau)$, where, 'τ' is age or elapsed time.

It is suitable for the case of physical channels such as under water acoustic channel and ionosphere radio channels. $h(\tau, t)$ is the response of the channel at time t.



$\therefore\ r(t) = s(t) * h(\tau, t) + n(t)$

$= \int_{-\infty}^{\infty} h(\tau, t)\, s(t-\tau)\, d\tau + n(t)$

Capacity of channel:

Capacity of channel is related to Bandwidth and the required signal to noise ratio (SNR) by Shannon-Hartley capacity theorem.

$\therefore\ C = B \log_2 (1 + SNR)$ bits/sec

where,

C = channel capacity or the maximum rate at which information may be transmitted without error

B = channel bandwidth in Hz

Analog signal are those signals whose amplitude are continuously varying with respect to time and cannot be defined by finite number of amplitude levels.

Digital signals are those signals having only finite number of amplitude levels.

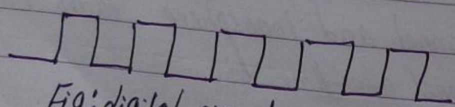


Fig: digital signals.

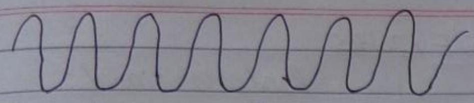


Fig: Analog signals.

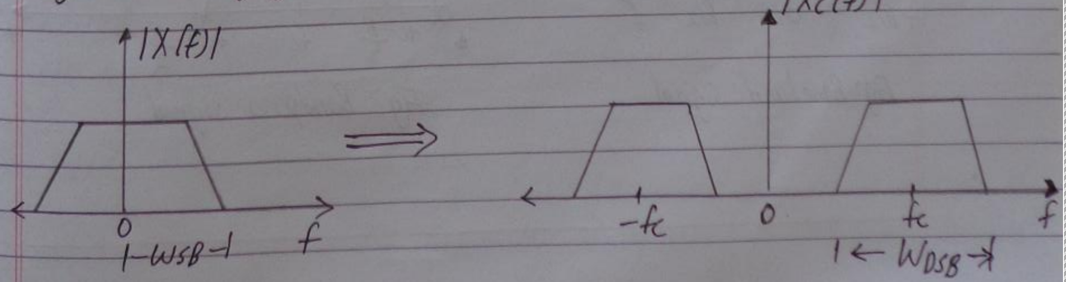
1.4 Baseband signal or Lowpass signal: 2015/F, 2014/S

A) A Baseband waveform has a spectral magnitude that is non-zero for frequency in the vicinity of the origin, (i.e f=0) & negligible elsewhere. In telecommunications, and signal processing, baseband is an adjective that describes signals and systems whose range of frequencies is measured from close to 0 Hz to a cutoff frequency, a maximum bandwidth or highest signal frequency. it is sometimes used as a noun for a band of frequencies starting close to zero.

It can often be considered as a synonym to Lowpass or non-modulated and antonyms to passband, bandpass, carrier-modulated or radio frequencies (RF).

B. Bandpass or Band limited signal: 2015/F, 2014/S

A Band limited signals are those whose spectral magnitude is non-zero within a certain range of frequencies or within a certain bandwidth. Bandpass signal is a class of signals, whose frequency domain representation X(f) is non-zero for frequencies in a small neighbourhood 'w' of some high frequency $\pm f_c$, the spectral magnitude is negligible elsewhere. It is also called as radio frequency.



Difference between: 2015/F, 2014/S

	Baseband signal	Bandpass signal:
i)	may have DC Component.	Doesnot have DC Component.
ii)	It is usually message signal.	It is usually the modulated signal or transmitted signal.
iii)	Refers to the signal & systems before modulation, which have frequencies/bandwidth much lower than the carrier frequency.	Refers to the signals & systems after modulation, which have frequencies/bandwidth around the carrier frequency.
iv	A baseband waveform has a spectral magnitude that is non zero for frequencies in the vicinity of the origin ($f=0$) and negligible elsewhere.	A Bandpass waveform has a spectral magnitude that is non-zero for frequencies in some band concentrated about a frequency $f = \pm f_c$, where $f_c >> 0$; f_c = carrier frequency.

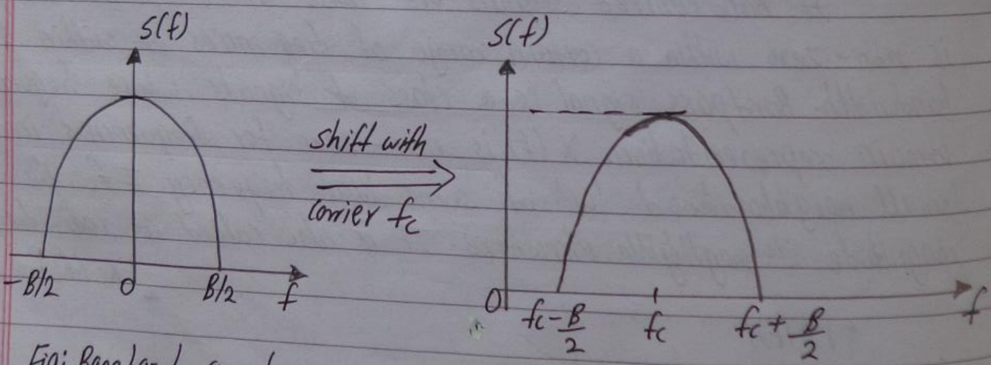


Fig: Baseband signal

Fig: Bandpass signal

11 Hours

9

2. Analog Communication:

2.1 The communication based on analog signals and analog value is called as analog communication. The process of superimposition of audio signal over the carrier wave is called the modulation. After modulation this wave can be fed to the antenna and the intelligence can be transmitted over a long distance. The carrier waves are of constant frequency and travel through space with a velocity of light i.e, 3×10^8 m/s.
The process of separating audio frequency signal from radio frequency carrier wave is known as detection or demodulation.

2.2 MODULATION: 2014/S

Any wave has three significant characteristics viz amplitude, frequency and phase, and modulation is a process of impressing information to be transmitted on a high frequency wave, called the carrier wave, by changing it's one of the characteristics.

Modulation may also be defined as the process of alternating some characteristics (amplitude, frequency or phase angle) of the carrier wave in accordance with the instantaneous value of some other wave called the modulating wave.

A) Carrier wave:

Carrier wave is a high frequency, constant amplitude, constant frequency and non-interrupted wave generated by radio frequency oscillators. These waves are inaudible ie, by themselves they are not able to produce any sound in the speaker.

Need For Modulation: 2015/S, 2014/S

Low frequency signals cannot be transmitted over-long distance if radiated directly into the space. This is because of following hurdles;

(i) short operating Range :

The energy of any wave depends on its frequency - the longer the frequency of the wave, the greater the energy associated with it. obviously, the audio signal having small frequency and consequently small power cannot be transmitted over long distance which radiated directly into the space. However, modulated wave can be transmitted over long distances.

(ii) Poor Radiation Efficiency:

At audio frequencies, radiation is not practiceable as efficiency of radiation is poor. However, electrical energy can be radiated efficiently at high frequencies (above 20 KHZ)

(iii) mutual interference :

If Low frequency signal are transmitted directly from different sources, all of them would be mixed up and completely blanket the air. However, by modulation different messages of different frequency levels can be transmitted simultaneously without any interference.

(iv) Huge Antenna Requirement :

For efficient radiation of a signal, the length of transmitting and receiving antenna should be atleast one quarter length, wavelength, ie,

$$l = \frac{1}{4} \text{ wavelength} = \frac{1}{4} \times \frac{\text{velocity}}{\text{frequency (HZ)}} \text{ meter}$$

$$= \frac{1}{4} \times \frac{3 \times 10^8}{f}$$

$$\therefore l = 75 \times \frac{10^6}{f} \text{ meter}$$

Thus, for transmitting a signal of frequency 2KHz, an antenna of length 37.5 km will be required, practically impossible. On the other hand, for transmitting a signal of frequency 2 MHz, an antenna of about 37.5 meter would be required which can be easily constructed.

Since, $\lambda = \frac{c}{f}$

c: velocity of light
f = frequency of the signal to be transmitted.

v. 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.

vi 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.

B. Types of modulation:

The sinusoidal carrier wave may be represented as,

$V_c = V_c \sin(\omega_c t + \theta)$
$= V_c \sin(2\pi f_c t + \theta)$

where,
V_c = maximum value
f_c = frequency

θ = phase relation w.r.to some reference of the carrier wave.

In amplitude modulation, the amplitude of the carrier wave is varied in accordance with the modulating signal, keeping the frequency and phase of the carrier wave unchanged.

In frequency modulation, the frequency of the carrier wave is varied in accordance with the modulating signal, keeping the amplitude and phase of the carrier wave unchanged.

In phase modulation, the phase of the carrier wave is varied in accordance with the modulating signal, keeping the amplitude and frequency of the carrier wave unchanged.

However, modulation can also be classified, according to the nature of carrier wave, into continuous wave modulation and pulse modulation. The digital form of pulse modulation is known as pulse-code-modulation (PCM).

c) Amplitude Modulation:

The process of varying amplitude of the high frequency or carrier wave in accordance with the intelligence (code, voice or music) to be transmitted, keeping the frequency & the phase of the carrier wave unchanged is known as amplitude modulation.

Time domain expression and spectrum (Frequency domain representation):

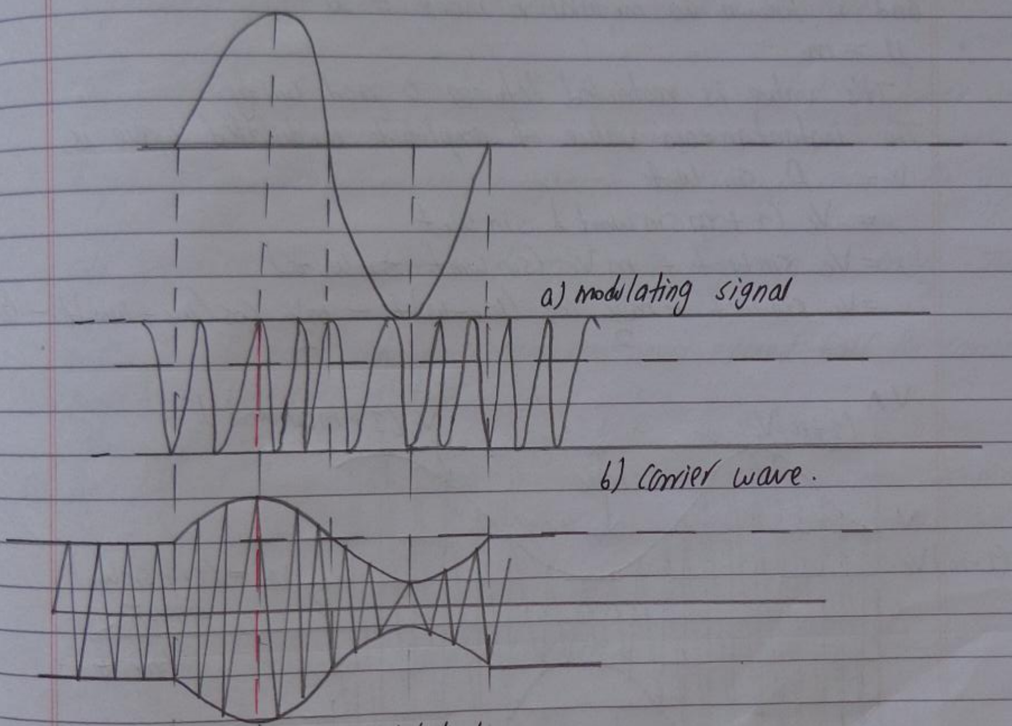
let, the carrier and modulating voltage waves be represented as,

$$v_c = V_c \sin \omega_c t \quad \text{——} \quad (1)$$

13

and, $v_m = V_m \sin \omega_m t$ ——— (11)

where, v_c, V_c and ω_c are instantaneous value, peak and angular velocity of the carrier and v_m, V_m, ω_m are instantaneous value, maximum value and angular velocity of the modulating signal. phase angle has been ignored in both eqn as it remains unchanged in amplitude modulation process.



c) amplitude modulated wave.

The amplitude of the carrier wave varies at a modulating signal frequency f_m. The amplitude of amplitude modulate wave is given as,

$$A = V_c + v_m$$

$$= V_c + V_m \sin \omega_m t$$
$$= V_c \left[1 + \frac{V_m}{V_c} \sin \omega_m t\right]$$
$$= V_c [1 + m \sin \omega_m t] \quad \text{--- (iii)}$$

where,
m = ratio of peak values of modulating signal & carrier wave and is known as modulation index = μ

$\therefore \mu = m$

It's value is restricted between 0 and unity.

The instantaneous value of amplitude modulated wave is,

$$v = A \sin \omega_c t$$
$$= V_c (1 + m \sin \omega_m t) \sin \omega_c t$$
$$= V_c \sin \omega_c t + m V_c (\sin \omega_m t \sin \omega_c t)$$
$$= V_c \sin \omega_c t + \frac{mV_c}{2} \cos(\omega_c - \omega_m)t - \frac{mV_c}{2} \cos(\omega_c + \omega_m)t \quad \text{--- (iv)}$$

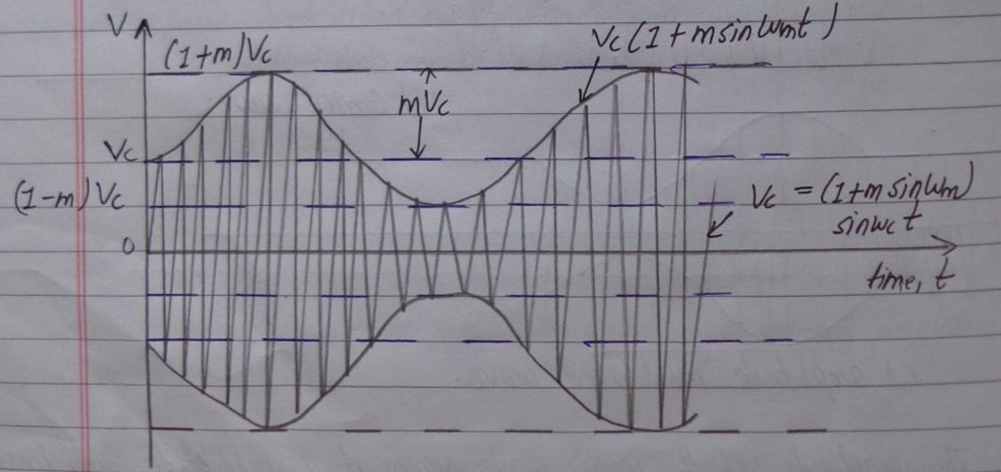


Fig: amplitude modulated sinewave with $m<1$

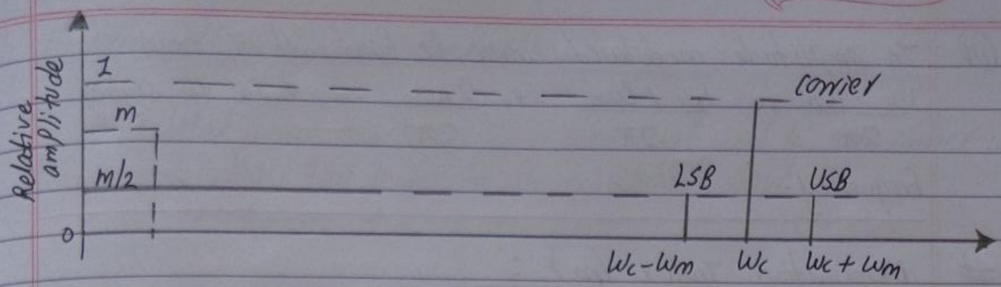


Fig: Frequency spectrum of a sinusoidally modulated wave.

The lower side frequency component $\frac{W_c - W_m}{2\pi}$ is called the lower side frequency. The upper frequency component $\frac{W_c + W_m}{2}$ is called the upper side frequency. The amplitude of each side frequency is $m\frac{V_c}{2}$. The amplitude of each side frequency can never exceed half the carrier amplitude because $m \not> 1$.

Hence, During the process of amplitude modulation;

(i) the original carrier frequency is not altered but two new frequencies $\frac{W_c + W_m}{2\pi}$ and $\frac{W_c - W_m}{2\pi}$, known as sideband frequencies are produced.

$\frac{W_c + W_m}{2}$ = upper sideband (USB)

$\frac{W_c - W_m}{2}$ = lower " (LSB)

(ii) The carrier voltage component doesnot transmit any information because the signal frequency $W_m/2\pi$ is contained in sidebands only which is evident from above figure.

(iii) In amplitude modulated wave, the bandwidth is from $\frac{\omega_c - \omega_m}{2\pi}$ to $\frac{\omega_c + \omega_m}{2\pi}$ ie, $\frac{2\omega_m}{2\pi}$ or twice the signal frequency.

Modulation Index (m) :

The extent to which the amplitude of the carrier wave is varied by the modulating signal is called degree of amplitude modulation or modulation index and is denoted by m or μ. Hence, the ratio of change in amplitude of carrier wave to the amplitude of normal carrier wave is called the modulation index, m.

ie,

$$m = \frac{V_m}{V_c}$$

$$\therefore m = \frac{\text{amplitude change of carrier wave}}{\text{amplitude of normal/unmodulated carrier wave}}$$

$$\% m = \% \mu = \frac{V_m}{V_c} \times 100$$

Also,

$$\therefore m = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} = \mu$$

Where,

V_{max} = maximum value of amplitude of modulated carrier wave.

V_{min} = minimum " " " " " "

D. Efficiency (power and Bandwidth):

Modulated wave has more power than that had by the carrier wave before modulation. Total power in the modulated wave is,

$$P_{total} = P_{carrier} + P_{LSB} + P_{USB} \quad \text{— (i)}$$

When an amplitude modulated wave is impressed upon some resistance (say antenna resistance), R, then

$$P_{carrier} = \frac{(V_c/\sqrt{2})^2}{R} = \frac{V_c^2}{2R} \quad \text{— (ii)}$$

Each side band has peak value of $\frac{\mu}{2} V_c$ and rms value of $\frac{\mu}{2} \cdot \frac{V_c}{\sqrt{2}}$.

Hence, power in each sideband is,

$$P_{LSB} = P_{USB} = \frac{\left(\frac{\mu}{2} \cdot \frac{V_c}{\sqrt{2}}\right)^2}{R} = \frac{\mu^2 V_c^2}{8R} = \frac{\mu^2}{4} \cdot \frac{V_c^2}{2R}$$

$$\therefore P_{LSB} = \frac{\mu^2}{4} \cdot P_{carrier} \quad \text{— (iii)}$$

We have,

$$P_{total} = P_c + P_{LSB} + P_{USB}$$
$$= \frac{V_c^2}{2R} + \frac{\mu^2}{4} \cdot \frac{V_c^2}{2R} + \frac{\mu^2}{4} \cdot \frac{V_c^2}{2R}$$

$$= \frac{V_c^2}{2R}\left(1 + \frac{\mu^2}{2}\right)$$

$$\therefore P_t = P_{carrier}\left(1 + \frac{\mu^2}{2}\right) \quad \text{— (iv)}$$

(i) Maximum power in the amplitude modulated wave (without distortion) will occur for $m = 1$,
ie, $P_{total} = 1.5\, P_{carrier}$ when $m = 1 = \mu$

(ii) Ratio of P_{SB} and P_{total} is given as,

$$\frac{P_{SB}}{P_t} = \frac{\frac{\mu^2}{4} \cdot P_c + \frac{\mu^2}{4} P_c}{P_c\left[1 + \frac{\mu^2}{2}\right]}$$

or $$\frac{P_{SB}}{P_t} = \frac{m^2/2}{1 + \frac{m^2}{2}} \qquad [\because \mu = m]$$

For $m = 1$,

$$\therefore \frac{P_{SB}}{P_t} = \frac{1/2}{3/2} = \frac{1}{3}$$

i.e, only one-third of the total power of the modulated wave is contained in the two sidebands and rest of the two-third power lies in the carrier component, which is of no-use.

(iii) In most applications, carrier is simultaneously modulated by several sinusoidal modulating signals. In such case, the total modulation index is,

$$\mu_t = \sqrt{\mu_1^2 + \mu_2^2 + \mu_3^2 + \cdots}$$

(iv) If I_c and I_t represents the rms value of unmodulated or carrier current and total modulated current and R is the resistance through which the current flow, then,

$$\frac{P_{total}}{P_{carrier}} = \frac{I_t^2 R}{I_c^2 R} = \left(\frac{I_t}{I_c}\right)^2$$

Also,

or, $$\frac{P_{total}}{P_{carrier}} = 1 + \frac{\mu^2}{2}$$

Hence,

$$\left[\frac{I_t}{I_c}\right]^2 = 1 + \frac{\mu^2}{2}$$

$$\therefore \quad I_t = I_c \sqrt{1 + \frac{\mu^2}{2}}$$

Limitations of Amplitude modulation (AM):

i) **Low efficiency:**

In AM, useful power that lies in the sidebands is quite small, so the efficiency of AM system is low.

(ii) **Limited operating Range:**

Transmitters employing amplitude modulation have small operating range. This is due to low efficiency. Hence, information cannot be transmitted over long distances.

(iii) **Noisy reception:**

In case of AM, the reception is generally noisy. This is because a radio receiver cannot distinguish between the amplitude variations that represent noise and those contain the desired signal.

iv) Poor audio quality:

In order to attain high-fidelity reception, all audio frequencies upto 15 kHz must be reproduced and this necessitates the bandwidth of 30 kHz while the AM broadcasting stations are assigned bandwidth of only 10 kHz to minimize the interference from the adjacent broadcasting stations. Hence, in AM broadcasting stations audio quality is usually poor.

2.3 Numericals :

1. A) A sinusoidal carrier wave of frequency 1MHz & amplitude of 100 V is amplitude modulated by a sinusoidal voltage of frequency 5 kHz producing 50% modulation. Calculate frequency and amplitude of USB and LSB.

Soln,

⟹ Frequency of carrier, $f_c = 1$ MHz

" " modulating signal, $f_m = 5$ kHz

modulation index, $\mu = 50\%$

$= 0.5$

Lower sideband frequency $= f_c - f_m$

$= 1000 - 5$

$= 995$ kHz

Upper sideband frequency $= f_c + f_m$

$= 1000 + 5 = 1005$ kHz

$\therefore$ Amplitude of each sideband $= \mu \cdot \frac{V_c}{2}$

$= 0.5 \times \frac{100}{2}$

$\therefore A = 25$ V

Q). The maximum or crest amplitude of a sinusoidally modulated wave is 1.6 V, it's minimum or trough amplitude is 200 mV, and its average (unmoulded) amplitude is 900 mV. The maximum amplitude of modulating signal is 700 mV. Find the modulation index by different relations.

soln,

⇒ maximum or crest amplitude, $V_{max} = 1.6\ V$

minimum or trough ", $V_{min} = 200\ mV = 0.2\ V$

Average (unmodulated) amplitude,

$$V_c = \frac{1600 + 200}{2}\ mV = 900\ mV = 0.9\ V$$

max. amplitude of modulating signal,

$$V_m = 700\ mV = 0.7\ V$$

i) modulation index, $\mu = \frac{V_m}{V_c} = \frac{0.7}{0.9} = 0.777$

ii) $$\mu = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} = \frac{1.6 - 0.2}{1.6 + 0.2} = 0.777$$

iii) $$\mu = \frac{V_{max} - V_{min}}{2V_c} = \frac{1.6 - 0.2}{2 \times 0.9} = 0.777$$

iv. $$\mu = \frac{V_{max} - V_c}{V_c} = \frac{1.6 - 0.9}{0.9} = 0.777$$

v. $$\mu = \frac{V_c - V_{min}}{V_c} = \frac{0.9 - 0.2}{0.9} = 0.777$$

The result is same, as expected.

C. The rms antenna current of a radio transmitter is 10 A when unmodulated rising to 12 A, when the carrier is sinusoidally modulated. calculate the modulation index.

Sol^n,

=> carrier current, $I_c = 10$ A

Total modulated current, $I_t = 12$ A

modulation index, $\mu = \sqrt{\left[\left(\frac{I_t}{I_c}\right)^2 - 1\right] \times 2}$

$= \sqrt{\left[\left(\frac{12}{10}\right)^2 - 1\right] \times 2}$

since, $\frac{I_t}{I_c} = \sqrt{1 + \frac{m^2}{2}}$

$\therefore \mu = 0.938$

D. Rms value of RF voltage after amplitude modulation to a depth of 60% by a sinusoidal voltage is 60 V. Calculate the rms value of modulated voltage when modulated to depth of 75%.

Sol^n,

=> For 60% modulation, modulation voltage,

$V_t(rms) = 60$ V

Also,

$\frac{P_{total}}{P_{carrier}} = \frac{V_t^2/R}{V_c^2/R} = \left(\frac{V_t}{V_c}\right)^2$

So, we have,

$\left(\frac{V_t}{V_c}\right)^2 = 1 + \frac{m^2}{2}$

or $V_c = \frac{V_i}{\sqrt{1+\frac{\mu^2}{2}}}$

$= \frac{60}{\sqrt{1+\frac{(0.6)^2}{2}}}$

$\therefore V_c = 55.23 \text{ V}$

Rms value of modulated voltage with 75 % modulation,

$= V_c\sqrt{1+\frac{\mu^2}{2}}$

$= 55.23 \times \sqrt{1+\frac{(0.75)^2}{2}}$

$\therefore = 62.5 \text{ V}$

E. In an AM system, the modulating signal is sinusoidal with frequency of fm Hz. If 80 % modulation is used, determine ratio of the total sideband power in the modulated signal.

sol^n,

$\Rightarrow \mu = 80\% = 0.8$

$\frac{\text{sideband power, } P_{SB}}{\text{total power in modulated wave, } P_{total}}$

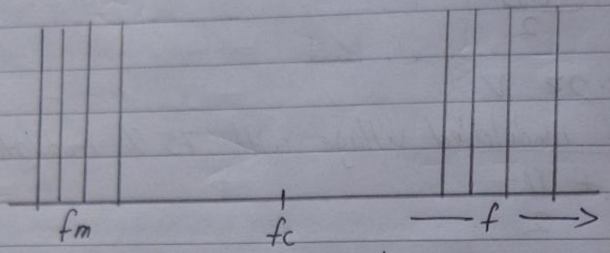
$= \frac{m^2/2}{1+\frac{m^2}{2}}$

$= \frac{0.8^2/2}{1+(0.8^2/2)}$

$\therefore \frac{P_{SB}}{P_{total}} = 0.242$

2.4) DSB-SC modulation:

In DSBSC system, as its name indicates, carrier component is altogether removed resulting in saving of enormous amount of power.



It is known that carrier signal contains two-third of total transmitted power for 100 % modulation.

A) Generation of DSBSC : 2015/S

DSBSC can be obtained by simply multiplying the modulating signal with the carrier signal. By simple multiplication of $V_c \sin w_c t$ and $V_m \sin w_m t$, we have the lower and upper sidebands without carrier is,

$$V_c \times V_m = V_c \sin w_c t \times V_m \sin w_m t$$
$$= \frac{V_c V_m}{2} \times 2 \sin w_c t \times \sin w_m t$$
$$= \frac{V_c V_m}{2} [\cos (w_c - w_m)t - \cos (w_c + w_m)t] \quad \text{--- (1)}$$

It means that a DSBSC signal is obtained by simply multiplying modulating signal ($V_m \sin w_m t$) with carrier signal ($V_c \sin w_c t$). This is achieved by a product modulator. There are two types of product modulator ; Balanced and Ring modulator.

25

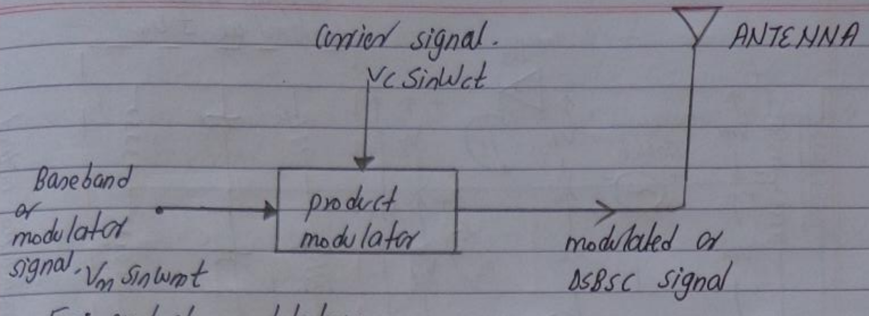


Fig: product modulator.

Balanced modulator : 2015/s

DSBSC contains only two sidebands. Thus, if two non-linear devices such as diodes, transistors etc are connected in a balanced mode so as to suppress the carriers of each other, then only sidebands are left. ie, DSB-SC signal is produced. Hence, a balanced modulator may be defined as a circuit in which two non-linear devices are connected in a balanced mode to generate a DSBSC signal.

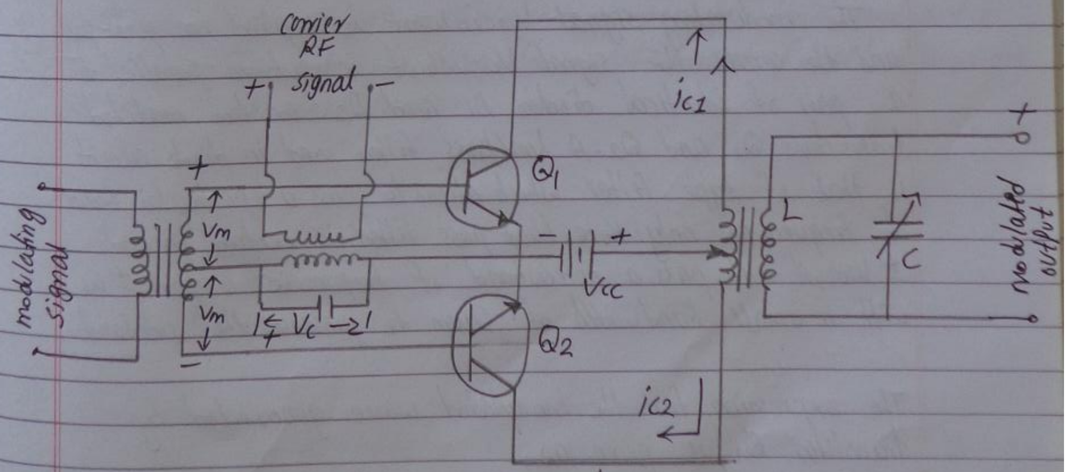


Fig: Balanced modulator using Transistors.

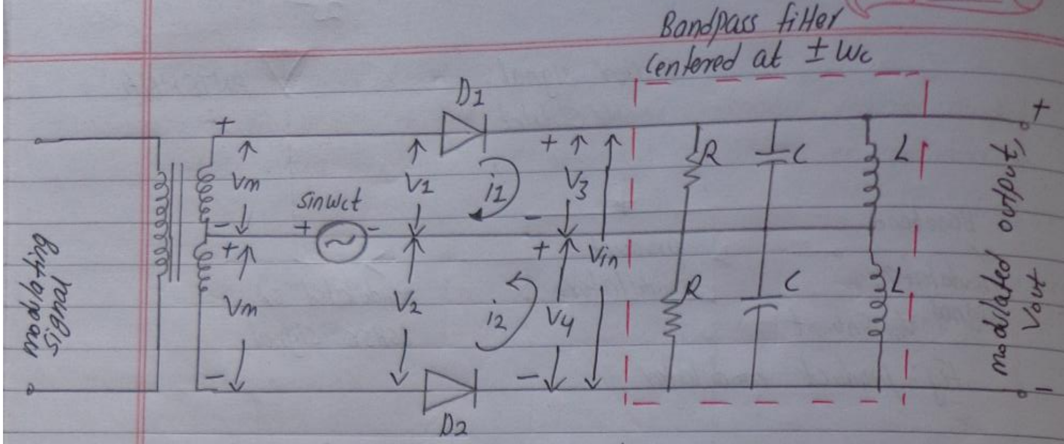


Fig: Balanced modulor using Diode.

It is assumed that both diodes and transistors used in circuits have identical characteritics and that the circuit is symmetrical with respect to the centre tap of the transformer. As the two halves have to be matched or balanced, such modulator is known as a balanced modulator.

The modulating signal $V_m \sin w_m t$ is applied in push-pull and the carrier RF signal $V_c \sin w_c t$ is put in parallel to a pair of identical diodes D_1 and D_2. or two matched transistors Q_1 and Q_2. A bandpass filter used in diode circuit is that of type filter which allows to pass a particular band of frequencies only. Here, band pass filter is centred around $\pm w_c$, it will pass a narrowband of frequencies centred $\pm w_c$ with a small bandwidth of $2w_m$ to preserve the sidebands.

The expression for the modulated wave generated by transistor Q_1 is given as,

$$V_1 = V_c \, (1 + m \sin w_m t) \sin w_c t$$

$= V_c \sin w_c t + m V_c \sin w_m t \cdot \sin w_c t$

The modulated wave generated by transistor Q2 will be same as generated by Q1 except that it will have a phase difference of 180° (ie, reversed phase), So,

$V_2 = V_c \sin w_c t - m V_c \sin w_m t \sin w_c t$

At the output transformer, output voltage, Vout, due to push-pull arrangement, is proportional to $(V_1 - V_2)$, So,

modulated output, $V_{out} = V_1 - V_2$

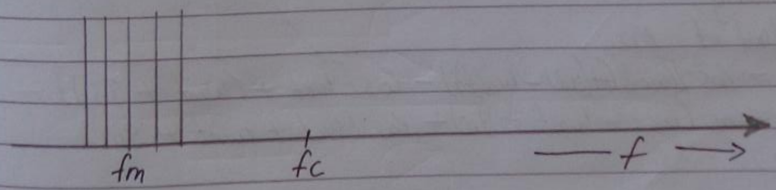
$= 2 m V_c \sin w_m t \cdot \sin w_c t$

$= m V_c [\cos (w_c - w_m)t - \cos (w_c + w_m t)]$

Thus, it is proved that the output signal consists of only side-bands and the carrier component is eliminated.

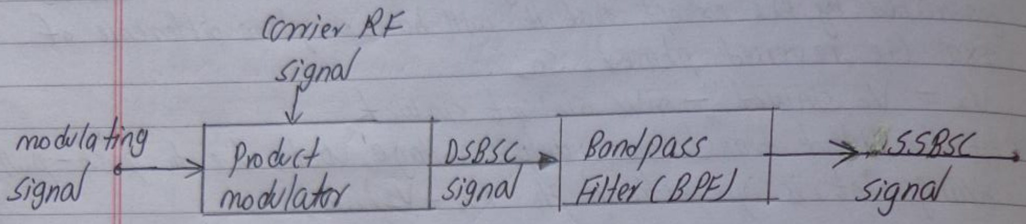
2.5 SSB Modulation: 2016/S

It consists in transmitting only one sideband and suppress the other sideband and the carrier. It utilizes the fact that the intelligence or messages is contained in each sideband and not in the carrier. Two sidebands being the exact images of each other, carry the same audio intelligence.



Generation:

SSB can be obtained by passing the output of a carrier suppressor (product modulator) through filter circuits that are selective to transmit one sideband while suppressing the other.



mathematical analysis:

let the expression for modulating signal & carrier signal be $\sin w_m t$ and $\sin w_c t$ respectively,

Now, the balanced modulator M_1 will receive $\sin w_m t$ and $\sin(w_c t + 90^\circ)$ and modulator M_2 will receive $\sin(w_m t + 90^\circ)$ and $\sin w_c t$.

We know, output of Balanced modulator M_1 will contain sum and difference frequencies,

Hence,

$$V_1 = \cos[(w_c t + 90^\circ) - w_m t] - \cos[(w_c t + 90^\circ) + w_m t]$$
$$= \cos[(\underbrace{w_c t - w_m t + 90^\circ)}_{LSB} - \underbrace{\cos(w_c t + w_m t + 90^\circ)}_{USB}]$$

Similarly,

output of M_2,

$$V_2 = \cos[w_c t + (90^\circ - w_m t)] - \cos[w_c t + (w_m t + 90^\circ)]$$
$$= \cos(w_c t - w_m t - 90^\circ) - \cos(w_c t + w_m t + 90^\circ)$$

Thus,

$$\therefore V_{out} = V_1 + V_2$$
$$= 2\cos(w_c t - w_m t + 90^\circ) \text{ --- LSB}$$
$$= |2V_1 - V_2|$$
$$= 2\cos(w_c t + w_m t + 90^\circ) \text{ --- USB}$$

The extent of SSB suppression depends upon:
i) Balancing capacity of modulator
ii) Control accuracy of the 90° phase of the carrier wave.
iii) error in maintenance of 90° shift between two modulating wave.

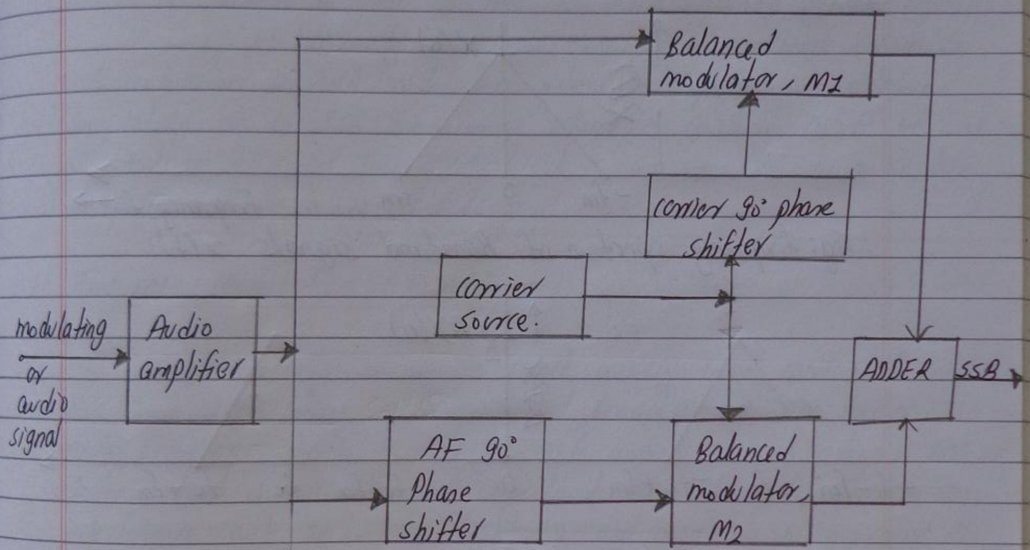


Fig: phase shift method for SSB generation

2.6 Vestigial SideBand Transmission (VSB): 2015/F, 2014/S

A VSB modulation is actually a compromise between DSBSC and SSB modulation system. In other words, it can be said that it is an optimum choice in which the advantages of DSBSC and SSB modulation systems have been explained.

As a matter of fact, the generation of VSB modulation signals is easier than other modulated signals such as conventional AM, DSBSC, and SSB signals. It's bandwidth is only slightly higher than

SSB signals but considerably less than DSBSC signals. SSB modulation is rather most suited for the transmission of voice signals because of the energy gap that exists in the frequency spectrum of the voice signals between zero and few hundred Hertz.

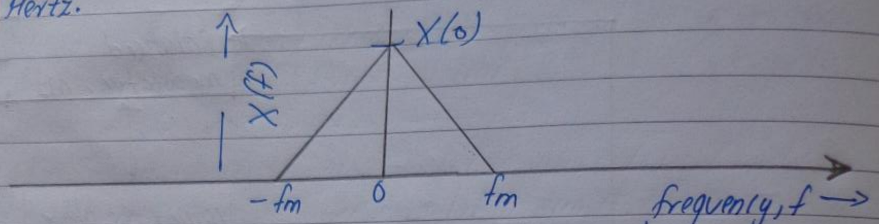
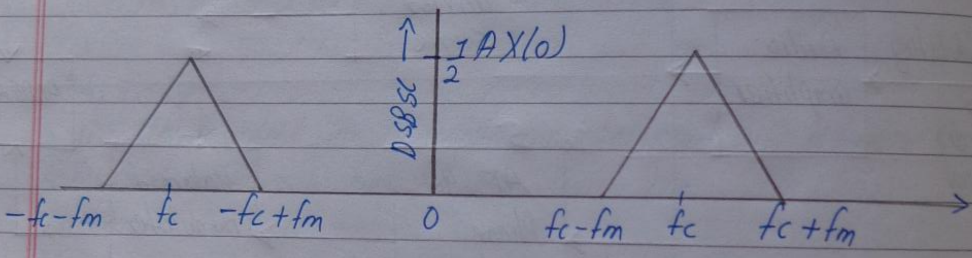


Fig: frequency spectrum of Baseband signals x(t)



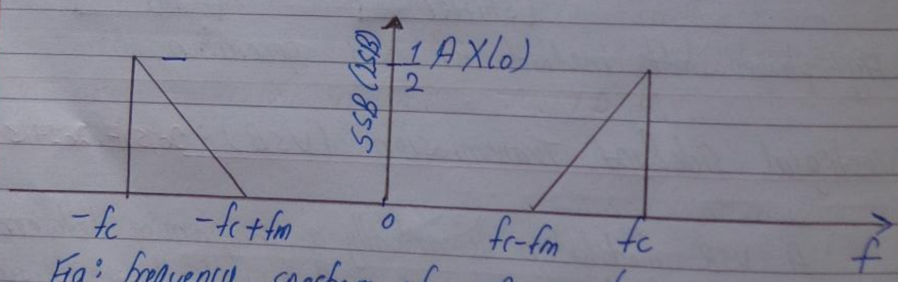
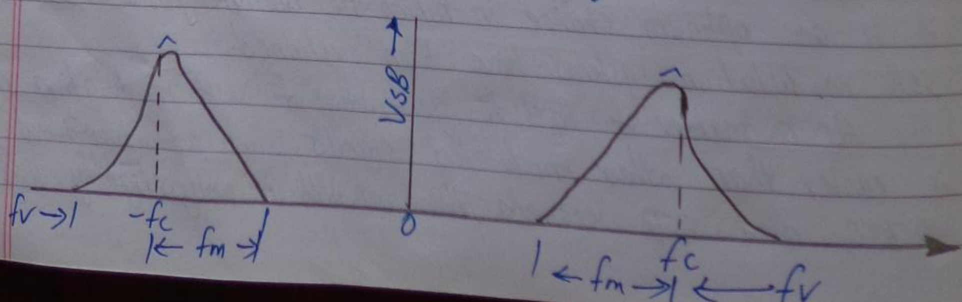


Fig: frequency spectrum of SSB signal.



When signals contain frequency components at extremely low frequencies, (as in telegraph frequencies) the USB and LSB of the translated signal tend to meet at the carrier frequencies. Under such circumstances, it becomes very difficult to isolate the unsuitable one sideband from the other. Hence, SSB scheme becomes unsuitable for handling such type of signals.

This difficulty has been overcome in a scheme known as VSB modulation. In VSB modulation, instead of rejecting one sideband completely as in SSB modulation scheme, a gradual cutoff of one sideband is allowed. This gradual cut is compensated by a vestige or portion of the other sideband.

2.7) Frequency Modulation (F.M):

It is virtually one of the most widely used modes of modulation. From applications in commercial broadcasting, FM radio, TV audio, cordless phone to cellular and mobile communication, FM is indeed both a reliable and important form of modulation. In AM, interference such as static, lightning and manmade noise, cause the amplitude of an RF signal to vary widely. This is because such noises, are predominately amplitude modulated signals in composition. The noise is added and superimposed on the transmitted AM signal carrying the desired intelligence.

2.7.A Analysis of FM carrier wave:

let, the carrier and modulating voltage wave be represented as,

$$v_c = V_c \cos(\omega_c t + \theta) \quad \text{—— (I)}$$

$$v_m = V_m \cos \omega_m t \quad \text{—— (II)}$$

where, v_c, V_c, ω_c and θ are the instantaneous value, peak value, angular velocity and the initial phase angle of the carrier and v_m, V_m and w_m are the instantaneous value, peak value and the angular velocity of the modulating signal.

let,

$$\phi_c(t) = w_c t + \theta \quad \text{--- (iii)}$$

$\phi_c(t)$ = total instantaneous phase angle of the carrier wave. at time, t.

$$v_c = V_c \cos \phi_c(t) \quad \text{--- (4)}$$

The instantaneous angular velocity w_c, defined as the instantaneous rate of increase of instantaneous phase, is related to phase angle, ϕ_c as,

$$w_c = \frac{d\phi_c}{dt} \quad \text{--- (5)}$$

In FM, the frequency of the carrier wave varies with time in accordance with the instantaneous value of modulating voltage, Hence, the frequency of the carrier and frequency modulation is given as,

$$w = w_c + k_f v_m$$

$$= w_c + k_f V_m \cos w_m t \quad \text{--- (6)}$$

where,

k_f = constant proportionality ie, frequency conversion factor.

The instantaneous phase of FM wave is obtained by integrating eqn (6),

$$\phi(t) = \int w \cdot dt = \int (w_c + k_f \cdot V_m \cos w_m t)\, dt$$

$$= w_c t + k_f V_m \cdot \frac{1}{w_m} \sin w_m t + \theta_1$$

θ_1 = initial phase.
The initial phase θ_1 can be neglected since it is insignificant in the modulation process, so,

$$\phi(t) = \omega_c t + k_f \cdot \frac{V_m}{\omega_m} \sin \omega_m t \quad \text{--- (7)}$$

The eqⁿ of FM wave is,

$$V_{FM} = V_c \sin \phi(t)$$
$$= V_c \sin\left[\omega_c t + k_f \cdot \frac{V_m}{\omega_m} \sin \omega_m t\right] \quad \text{--- (8)}$$

The instantaneous frequency of FM wave,

$$f = \frac{\omega}{2\pi}$$
$$= \frac{\omega_c}{2\pi} + k_f \frac{V_m}{2\pi} \cos \omega_m t \quad \text{--- (9)}$$
$$= f_c + k_f \frac{V_m}{2\pi} \cos \omega_m t \quad \text{--- (10)}$$

From eqⁿ (10),
maximum & minimum frequency are,

$$\therefore f_{max} = f_c + k_f \cdot \frac{V_m}{2\pi} \qquad \because \cos 0^\circ = +1$$

$$\therefore f_{min} = f_c - k_f \cdot \frac{V_m}{2\pi} \qquad \because \cos 180^\circ = -1$$

Thus, the frequency deviation, defined as the maximum change in frequency from mean value, f_c is given as,

$$f_d = f_{max} - f_c = f_c - f_{min} = k_f \cdot \frac{V_m}{2\pi} \quad \text{--- (11)}$$

The total variation in frequency from the minimum to maximum value, ie, $f_{max} - f_{min}$ is called carrier swing and it is given as,

$$\text{carrier swing}(cs) = 2 \cdot f_d$$
$$= k_f \cdot \frac{V_m}{2\pi}$$

The frequency modulation index, m_f or μ_f defined as the ratio of frequency deviation to modulation frequency,

$$m_f = \mu_f = \frac{\omega_d}{\omega_m} = \frac{k_f \cdot V_m}{\omega_m} = \delta$$

Hence, From eq^n (8),

$$\therefore V_{fm} = V_c \sin[\omega_c t + m_f \sin \omega_m t]$$
$$= V_c[\sin \omega_c t \cos(m_f \sin \omega_m t) + \cos \omega_c t \sin(m_f \sin \omega_m t)]$$

unlike amplitude modulation, the frequency modulation index can exceed unity.

2.7. B NarrowBand & wideband FM : 2016/S, 2.

I) NarrowBand FM:

- modulation index is less than 1.
- maximum modulation frequency is usually 3 kHZ & maximum frequency deviation is 75 kHZ
- A narrowband Fm wave consists of a carrier, an upper side - frequency component & lower side component.
- The modulated narrowband signal differs from the ideal response in two fundamental aspects;

+ The envelope consists a residual AM, so it varies with time.
+ For sinusoidal modulating wave, the angle $\theta(t)$ contains harmonic distortion in the form of 3rd and higher order harmonics of modulation frequency, f_m.

By restricting $\beta \leq 0.3$ radians, the effects of residual AM & harmonic PM are limited to negligible labels. A narrow band signal may be represented by phasor diagram as,

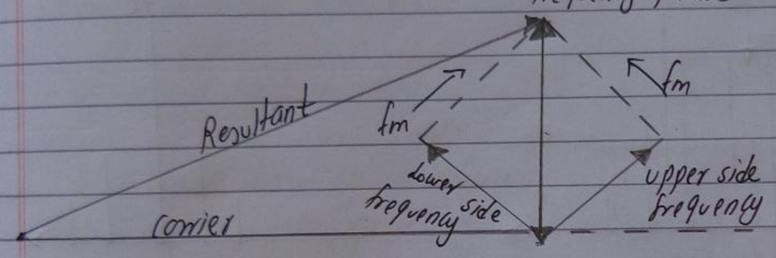


Fig: phasor diagram of narrowband FM

Uses:

mobile communications services such as police wireless, ambulance, taxicabs, short range very high frequency (VHF) ship to shore sources and defence.

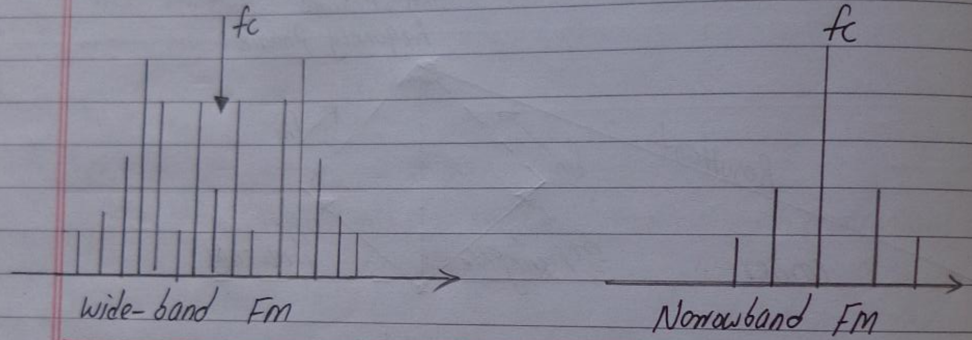
II) Wideband / Broadband FM:

- modulation index normally exceeds unity.
- modulation frequencies extend from 30 Hz to 15 KHz
- maximum permissible deviation = 75 KHZ
- wideband FM system needs large bandwidth, typically 15 times that

of narrowband FM system.

- Used in entertainment broadcasting.
- For large values of β, compared to 1 radian, the FM wave contains a carrier and infinite number of side-frequency components located symmetrically around the carrier.

- The amplitude of the carrier component contained in a wide band FM wave varies with the modulation index, β in accordance with Bessel function $J_n(\beta)$.



C. Transmission Bandwidth of FM signal / Carson's Rule : 2015/S 2017/F

In practice, FM wave is limited to finite number of significant side-frequencies compatible with a specified amount of distortion. Thus, an effective bandwidth is required for the transmission of an FM wave.

Approximate rule for transmission bandwidth of an FM wave generated by a single-tone modulating wave of frequency f_m is,

$$B_T \simeq 2\Delta f + 2f_m$$
$$= 2\Delta f\left(1 + \frac{1}{\beta}\right)$$

This relation is known as Carson's Rule.
According to international regulations of FM broadcast,
maximum frequency deviation, $f_d = \pm 75$ kHz
allowable bandwidth per channel, $= 200$ kHz

In telecommunication, carson's bandwidth rule defines the approximate bandwidth requirement of communication system components for a carrier signal that is frequency modulated by a continuous or broad spectrum of frequencies rather than a single frequency. Carson's rule doesnot apply well when the modulating signal contains discontinuities, such as square wave.
Carson's bandwidth rule is,

$$CBR = 2(\Delta f + f_m)$$

where,
Δf = peak frequency deviation.
f_m = highest frequency in the modulating signal.

Conditions for application of carson's rule is only sinusoidal signals,

$$\therefore B_T = 2(\Delta f + W)$$
$$= 2W(D+1)$$

where,
W = highest frequency in the modulating signal but non-sinusoidal in nature

D = deviation ratio of frequency deviation to highest frequency of modulating non-sinusoidal signal.

Carson's Bandwidth rule is often applied to transmitters, antennas, optical resources, receivers, photodetectors. etc.

Q. I) Consider a carrier waveform 10 Cos Wct and a modulating message signal 3 Cos Wmt with fc = 100 KHZ and fm = 4 KHZ. Find, the modulation index and the channel bandwidth for amplitude and frequency modulation. Assume the sensitivity of the frequency modulator to be 5 KHZ per volt.

Sol^n,

⟹ carrier waveform, $V_c = 10 \cos W_c t$

modulating message signal, $V_m = 3 \cos W_m t$

carrier frequency, $f_c = 100$ KHZ

modulating ″ , $f_m = 4$ KHZ

(i) Am signal,

$$\mu = \frac{V_m}{V_c}$$

$$\therefore \mu = \frac{3}{10} = 0.3$$

channel Bandwidth, $BW = 2 \times f_m = 2 \times 4 = 8$ KHZ.

(ii) Fm signal, $m_f = K_f \cdot \frac{V_m}{f_m}$

$$= \frac{5 \times 3}{4}$$

$$= 3.75$$

Bandwidth, $BW = 2\Delta f + 2f_m$

$= 2m_f f_m + 2f_m$

$= 2 \times 3.75 \times 4 + 2 \times 4$

$\therefore$ $BW = 38$ KHZ

II. An Fm transmission has a frequency deviation of 18.75 KHZ. calculate per cent modulation if it is broadcast (i) in 88-108

MHZ band (ii) as a portion of a TV broadcast.

Sol^n,

=> Frequency deviation, $\Delta f = 18.75$ KHZ

A maximum frequency deviation of 75 KHZ is allowed for commercial broadcast,

ie, $(\Delta f)_{max} = 75$ KHZ

$$\%\ \text{modulation},\ m = \frac{\Delta f}{(\Delta f)_{max}} \times 100$$

$$= \frac{18.75}{75} \times 100$$

$$= 25\ \%$$

A maximum frequency deviation of 25 Hz is allowed for sound portion of the TV broadcast,

ie, $(\Delta f)_{max} = 25$ KHZ

So,

$$\%\ \text{modulation},\ m = \frac{18.75}{25} \times 100$$

$$= 75\ \%$$

E. Signal to Noise Ratio (SNR):

It is defined as the ratio of signal power to the noise power, often expressed in decibels. A ratio higher than 1:1 (greater than 0 dB) indicates more signal than noise.

$$SNR = \frac{P_{signal}}{P_{noise}}$$

Where, P = average power.

Both signal and noise power must be measured at the same or equivalent points in a system and within same system bandwidth.

If the variance of the signal and noise are known, and the signal and noise are both zero-mean, SNR can be,

$$SNR = \frac{\sigma^2_{signal}}{\sigma^2_{noise}}$$

If the signal and noise are measured across the same impedance, the SNR can be obtained by calculating the square of the amplitude ratio,

$$SNR = \frac{P_{signal}}{P_{noise}} = \left(\frac{A_{signal}}{A_{noise}}\right)^2$$

where,

A = root mean square (rms)

SNR in decibels:

$$P_{signal} = 10 \log_{10}(P_{signal})$$

and,

$$P_{noise} = 10 \log_{10}(P_{noise})$$

So,

$$SNR_{dB} = 10 \log_{10}(SNR)$$

$$= 10 \log_{10}\left(\frac{P_{signal}}{P_{noise}}\right)$$

Using quotient rule of logarithms,

$$\therefore 10 \log_{10}\left(\frac{P_{signal}}{P_{noise}}\right) = 10 \log_{10}(P_{signal}) - 10 \log_{10}(P_{noise})$$

$$\therefore SNR_{dB} = P_{signal,dB} - P_{noise,dB}$$

Similarly,

$$SNR_{dB} = 10 \log_{10}\left[\left(\frac{A_{signal}}{A_{noise}}\right)^2\right]$$

$$= 20 \log_{10}\left(\frac{A_{signal}}{A_{noise}}\right)$$

$\therefore SNR_{dB} = A_{signal,dB} - A_{noise,dB}$.

Noise Figure :

Noise Figure (NF) and Noise Factor (F) are measures of degradation of the signal to noise ratio (SNR), caused by components in a radio frequency (RF) signal chain. It is a number by which the performance of an amplifier or a radio receiver can be specified, with lower values indicating better performance.

The noise figure is the difference in decibels (dB) between the noise output of the actual receiver to the noise output of an ideal receiver with the same overall gain and bandwidth when the receivers are connected to match the sources at the standard noise temperature T_0.

Noise factor is defined as the ratio of the output noise power of a device to the portion thereof attributable to thermal noise in the input termination at standard noise temperature, (T_0).

ie,

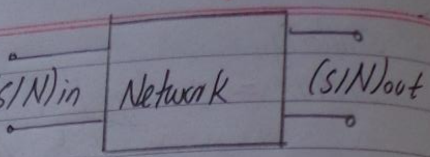
$$\text{Noise factor, } (F) = \frac{SNR_{in}}{SNR_{out}}$$

Where,

SNR_{in} = input signal to noise ratios

SNR_{out} = output " " "

The SNR quantities are power ratios. The noise figure (NF) is defined as the noise factor in dB,



$$\therefore NF = 10 \log_{10} (F)$$

$$= 10 \log_{10} \left(\frac{SNR_{in}}{SNR_{out}} \right)$$

$$= SNR_{in,dB} - SNR_{out,dB}$$

These formulas are only valid when the input termination is at standard noise Temperature, $T_0 = 290$ k,

The noise factor of a device is related to its noise temperature T_e;

$$\therefore F = 1 + \frac{T_e}{T_0}$$

Attenuators have a noise factor F equals to their attenuation ratio L when their physical temperature equals T_0. more generally, for an attenuator at a physical temperature, T, the noise temperature, is $T_e = (L-1)T$ giving a noise factor of

$$\therefore F = 1 + \frac{(L-1)T}{T_0}$$

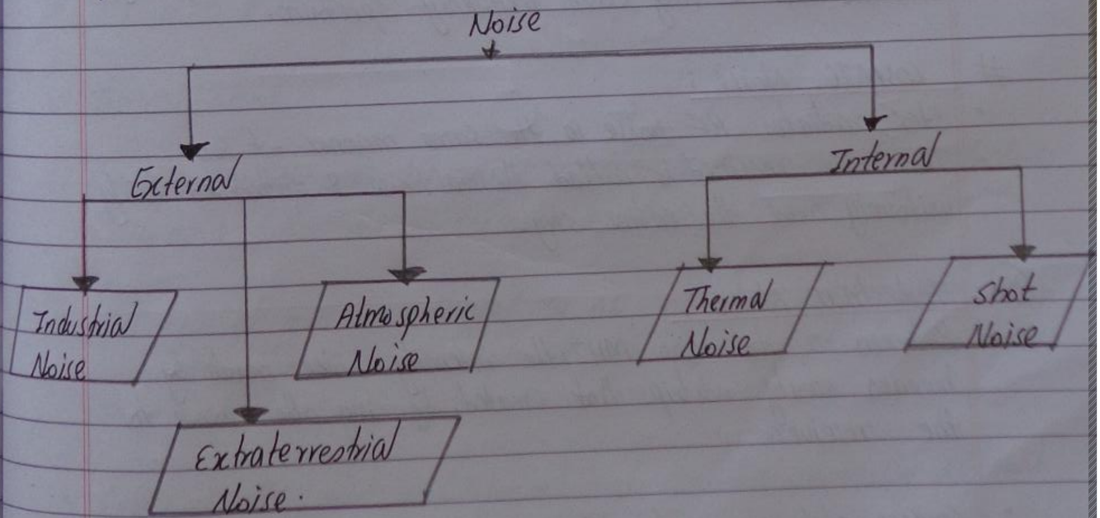
2.8.A) Noise and classifications:

Noise is a random fluctuation in an electrical signal, a characteristic of all electronic circuits. Noise comes in many forms. It can be generated in many ways and noise can affect electronic and radio frequency, RF circuits and systems. In communication system, noise is an error or undesired random disturbance of a useful information signal. The noise is a summation of unwanted or disturbing energy from natural and sometimes man-made sources. Noise, is however, typically distinguished from interference.

For example; in the signal to noise ratio (SNR), signal-to-interference ratio (SIR) and signal to noise plus interference ratio (SNIR) measures.

Noise is also typically distinguished from distortion, which is an unwanted systematic alternation of the signal waveform by the communication equipment; while noise is generally unwanted, it can serve a useful purpose in some applications, such as random number generator or dither.

Practically, we cannot avoid the existence of unwanted signal together with the modulated signal transmitted by transmitter. This unwanted signal is called noise. Noise can limit the range of systems.



(I) Atmospheric Noise:
- caused by lightning dischorges, in thunderstorms and other natral electric disturbances occurring in the atmosphere.
- Consists of spurious radio signal with components distributed over a wide range of frequencies.
- It propagate over the earth in the same way as ordinary radio waves of the same frequencies.
- Became less severe at frequencies above 30 MHZ because,
  - Higher frequencies are limited to line of sight propagation.
  - Natre of mechanism generating this noise is such that very little of it is created in the VHF range & above.

(II) Extra-terrestrial Noise:

SOLAR Noise:
- Normal condition, there is a constant noise radiation from the sun, simply because large body at a very high frequency.
- Radiates over a very broad frequency spectrum.

COSMIC Noise:
- Star radiates RF noise in the same manner of sun.
- The noise received is called thermal noise & distributed fairly uniformly over the entire sky.

(III) Industrial Noise:
- Between 1 to 600 MHZ, the intensity noise made by humans easily outstrips that created by any other source to the receiver.

- Sources such as: automobile, aircraft, electric motors, and other heavy machines.

- Nature of industrial noise is so variable that it is difficult to analyze.

IV. Shot Noise :

- caused by the random arrival of carriers at the output element of an electronic device.
- sometimes called transistor noise.
- Randomly varying and superimposed onto any signal present.
- The circuit current carriers are not moving continuously steady flow.
- First observed in the anode current of a vacuum-tube amplifier.

- Shot noise also occurs in semiconductors due to liberation of charges carrier.
- For pn junction, the mean square shot noise current is,

$$I_n^2 = 2(I_{dc} + 2I_o) q_o B \ (amp)^2$$

V. Thermal Noise :

- Is associated with the rapid and random movement of electrons within a conductor due to thermal agitation.

- This type of noise is generated by all resistances (eg: resistor, semiconductor, resistance of a resonant ckt. ie, real part of impedance, cable etc)

- Experimentally result by (Johnson) and theoretical studies (by Nyquist) give the mean square noise voltage as,

$V_n^2 = 4RKTB$

where,

K = Boltzmann's constant = 1.38×10^{-23} J/K

T = absolute temperature

B = bandwidth noise

R = resistance

- Referred as white noise.
- Is a form of additive noise, cannot be eliminated. It increases in intensity with the number of devices in a circuit.
- Thermal noise power is proportional to the product of bandwidth and temperature, mathematically,

  Noise power, $N = KTB$

- Noise power can be modeled using voltage equivalent circuit (Thevenin equ. ckt) or Norton equ ckt.

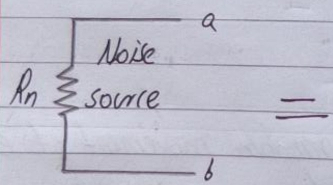


Fig: Noise source ckt

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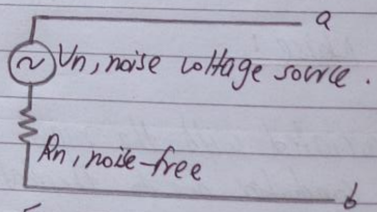


Fig: Thevenin equ ckt.

vi. Flicker Noise:

Flicker noise, also known as 1/f noise, is a signal or process with a frequency spectrum that falls off steadily into the higher frequencies, with a pink spectrum. It occurs in almost all electronic devices, and results from a variety of effects, though always related to the direct current.

vii) **Burst Noise : / popcorn Noise :**

Burst noise contains of sudden step-like transitions between two or more levels (non-Gaussian) as high as several hundred microvolts, at random and unpredictable times. Each shift in offset voltage or current lasts for several milliseconds and the interval between pulses tend to be in the audio range (less than 100 Hz), leading to the term popcorn noise for the popping or crackling sounds it produces in audio circuits.

2.9) **Pulse Modulation :**

The transmission of analog data or speech which is continuous in form is known as pulse modulation. The process of transmitting signals in the form of pulse (discontinuous signals) by using special techniques.

Pulse-modulation

- Analog PM
  - pulse amplitude (PAM)
  - Pulse width (PWM)
  - pulse position (PPM)
- Digital PM
  - Delta (DM)
  - pulse code (PCM)

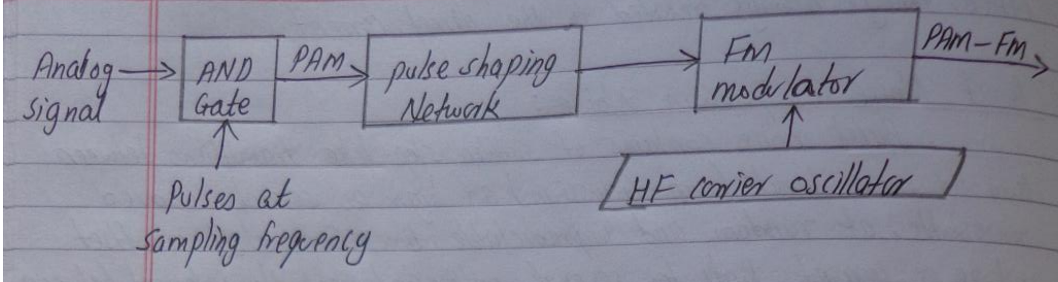
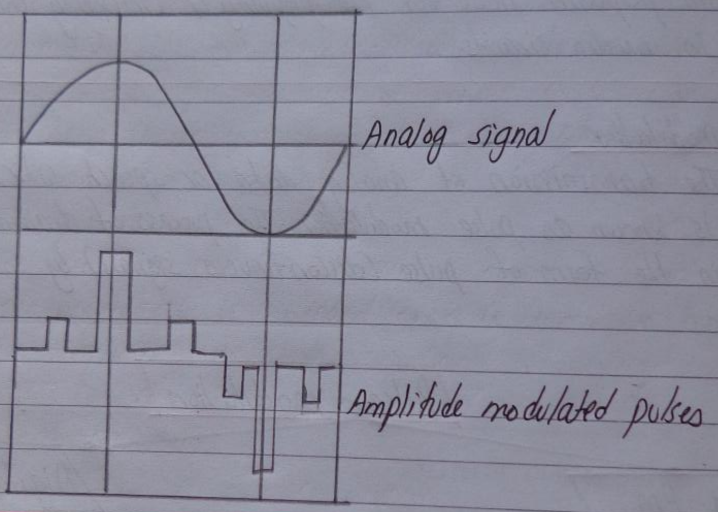


Fig: pulse amplitude modulator.



2.9.B Demodulation / Detection:

Demodulation or detection is a process of recovering the original modulating signal (intelligence) from the modulated carrier wave, i.e, the demodulation is a reverse process of the process of modulation.

Necessity of Demodulation:

The wireless signals transmitted from a transmitter consists of RF carrier waves and audio frequency signal waves. If the modulated wave is directly fed to the loudspeaker, no sound will be heard from the loudspeaker. This is because of the simple reason that the frequency of the carrier wave is very high and the loudspeaker diaphgram cannot respond to such high frequency due to large intertia of their vibrating discs etc. Neither will such RF waves produce any effect on human ear as their frequency are much beyond the audible frequency. Hence, it becomes essential to separate the audio frequency signal from the modulated carrier wave.

Am Detection:

The process of demodulation demands that the modulated wave has some definite average value and the carrier wave is separated out. Hence, in demodulation of an AM wave, two operations viz, rectification of the modulated wave & elimination of the RF component of the modulated wave are involved.

Am Demodulators or Detectors:

The device used for demodulation or detection are called demodulators or detectors. For amplitude modulation (Am) detectors used are:

(i) square law detectors
(ii) envelope detectors.

AM signal with large carrier are detected by using the envelope detector. The envelope detector uses the circuit which extracts the envelope of the AM wave. In fact, the envelope of the AM wave is the baseband or modulating signal. But a square law detectors are used to detect low level signal in which a device operating in the non-linear region.

FM Detection:

For the FM detection, the method usually employed involves the conversion of FM into AM and then applications of conventional method of detection. Thus, demodulation of an FM wave involves three operations which are given to be below:

i) Rectification of modulating signal
ii) Conversion of frequency variations produced by modulating signals into corresponding amplitude variations.
iii) Elimination of RF component of the modulated wave.

For the FM detection, we need a circuit in which magnitude of output voltage varies in accordance with the instantaneous frequency variations in the input voltage. Such circuits are called Discriminators. There are two types of FM detectors, viz

(I) Simple slope detector
(II) Balanced slope detector.

Foster-seeley detector is a phase difference detector and is widely used. FM transmitting and receiving equipments particularly used for modulation and demodulation tend to be more complex and hence more costly. Since, the amplitude of FM signal remains unchanged, the power of FM signal will be same as that of the unmodulated carrier.

Chapter -3 Digital Communications

10 Hours

51

1) A. Digital Communication:

Data transmission, digital transmission or digital communication is the transfer of data (a digital bit stream or a digitized analog signal) over a point-to-point or point to multipoint communication channel. Data communication traditionally belongs to telecommunications and electrical Engineering. Transmitting analog signals digitally allows for greater signal processing capability. The data are represented as an electromagnetic signal such as an electrical voltage, radiowave, microwave or infrared signal.

B) Differences:

	Digital Communications	Analog Communications.
i)	Noise immunity	more noise.
ii)	Greater Bandwidth	lesser Bandwidth
iii	synchronization problem is relatively difficult.	— relatively easier.
iv	error correction by coding	no error correction capability.
v.	can merge different data (voice, audio, video, data) and transmit over a common digital transmission system.	cannot merge data from different sources.
Vi	privacy preserved (data encryption)	no privacy.
Vii	High cost & not portable.	low cost and portable.
Viii	— Flexible in implementation.	Analog Hardware is not flexible.

ix.	Long distance	short distance.

C. Block diagram and Elements of Digital communication : 2017/F 2015/F

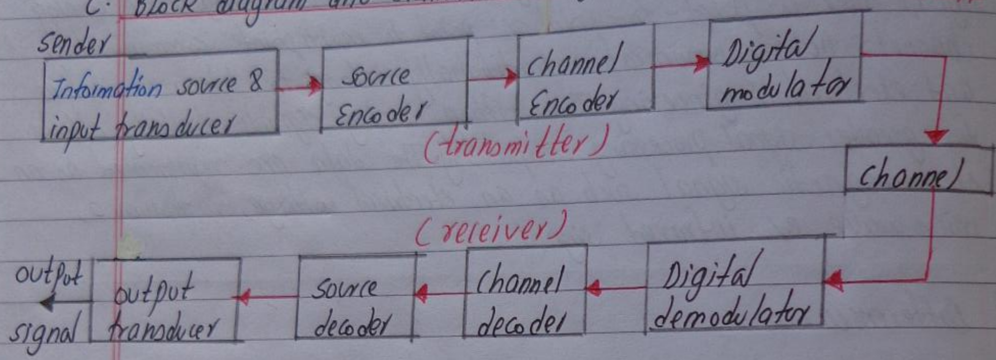


Fig: Block diagram & elements of digital communication.

① Source of Information:

- Analog information sources
- Digital " "

→ Analog information source :

microphone actuated by a speech, TV camera scanning a scene, Continuous amplitude signals

→ Digital information sources:

This are teletype or the numerical output of computer which consists of a sequence of discrete symbols.

An analog information is transformed into a discrete information through the process of sampling & quantizing.

(ii) Source Encoder / Decoder :

The source encoder or source coder converts the input ie symbol sequence into a binary sequence of 0's and and 1's by assigning code words into symbol in the input sequence.

At the receiver, the source decoder converts the binary output of the channel decoder into a symbol sequence. The decoder for a system using fixed-length code words is quite simple, but the decoder for a system using variable-length code words will be very complex.

Aim of the source coding is to remove the redundancy in the transmitting information, so that bandwidth required for transmission is minimized. Higher the probability, shorter is the codeword.
Eg: Huffman coding.

(iii) Channel Encoder / Decoder :

Error control is accomplished by the channel coding operation that exists of systematically adding extra bits to the output of the source coder. These extra bits donot convey any information but helps the receiver to detect and/or correct some of the errors in the information bearing bits.
There are two methods of channel coding:
- Block coding
- Convolution coding

The channel decoder recovers the information bearing bits from the coded binary streams. Error detection and possible correction is also performed by the channel decoder.

iv Modulator:

It converts the input bit stream into an electrical waveform suitable for transmission over the communication channel. It can be effectively used to minimize the effects of channel noise, to match the frequency spectrum of transmitted signal with channel characteristics, to provide the capability to multiplex many signals.

V. Demodulator:

The extraction of the messages from the information beering waveform produced by the modulation is accomplished by the demodulator. The output of the demodulator is bit stream.

vi CHANNEL:

The channel provides the electrical connection between the source and destination. The different channels are pair of wires, optical fibre, Radio channel, satellite channel or combination of any of these.

The communication channel have only finite bandwidth, non ideal frequency response, the signal often suffers amplitude and phase distortion as it travels over the channel. Also, the signal power decreases due to the attenuation of the channel. The important parameter of the channel are signal to noise power ratio (SNR), useable bandwidth, amplitude and phase response and the statistical properties of noise.

Vii MuX:

multiplexer is used for combining signals from different signals so that they share a portion of the communication system.

viii DE-MUX:
Demultiplexer is used for separating the different signals so that they reach their respective destinations.

D) Advantages of Digital communication system over Analog: 2015/F

i) In digital communication system, the error correction & detection technique can be implemented easily.

ii) Digital communication is cheaper than Analog.

iii) Digital " system has more immunity to noise and external interference.

iv) Digital information can be saved and retrived when necessary while it is not possible in analog.

V. Security is another priority of messaging the services in the modern days. The digital communication provides better security to messages than the analog communication.

Vi Bandwidth is another score resources. Various digital communication techniques are available that use the available bandwidth much efficiently than analog.

Vii Digital signal amplifier regenerate an exact signal, eliminating cumulative errors. An incoming (analog) signal is sampled, its value is determined and the node then generates a new signal from the bit value; the incoming signal is discarded. With analog signal circuits, intermediate nodes amplify the incoming signal, noise and all. Analog circuit requires amplifier and each amplifier adds distortion and noise to the signal.

VIII. The configuring process of digital communication system is simple as compared to analog communication system, although, they are complex.

IX. Digital transmission provides higher maximum transmission rates via medium such as optical fibres.

X. supports data integrity. simple to integrate voice, video, & data Digital transmission provides easier way to integrate different digital formats.

xi Digital data can be compressed & therefore possible to pass over higher bandwidths.

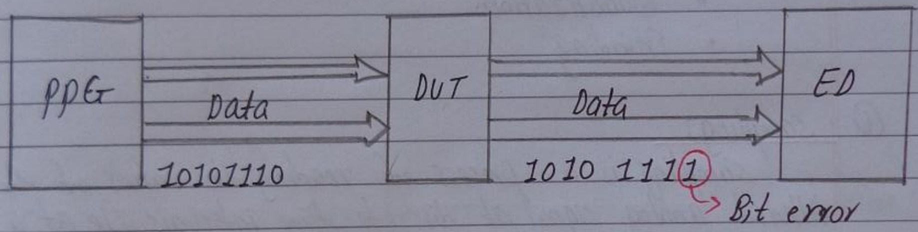
E. Bit error rate :

In digital transmission, the number of bit errors is the number of received bits of a data stream over a communication channel that have been altered due to noise, interference, distortion or bit synchronization errors. The bit error rate (BER) is the number of bit errors per unit time.

The bit error ratio also (BER) is the number of transferred bits during a studied time interval. BER is a unitless performance measure, often expressed as percentage.

The bit error probability, Pe is the expectation value of the bit error ratio. The bit error ratio can be considered as an appropriate estimate of the bit error probability. This estimates is accurate for a long time interval and a high number of bit errors.

$$BER = \frac{\text{total number of error bits}}{\text{" " " bits}}$$



The BER may be evaluated using monte carlo (stochastic) computer simulations.

Factors affecting bit error rate, BER

- Interference
- Reduce bandwidth
- Lower order modulation
- Increase transmitter power
- Noise

Bit error rate is an empirical record of a system's actual bit error performance.

2.A) Generation of PCM signal :

Pulse-code-modulation (PCM) is a method of representing digitally sampled analog signals. Analog transmission is not particularly efficient. PCM is simplest form of waveform coding. The digital signal is subsequently used to reconstruct the analog signal. PCM is the extension of pulse-amplitude-modulation (PAM). A PCM performs three functions,

- Sampling
- Quantization.
- Encoding

① Sampling:

Sampling is a process of reading the values of the filtered analog signal at discrete time intervals (ie, at a constant sampling rate, called the sampling frequency).

- Analog signal is sampled every T_s sec
- T_s is referred to as the sampling interval.
- $f_s = \frac{1}{T_s}$ is called sampling rate / sampling frequency.

- There are three sampling methods:

a) Ideal:

an impulse at each sampling instant.

b) Natural:

a pulse of short width with varying amplitude.

c) Flattop:

sample and hold, like natral but with single amplitude value.

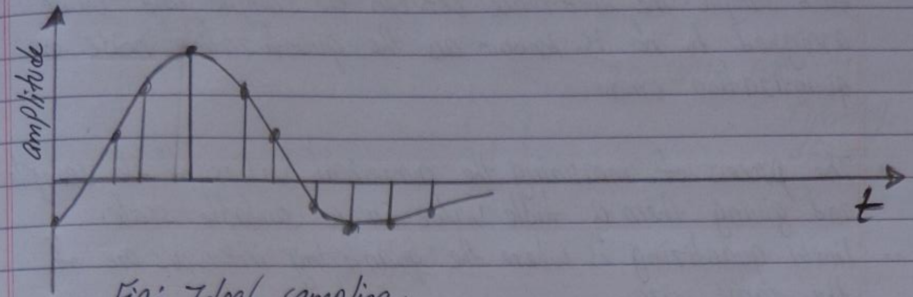


Fig: Ideal sampling.

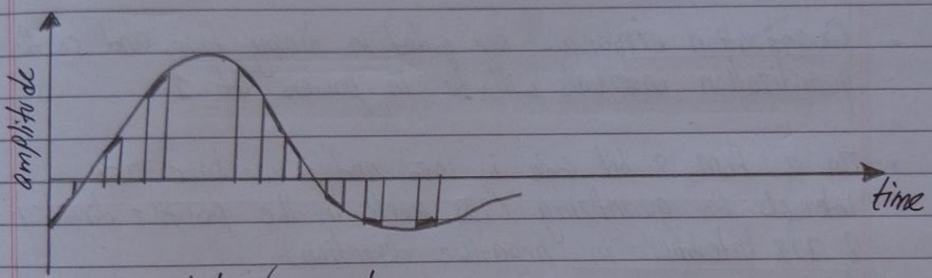


Fig: Natral sampling

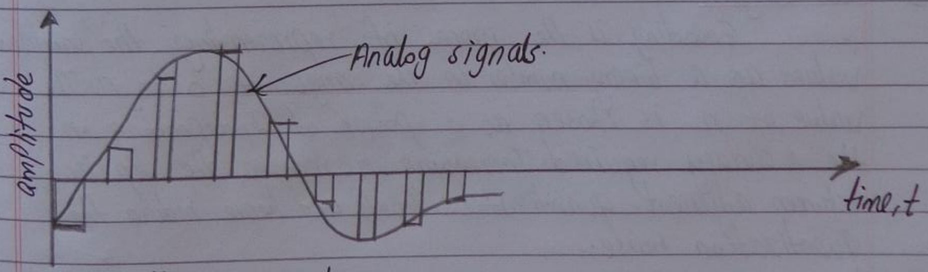


Fig: flattop sampling.

(ii) Quantizing:

It is the process of assigning a discrete value from a range of possible values to each sample obtained. The number of possible values will depend on the number of bits used to represent each sample. The difference between the sample and value assigned to it is known as the quantization noise or quantization error.

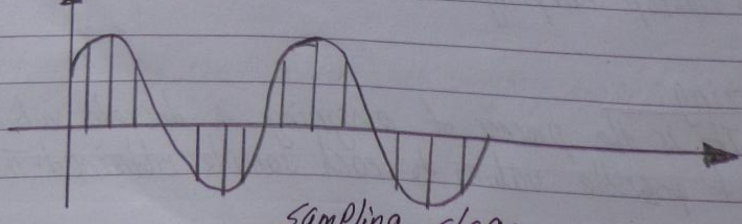
- The process of measuring the numerical values of the samples and giving them a table value in a suitable scale.
- Linear quantizing is where the quantizing intervals are of the same size.

- Quantization intervals are coded in binary form, and so the quantization intervals will be in powers of 2.

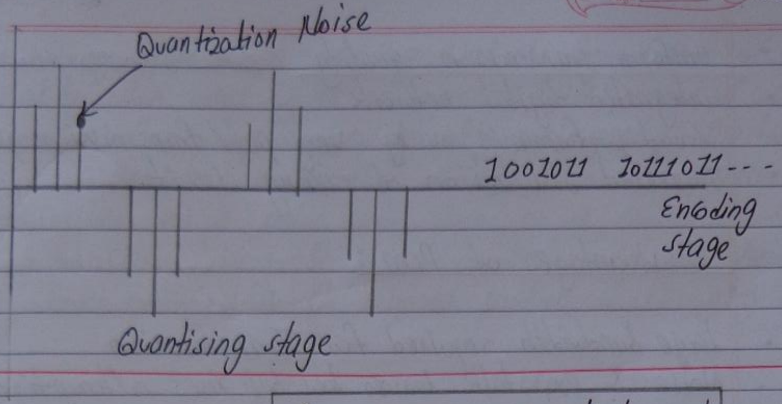
- In a PCM, 8 bit code is used and so we have 256 intervals for quantizing (128 levels in the positive direction & 128 levels in negative direction).

(iii) Encoding:

Encoding is the process of representing the sampled values as a binary number in the range of 0 to n. The value of n is chosen as a power of 2, depending on the accuracy required. Increasing n reduces the step size between adjacent quantizations levels and hence reduce the quantization noise.



Sampling stage



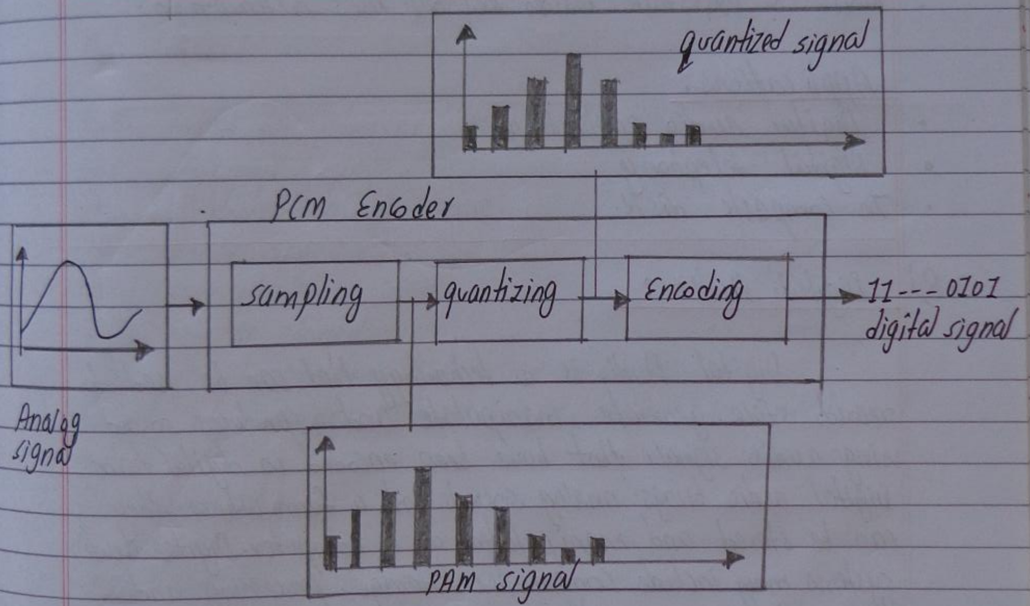


Fig: Block diagram of PCM.

Advantages of PCM:

- Low manufacturing cost.

- uniform transmission quality
- integrated digital network
- Good performance over very poor transmission paths.
- Increased utilization of existing circuit

Disadvantages of PCM:

- Large bandwidth required for Transmission
- Noise & crosstalk leaves low but rises attenuation.

Applications:
- Digital audio applications
- Digital telephony
- In compack disk.

2·B) Digital Audio:

Digital Audio is a technology that can be used to record, store, generate, manipulate and reproduces sound using audio signals that have been encoded in digital form. Digital audio brings analog sounds into a form where they can be stored and manipulated on a computer. Digital audio systems may include compression, storage, processing and transmission components.

Conversion to a digital format allows convenient manipulation, storage, transmission components. Conversion to a digital format allows convenient manipulation, transmission and retrieval of an audio signal. unlike analog signal audio, in which making copies of a recording leads to degradation of the signal quality, when using digital audio, an infinite number of copies can be made without any degradation of signal quality. Digital audio is a mathematical description of the pattern of pressure. It is transmitted as a pulse wave, stored as a series on/off switches (transistors), magnetic pulses, or optical pits and lands etc and looks nothing like the original pattern of changing air pressure.

Advantages:

- Better sound for cheaper equipment.
- wider dynamic range.
- very low distortion & noise.
- Non-linear operation.
- Highly portable.
- Data can be cloned.
- Data can be transmitted over networks.
- variety of recorder options (tape, SSD, RAM etc)

Disadvantages:

- Data can become corrupted.
- Computer crash.
- Software compatibility issues, OS, Chosts, drivers, plugins)
- Sound quality can be too clinical /cold.
- Can Have poor multi-user interfaces.
- Confusing array of formats and systems.

2.c) Shannon Capacity Theorem: 2014/S, 2015/S, 2016/F

Shannon capacity theorem tells the maximum rate at which information can be transmitted over a communications channel of a specified bandwidth in the presence of noise. It is an application of the noisy channel coding theorem to the archetypal case of a continuous time analog communications channel subject to Gaussian noise.

The shannon capacity theorem states that the channel capacity C, meaning theoretical tightest upper bound on the information rate of data that can be communicated at an arbitraily low error rate using an average received signal power S through an analog communication channel subject to additive white gaussian noise of power N.

$$C = B \log_2\left(1 + \frac{S}{N}\right) \text{ bits/sec}$$

Where,

B = Bandwidth of the channel in Hz

S = average received signal power over the bandwidth.

N = average power of the noise & interference over the Bandwidth.

$\frac{S}{N}$ = signal to noise ratio (SNR) or carrier to noise ratio (CNR) of the communication signal to noise.

C = channel capacity in bits/second

capacity increases (almost) linearly with B, whereas S determines only a logarithmic increase.

3. A) Bandwidth Efficiency:

Spectral efficiency, spectrum efficiency or bandwidth efficiency refers to the information rate that can be transmitted over a given bandwidth in a specific communication systems. It is a measure of how efficiently a limited frequency spectrum is utilized by the physical layer protocols and sometimes by the media access control.

Bandwidth efficiency is defined as the number of bits that can be transmitted within 1 Hz of bandwidth,

$$\therefore \quad \eta = \frac{R_b}{B_T} \quad \text{bits/sec/Hz}$$

$$\therefore \text{B.W efficiency} = \frac{\text{transmission rate (bps)}}{\text{minimum Bandwidth (Hz)}}$$

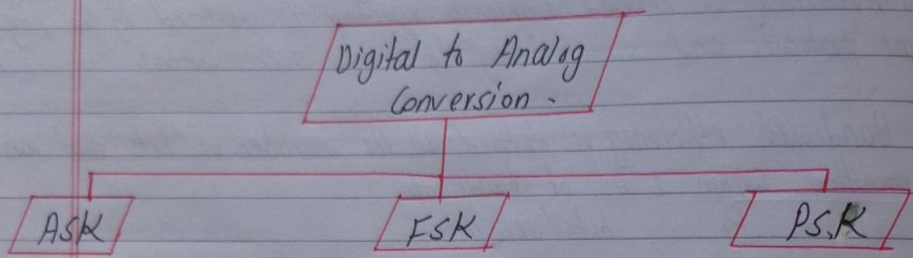
$$= \frac{\text{bits/second}}{\text{Hertz}}$$

$$= \frac{\text{bits}}{\text{cycle}}$$

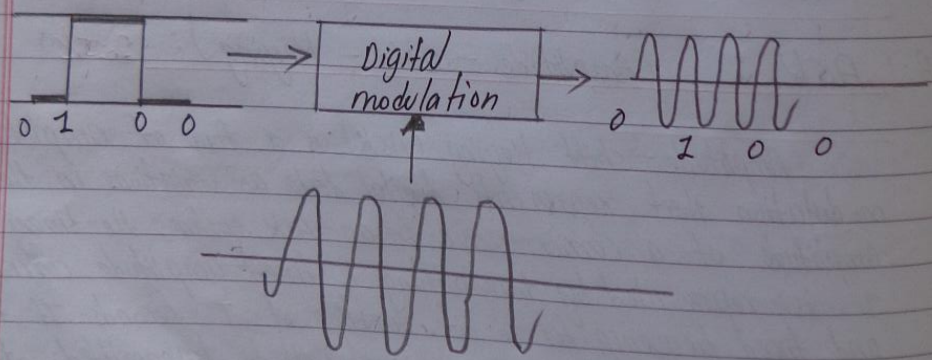
B. ASK : (Amplitude - shift - Keying): 2015/S

Amplitude-shift keying (ASK) is a form of amplitude modulation that represents digital data as variations in the amplitude of a carrier wave. In an ASK system, the binary symbol 1 is represented by transmitting a fixed-amplitude carrier wave and fixed frequency for a bit duration of T seconds. If the signal values is 1, then, the carrier signal will be transmitted, otherwise, a signal value of 0 will be transmitted.

Like AM, ASK is also linear and sensitive to atmospheric noise, distortions, propagation condition on different routes in PSTN etc. Both ASK modulation and demodulation process are relatively inexpensive. The ASK technique is also commonly used to transmit digital data over optical fiber.



In this technique, amplitude of the RF carrier is varied in accordance with baseband digital input signal. The figure depicts operation of ASK modulation. As shown in figure, binary 1 will be represented by carrier signal with some amplitude while binary 0 will be represented by carrier of zero amplitude (ie, no carrier)



ASK modulation can be represented by,
$s(t) = A \cos(2\pi f_c t)$ for Binary 1.

$s(t) = 0$ for Binary 0

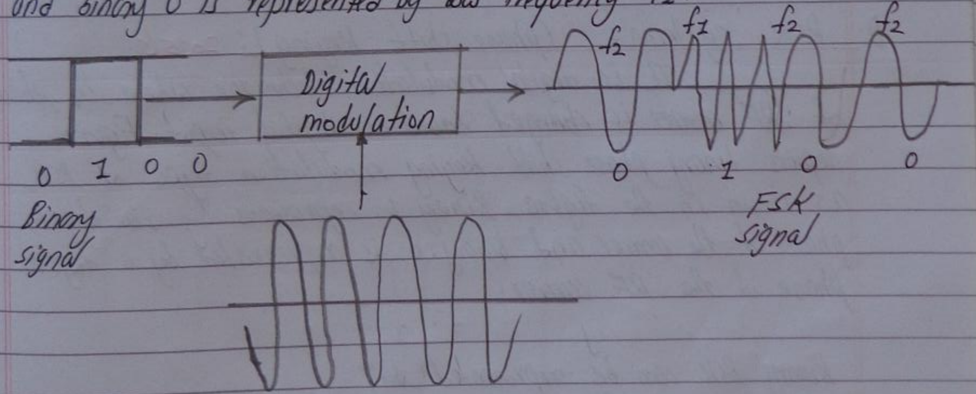
Bandwidth requirement for ASK is,

$$BW = \frac{2}{T_b} = 2 \times R_b.$$

- In ASK, probability of error (Pe) is high and SNR is less.
- It has lowest noise immunity against noise.
- ASK is a bandwidth efficient system but it has lower power efficiency.

c. FSK : (Frequency shift Keying) : 2015/S

It is also known as digital modulation technique. In this technique, frequency of the AF carrier is varied in accordance with baseband digital input. The figure depicts the FSK modulations. As shown, binary 1 and 0 represented by two different carrier frequency. Figure depicts that binary 1 is represented by high frequency f_1 and binary 0 is represented by low frequency f_2.



Binary FSK can be represented by,

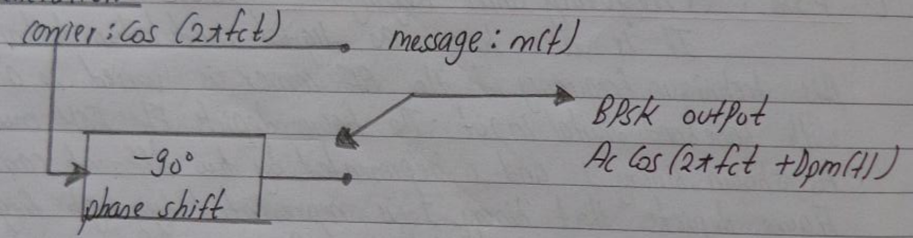
$s(t) = A \cos(2\pi f t)$ for Binary 1.

$s(t) = A \cos(2\pi f t)$ for binary 0

Bandwidth (BW) $= 2 \times R_b + (f_1 - f_2)$

- In case of FSK, P_e is less and SNR is high.
- This technique is widely employed in modem design and development.
- It has increased immunity to noise but requires larger bandwidth compare to other modulation types.

Generation:



D. PSK systems: (phase shift Keying): 2015/s

It is digital modulation technique where in phase of RF carrier is changed based on digital input. Figure depicts binary phase shift keying modulation type of PSK, as shown in the figure, Binary 1 represents by 180 degree phase of the carrier and binary 0 is represented by 0 degree phase of the RF carrier.

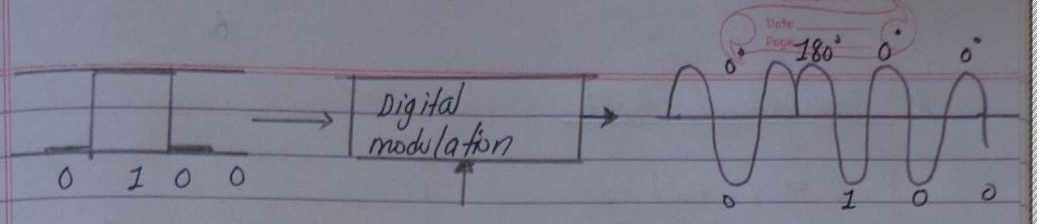
Binary PSK can be represented by,

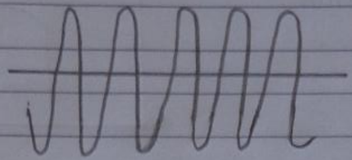
$s(t) = A \cos(2\pi f_c t)$ for binary 1

$s(t) = A \cos(2\pi f_c t + \pi)$ for binary 0

$\therefore$ Bandwidth $= 2 \times R_b = 2 \times$ bit rate

69

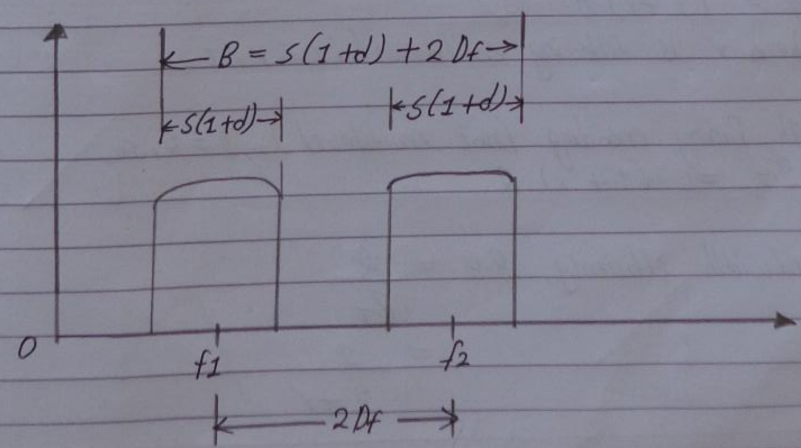




- PSK modulation is widely used in wireless transmission.
- The variants of basic PSK and ASK modulations are QAM, 64-QAM, 16-QAM and so on.
- It is a power efficient system but it has lower Bandwidth efficiency.
- Probability of error is less and SNR is high.

4.A) Bandwidth of FSK:

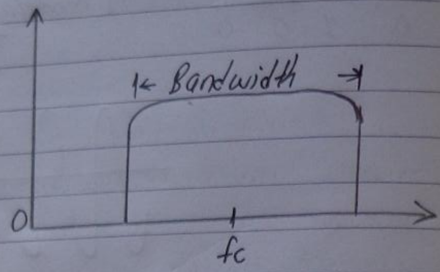
$$\text{Bandwidth} = (1+d) \times S + 2\Delta f$$



B) Bandwidth of PSK:

$B = (1+d) \times S$

- Less than that for BFSK



C. Bandwidth of QAM:

- quadrature amplitude modulation
- Combination of ASK & PSK
- Large bandwidth savings.

D. Bandwidth of BPSK:

- B_T directly related to the (signaling, modulation, baud) rate, D.

=> $B_T = (1+r) D$.
where, r is filtering coefficient; $0 < r < 1$.

=> with Binary encoding (not multilevel), $D = R$, so,
$B_T = (1+r) R$

Bandwidth efficiency, $B.E = \frac{R}{B_T}$
$= \frac{1}{1+r}$

Board Exam solutions of chapter 1, 2, 3

1) A. What is DSB-SC modulation? Justify with necessary spectrums "DSB-SC" is wasteful for transmission bandwidth than SSB-SC". But, why SSB not used for broadcasting. Give reasons. 2015/F, 2014/S

(8)

⇒ Double-side band suppressed-carrier transmission (DSB-SC) is transmission in which frequencies produced by amplitude modulation (Am). are symmetrically spaced above and below the carrier frequency and the carrier level is reduced to the lowest practical level, ideally being completely suppressed. It is used for radio data system.

In DSBSC system, carrier component is altogether removed resulting in saving of enormous amount of power. DSB-SC is generated by a mixer.

$s(t) = c(t) \cdot m(t)$.

$= A_c \cdot \cos(2\pi f_c t) \cdot m(t)$.

By taking fourier transform on both sides.

$\therefore \; S(f) = \frac{A_c}{2} [M(f-f_c) + M(f+f_c)]$

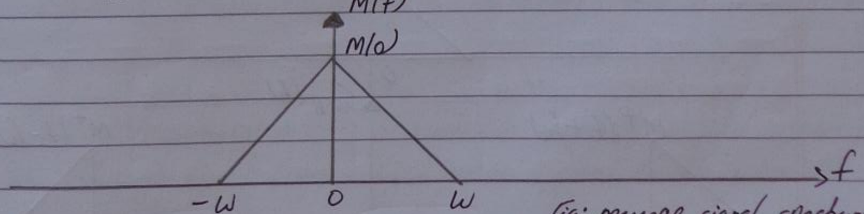


Fig: message signal spectrum

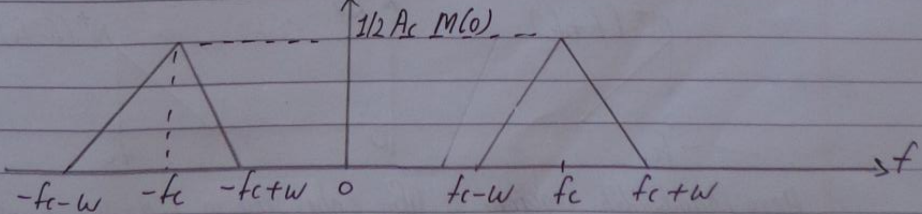
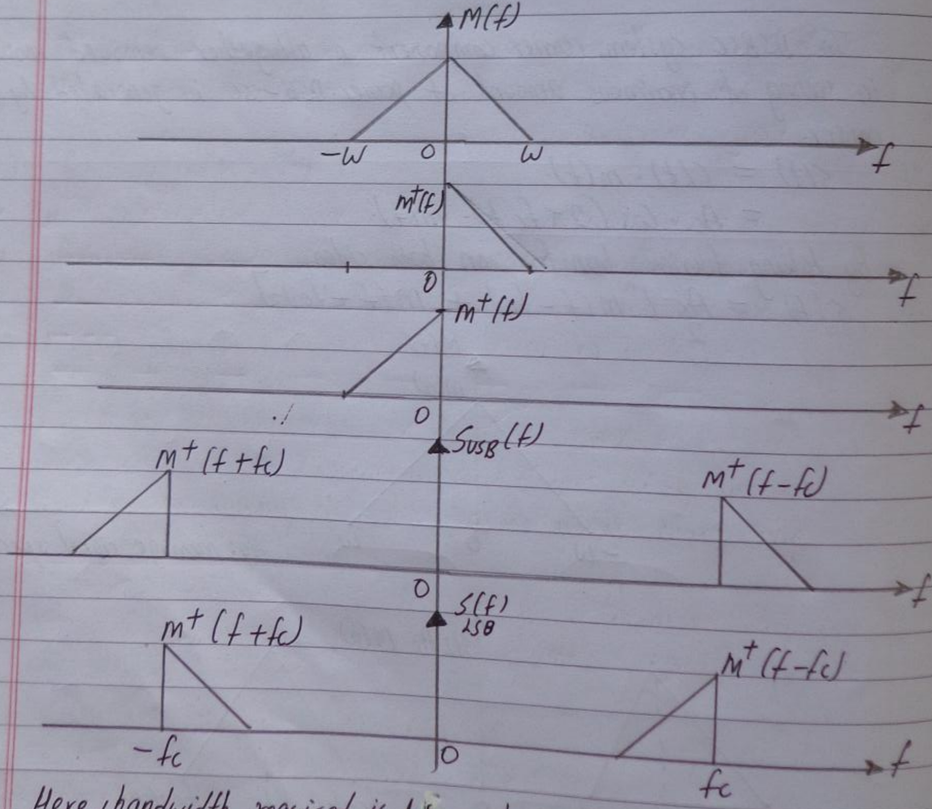


Fig: DSB-SC signal spectrum

The transmision bandwidth required by DSB-SC modulation is the same as that for standard amplitude modulation ie, 2W.

SSBSC system, consists in transmitting only one sideband and suppress the other sideband and the carrier. It utilizes the fact that the intelligence or message is contained in each sideband & not in the carrier. Thus, all information is available in one sideband only and one sideband along the carrier can be discarded with no loss of intelligence.



Here, bandwidth required is W only. Since, in DSB-SC, $\frac{2}{3}$ of the total transmitted power taken up by the carrier. DSB has a wide bandwidth, ie, info of USB = info of LSB. But, in SSB, half as much as bandwidth required in DSBSC.

ie, $BW_{SSB} = \frac{1}{2} \cdot BW_{DSBSC}$

Hence, DSB-SC is wasteful for transmission.

Reason for not using SSB for Broadcasting:

(i) The generation and reception of SSB signal is complicated & therefore, limited to radio telephony.

(ii) The SSB transmitter and receiver need to have an excellent frequency stability. A slight change in frequency will hamper the quality of transmitted and received signal. Hence, it is used for speech transmission.

However, DSBSC transmission system is widely used in broadcasting because of its relative simplicity of its modulating equipment.

B) Define baseband transmission? Why is modulation used in communication system? 2017/F

5

⟹ Baseband transmission is a transmission in which digital signals are sent through direct current (DC) pulses applied to the wire. It supports half duplexing.

Example: Ethernet is an example of a Baseband system found on many LANS.

Modulation Necessity in Telecommunication are:

Copy from page No 9, 10, 11.

(8) c) Perform the performance comparision between Fm & Am system.
2017/F

⟹

	FM	AM
①	Fm can be received as digital.	Am can be received as analog.
(ii)	The amplitude of carrier remains constant.	The amplitude of carrier changes with modulation.
(iii)	The frequenty changes with the modulation.	The carrier frequenty remains constant with modulation.
(iv)	The value of modulation index (M) can be more than 1.	The value of modulation factor cannot be more than 1 for distortionless Am signal.
V.	FM receivers are immune to noise.	Am receivers arenot immune to noise.
Vi	All transmitted power is useful.	Carrier power and one sideband power is useless.
Vii.	FM broadcasts operate in the upper VHF and UHF frequenty ranges at which there happens to be less noise.	Am broadcast operate in the MF and HF ranges where there exists more noise.
Viii	Fm is taken in high frequency & its stereo.	Am is taken in low frequenty and not stereo.
ix.	The amplitude & phase remains constant.	The frequency & phase remain constant.

x.	Frequency range : in a higher spectrum from 88 to 108 MHZ or 1200 to 2400 bits/sec.	AM ranges from 535 to 1705 KHZ or upto 1200 bits/sec
xi	Bandwidth required is twice the sum of the modulating signal frequency and the frequency deviation.	Bandwidth required is twice the highest modulating frequency.
xii	Zero crossing in modulated signal is not equidistant.	Equidistant.
xIII.	Good for local radio because of the smaller range.	Not ideal because the range is too big.

D) Define amplitude modulation. Derive the expression for AM in frequency domain and show that transmission bandwidth for an AM wave is exactly twice the message signal bandwidth. 2016/F

8

⇒ The process of varying amplitude of the high frequency or carrier wave in accordance with the intelligence (code, voice, music) to be transmitted, keeping the frequency and phase of the carrier wave unchanged, is known as amplitude modulation (AM).

Consider a sinusoidal carrier wave, $c(t)$,

$c(t) = A_c \cos(2\pi f_c t)$ — ①

where, A_c = peak value of carrier amplitude

f_c = carrier frequency

The standard form of amplitude modulation (AM) wave is,

$$s(t) = A_c[1 + k_a \cdot m(t)] \cos(2\pi f_c t) \quad \text{—— (11)}$$

where, k_a = constant called amplitude sensitivity of the modulator.

Now,

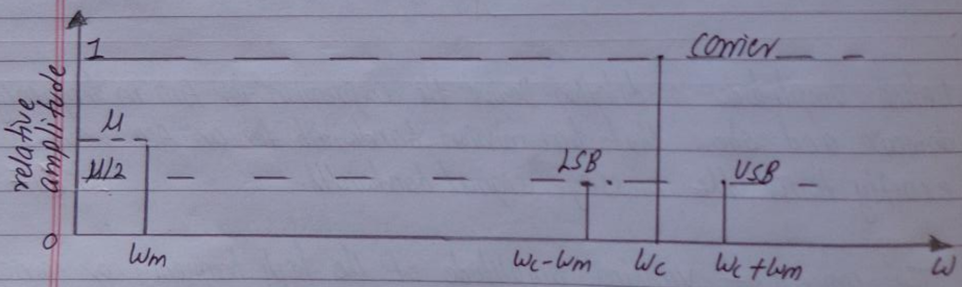
frequency domain / spectral description of Am:

or $s(t) = A_c \cos(2\pi f_c t) + A_c k_a m(t) \cos(2\pi f_c t)$

$= A_c \cos(2\pi f_c t) + \frac{A_c k_a m(t)}{2} e^{j2\pi f_c t} + \frac{A_c k_a m(t)}{2} \cdot e^{-j2\pi f_c t}$

For frequency domain, taking fourier transform,

$$\therefore S(f) = \frac{A_c}{2}[\delta(f - f_c) + \delta(f + f_c)] + \frac{A_c k_a}{2}[M(f - f_c) + M(f + f_c)]$$



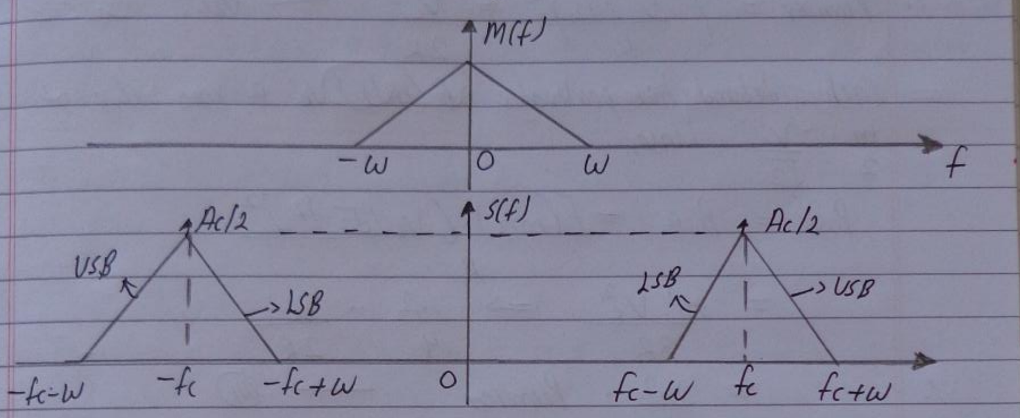
Bandwidth for an Am:

The bandwidth of an Am signal is equal to twice the bandwidth of the modulating signal & covers a range centered on the carrier frequency.

$BW_t = 2 \times BW_m$.

For positive frequency, the highest component of the AM wave is

$f_c + W$ and lowest frequency component is $f_c - W$. The difference between these two frequencies define the transmission Bandwidth 'B' for an AM wave, which is exactly twice the message bandwidth.



Hence,

$$\text{Bandwidth} = \underline{f_c + W - (f_c - W)}$$

$$\therefore \quad B.W = 2W$$

So, Bandwidth for an AM is exactly twice the message signal Bandwidth.

$\because$ message signal Bandwidth $= W - 0 = W$

⑧ E. What are the Bandwidth and power of an AM wave? Derive the expression for efficiency of AM signal. 2015 /s

⇒ Bandwidth is the difference between the two extremes frequencies.

Bandwidth, $BW = (\omega_c + \omega_m) - (\omega_c - \omega_m) = 2\omega_m$

In practice, the AM wave $s(t)$ is a voltage or current wave. The modulated wave has more power than that had by carrier wave before modulation.

$$P_{total} = P_{carrier} + P_{USB} + P_{LSB} \quad \text{——} \; (1)$$

When an amplitude modulated wave is impressed upon resistance (say antenna resistance) R, then,

$$\therefore P_{carrier} = \left(\frac{V_c/\sqrt{2}}{R}\right)^2 = \frac{V_c^2}{2R} \quad \text{—— (II)}$$

Each sideband has peakvalue of $(m/2) V_c$ & rms value of $\frac{m}{2} \cdot \frac{V_c}{\sqrt{2}}$, hence,

$$P_{USB} = P_{LSB} = \frac{[(m/2)(V_c/\sqrt{2})]^2}{R}$$

$$= \frac{m^2 V_c^2}{8R} \Rightarrow \frac{m^2}{4} \cdot \frac{V_c^2}{2R}$$

$$\therefore = \frac{m^2}{4} \cdot P_{carrier} \quad \text{—— (III)}$$

Hence,

$$P_{total} = \frac{V_c^2}{2R} + \frac{m^2}{4} \cdot \frac{V_c^2}{2R} + \frac{m^2}{4} \cdot \frac{V_c^2}{2R}$$

$$= \frac{V_c^2}{2R}\left(1 + \frac{m^2}{2}\right)$$

$$= P_{carrier}\left(1 + \frac{m^2}{2}\right)$$

Maximum power in the AM wave will occur when $m = 1$, ie,

$$\therefore P_{total} = P_{carrier} \times \left(1 + \frac{1}{2}\right) = 1.5\, P_{carrier}$$

Also,

$$\frac{P_{SB}}{P_{total}} = \frac{\frac{m^2}{4} \cdot P_{carrier} + \frac{m^2}{4} P_{carrier}}{P_{carrier}\left[1 + \frac{m^2}{2}\right]} = \frac{\frac{m^2}{2}}{1 + \frac{m^2}{2}}$$

For $m = 1$,

$\therefore \frac{P_{SB}}{P_{total}} = \frac{1}{3}$

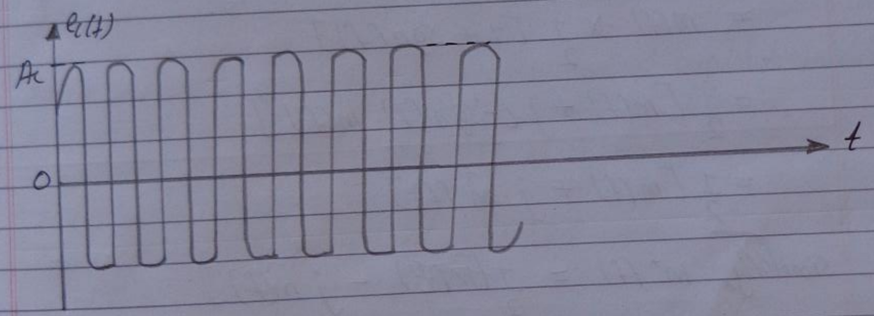
ie, only 1/3 of the total power of the modulated wave is contained in the two side bands & rest of the 2/3 power lies in the carrier component which is of no use.

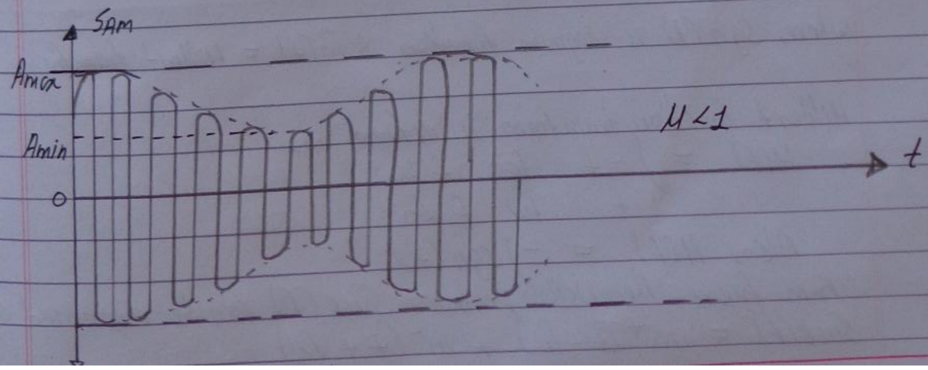
F. (5) The carrier wave is represented by $e_c(t) = I_0 \sin\omega t$. Draw the waveform of an AM wave for $m = 0.5$ 2017/F

sol^n,

$\Rightarrow \quad e_c(t) = I_0 \sin\omega t$

modulation factor $(\mu) = m = 0.5 < 1$, so,





G. Prove that $S_{SSB}(t) = \frac{A_c}{2}[m(t)\cos 2\pi f_c t \mp \hat{m}(t)\sin 2\pi f_c t]$

Where, $S_{SSB}(t)$ = modulated signal & $m(t) = A_m \cos(2\pi f_m t)$, 2015 k

$c(t) = A_c \cos(2\pi f_c t)$

⇒ Copy figure from page No: 72 ; Figure only.

let, S_{SSB} be an SSB modulated wave with upper sideband & $S_{LSB}(t)$ with lower sideband and $S_{USB}(f)$ & $S_{LSB}(f)$ be the corresponding in frequency domain. let, $M(f)$ be message spectrum. Right half $M^+(f)$ and left half $M^+(f)$ of the message spectrum.

$$M^+(f) = M(f) \cdot U(f)$$

$$= M(f) \cdot \times \frac{1}{2}[1 + sgn(f)]$$

$$= \frac{1}{2}[m(f) + j(-j sgn(f) \cdot m(f)]$$

$$= \frac{1}{2}[m(f) + j\hat{m}(f)]$$

similarly, $m^-(f) = \frac{1}{2}[m(f) - j\hat{m}(f)]$

where, sgn( ) is signum function & $\hat{m}(f)$ = Hilbert transform

Hilbert function transform is defined as,

$$H(f) = \begin{cases} -f & \text{for } f > 0 \\ f & \text{for } f < 0 \end{cases}$$

Also, $H(f) = -j\,sgn(f)$

From figure, the USB spectrum $S_{USB}(f)$ can be expressed as,

$$S_{USB}(f) = m^+(f - f_c) + m^-(f + f_c)$$

$$= \frac{1}{2}[m(f-f_c) + j\hat{m}(f-f_c)] + \frac{1}{2}[m(f+f_c) - j\hat{m}(f+f_c)]$$

$$= \frac{1}{2}[m(f-f_c) + m(f-f_c)] - \frac{1}{2j}[\hat{m}(f-f_c) + \hat{m}(f+f_c)]$$

Using frequency shifting property, & taking inverse fourier transform,

$S_{USB}(t) = m(t)\cos(w_c t) - \hat{m}(t)\sin(w_c t)$

This is the required time-domain expression for upper sideband.

similarly, for lower sideband SSB,

$S_{LSB} = m(t)\cos(w_c t) + \hat{m}(t)\sin(w_c t)$

In general,

$\therefore S_{SSB}(t) = m(t)\cos(w_c t) \pm \hat{m}(t)\sin(w_c t).$

H. An audio signal given by $15 \sin 2\pi(2000t)$ amplitude modulates a sinusoidal carrier wave $60 \sin 2\pi(100000t)$, determine,

7

(i) modulation index

(ii) Frequency of signal & carrier

(iii) Frequency of USB & LSB of modulated wave.

(iv) % modulation

2015/S, 2015/F

Sol^n:

$\Rightarrow$ carrier wave, $c(t) = 60 \sin(2\pi \times 100000t)$

message ", $m(t) = 15 \sin(2\pi \times 2000t)$

Comparing with,

$m(t) = A_m \sin(2\pi f_m t + \phi)$

$= 15 \sin(4\pi \times 1000t + 0)$

$\therefore f_m = 2\ KHZ \quad ; A_m = 15$

and,

$c(t) = A_c \sin(2\pi f_c t + \phi)$

$= 60 \sin(2\pi \times 200000t + \phi)$

$\therefore f_c = 100$ KHZ ; $A_c = 60$

(i) modulation index, $\mu = m = |k_a \cdot m(t)|_{max}$
letting, $k_a = 0.02$, (Not given),
So,
$\mu = 0.02 \times 15 = 0.3$

(ii) % modulation $= 0.3 \times 100 = 30\%$

(iii) Frequency of carrier = 100 KHZ
" " message = 2 KHZ

(iv) Frequencies of USB $= f_c + f_m = (2 + 100)$ KHZ
$= 102$ KHZ
" " LSB $= f_c - f_m = (100 - 2)$ KHZ
$= 98$ KHZ

7. Calculate the % power saving for a DSB-SC signal for the
(5) % modulation of (i) 100% (ii) 50%. 2017/F
Sol^n,
=> $\% \mu = 100$
$\mu = 1$
% power saving $= \frac{P_c}{P_t} \times 100\%$

(i) $= \frac{P_c}{P_c\left(1+\frac{\mu^2}{2}\right)} = \frac{1}{\left(1+\frac{1^2}{2}\right)} = 66.67\%$

(ii) $\mu = 0.5$
% Power saving $= \frac{1}{\left(1+\frac{0.5^2}{2}\right)} = 88.89\%$

J. A certain AM transmitter radiates 9 kW with the carrier unmodulated and 10.125 kW when the carrier is simultaneously modulated, find,

(i) modulation index 2016/F

(ii) If another sine wave corresponding to 60 % modulation is transmitted simultaneously, determine total radiated power.

(7)

Soln.

⇒ power of carrier wave, $P_c = 9$ kW

total power when modulated, $P_t = 10.125$ kW

(i) modulation index $(\mu) = \sqrt{2\left(\frac{P_t}{P_c} - 1\right)}$

$$= \sqrt{2 \times \left(\frac{10.125}{9} - 1\right)}$$

$$= \sqrt{0.25}$$

$\therefore \mu = 0.50$

(ii) $\% \mu_2 = 60 \% \Rightarrow \mu_2 = 0.60$

Total radiated power, $P_t = ?$

$\mu_1 = 0.50$

$P_c = 9$ kW,

The total modulation index by two sinusoidal signal is

$\therefore \mu_t = \sqrt{\mu_1^2 + \mu_2^2} = \sqrt{0.6^2 + 0.5^2} = 0.7810$

we have,

$$P_t = P_c\left(1 + \frac{\mu^2}{2}\right) = 9 \times \left(1 + \frac{0.78^2}{2}\right)$$

$\therefore P_t = 11.745$ kW

K. The total power content of an AM signal is 900 watt. Determine the power being transmitted at the carrier frequency & at each of the sidebands when % modulation is 75 %. 2014/S, 2015/F

(7)

Sol^n,

$\Rightarrow$ Total power (P_t) = 900 watt.

% modulation (μ) = 75 %

μ = 0.75

we have,

$P_t = P_c + P_{SB}$

$= P_c + P_c \times \frac{\mu^2}{2}$

or $P_t = P_c \left(1 + \frac{\mu^2}{2}\right)$

or $P_c = \frac{900}{\left(1 + \frac{0.75^2}{2}\right)}$

$\therefore P_c = 702.44$ watts.

Also,

$P_{LSB} = P_{USB} = \mu^2 \times \frac{P_c}{4}$

$= (0.75)^2 \times \frac{702.44}{4}$

$\therefore P_{LSB} = P_{USB} = 98.78$ watt

2) A system has bandwidth of 44 KHZ & signal to noise ratio of 28 dB at input to receiver, calculate

(i) information carrying capacity

(ii) capacity of channel if its Bandwidth is double & transmitted signal power remains constant. 2015/F

Sol^n,

$\Rightarrow$ Bandwidth (B.W) = 44 KHZ

signal to noise ratio (SNR) = 28 dB.

Also,

$$\boxed{C = 0.332 \times Bandwidth \times SNR\ in\ dB}$$

We know,

(i) Carrying capacity (C) = $B \log_2 (1 + SNR)$

Here,

SNR in dB = 28 dB = $10 \log_{10} (SNR)$

or $SNR = 10^{2.8} = 10^{28/10}$

$\therefore SNR = 630.957$

So,

$C = B\log_2 (1 + SNR) = 44 \times 10^3 \log_2 (1 + 630.957)$

$\therefore$ Carrying capacity (C) = 409362.033 bps
= 0.4094 Mbps

(ii) Bandwidth is doubled (B) = 2 × 44 = 88 KHZ,

since, power transmitted remains constant, $P = E_b \cdot R_b$,

$C = B \log_2 (1 + SNR)$

$= 88 \times 10^3 \log_2 \left(1 + \frac{\text{signal power}}{\text{Noise power}}\right)$

$= 88 \times 10^3 \log_2 \left(1 + \frac{630.957}{2}\right)$

$= 730924.8041$ bps

$= 0.7309$ Mbps.

M. (5) A single tone FM is represented by the voltage eq^n as $v(t) = 15 \cos (8 \times 10^{-8} t + 10 \sin 1520 t)$. Determine: 2017/S

(i) C.F

(ii) modulating frequency

(iii) mf

(iv) $(\Delta \omega)_{max}$.

V. What power will this FM wave dissipate in 20 Ω resistor.

Sol^n:

$\Rightarrow v(t) = 15 \cos (8 \times 10^{-8} t + 10 \sin 1520 t)$

Comparing with $s(t) = A_c \cos[2\pi f_c t + \beta \sin(2\pi f_m t)]$

$\therefore$ $A_c = 15$ V

(i) $\therefore$ $f_c = 4 \times 10^{-8}$ Hz

(ii) $\therefore$ $f_m = \frac{1520}{2} = 760$ Hz

$\therefore$ $\beta = 10$

(iii) Hence, $\Delta f = \beta f_m = 10 \times 760 = 7600$ Hz

Using Carson's rule,

(iv) $\therefore$ $B.W = 2(\beta + 1) f_m$
$= 2(1+10) \times 760$
$= 16720$ Hz

v) power dissipated across resistor, $P = \frac{A_c^2}{2R}$
$= \frac{15^2}{2 \times 10}$
$\therefore P = 11.25$ watt

---

x) Compute the maximum bit rate for a channel having Bandwidth of 3.1 kHz and the SNR of 40 dB. Also calculate the no. of levels required to transmit at the maximum bit rate. 2016/S

(8)

Sol^n,

$\Rightarrow$ channel capacity (C) = ?

Bandwidth (B) = 3.1 kHz

SNR = 40 dB.

we have,

$C = B \log_2(1 + SNR)$

Also, $C = 0.332 \times B \times SNR$
$= 0.332 \times 3.1 \times 40$

$\therefore$ $C = 41.168$ Kbps

Now,

No. of levels Required (L) = ?

We have,

Using Nyquist formula,

$$C = 2 \times B \times \log_2(L)$$

or $41.168 \times 1000 = 2 \times 3.1 \times 1000 \times \log_2(L)$

or $\log_2(L) = 6.64$

or $L = 2^{6.64} = 6.64$ bits/baud.

$\therefore$ $L = 99.73 \simeq 100$

Hence, No. of levels required to transmit at maximum bit rate is 6.64 bits/baud.

Q. Verify that the transmission efficiency of single-tone modulating signal is $P_t = P_c\left(1 + \frac{\mu^2}{2}\right)$ 2016/S

(5)

=> Consider a modulating wave m(t) that consists of a single tone or frequency component, ie,

$$m(t) = A_m \cos(2\pi f_m t) \quad — ①$$

where, A_m is peak amplitude of modulating wave.

f_m = frequency of modulating wave.

Carrier frequency is, $c(t) = A_c \cos(2\pi f_c t)$ — ⑪

We know,

$$s(t) = [A_c + m(t)] \cos(2\pi f_c t)$$

$$= [A_c + A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$$

$$= A_c\left[1 + \frac{A_m}{A_c} \cos(2\pi f_m t)\right] \cos(2\pi f_c t)$$

$s(t) = A_c [1 + \mu \cos(2\pi f_m t)] \cos(2\pi f_c t)$

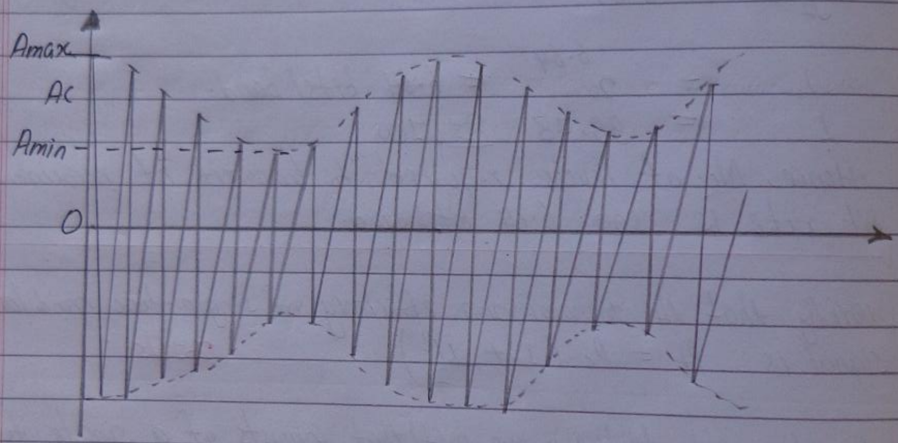
Comparing with standard AM,

$\mu = k_a \cdot A_m = \max \cdot |k_a m(t)|$

Where,

μ = dimensionless quantity = modulation factor

To avoid envelope distortion due to over-modulation, μ must be kept below unity.



Here, $A_{max} = A_c(1 + \mu)$

$A_{min} = A_c(1 - \mu)$

Now, $\dfrac{A_{max}}{A_{min}} = \dfrac{1 + \mu}{1 - \mu}$

or $\mu = \dfrac{A_{max} - A_{min}}{A_{max} + A_{min}}$

Also, $A_m = \dfrac{A_{max} - A_{min}}{2}$

$A_c = \dfrac{A_{max} + A_{min}}{2}$

$\therefore \mu = \dfrac{A_m}{A_c} \times 100\ \%$

Again,

$s(t) = A_c [1 + \mu \cos(2\pi f_m t)] \cos(2\pi f_c t)$

Hence,

carrier power, $P_c = \frac{1}{2} A_c^2$

Upper side-frequency power, $P_{USB} = \frac{1}{8} \mu^2 A_c = \frac{1}{4} \mu^2 P_c$

lower " " ", $P_{LSB} = \frac{1}{8} \mu^2 A_c = \frac{1}{4} \mu^2 P_c$

Hence,

Total side band power, $P_{SB} = P_{USB} + P_{LSB} = \frac{1}{4} \mu^2 A_c = \frac{1}{2} \mu^2 P_c$

Total power, $P_t = P_c + P_{SB}$

$= P_c + \frac{1}{2} \mu^2 P_c$

$\therefore P_t = P_c \left(1 + \frac{1}{2} \mu^2\right)$

Q. Briefly explain the concept of exchangeability between bandwidth, signal power and SNR, then finally state shannon's eqn for channel capacity. 2014/S

Explain channel bandwidth importance with the help of Hartley-shannon law. 2015/S

⟹ shannon's channel capacity theorem states that,

$C = B \log_2 \left(1 + \frac{S}{N}\right)$ bits.

where, C = channel capacity

B = Channel bandwidth. in HZ

S = signal power
N = Noise power

- The channel capacity, C increases as the available bandwidth increases and as the signal to noise ratio increase (improves).
- This expression applies to information in any format & to both analog and data communications, but its application is most common in data communication.

- As the bandwidth increases, the capacity should increase proportionately. But this doesnot happen, because increasing the Bandwidth, B, also increases the noise power, $N = N_0 \cdot B$,

$$C = B \log_2\left(1 + \frac{S}{N}\right)$$

$$= B \log_2\left(1 + \frac{S}{N_0 B}\right)$$

$$= \frac{S}{N_0} \cdot \frac{N_0 \cdot B}{S} \log_2\left(1 + \frac{S}{N_0 \cdot B}\right)$$

$$\therefore \quad C = \frac{S}{N_0} \log_2\left(1 + \frac{S}{N_0 \cdot B}\right)^{(N_0 \cdot B)/S}$$

- A SNR of zero dB means that noise power equals the signal power. It is not possible to transmit at rate higher than 'C' reliability by any means.

$\therefore$ Signal to noise ratio $(S/N)_{dB} = 10 \log\left(\frac{\text{Signal power}}{\text{Noise }''}\right)$

Q. Define modulation & demodulation. Why modulation needed and what will happen if signal transmitted without modulation? 2014/S

=> Demodulation is extracting the original information bearing signal from a modulated carrier wave. ie, it is reverse process of modulation.

Modulation is the process of superimposing audio signal over the carrier (high frequency) wave. Modulation is the process of varying one or more properties of a periodic waveform.

Modulation is needed for:
Copy from page No 9, 10, 11

If signal is transmitted without modulation, there can occurs mutual interference between the signals. and the signal cannot be fed to long operating range.

R. Justify that, how, fundamental limitations of communication systems are
5 Noise, Bandwidth & equipment limitation. 2017/F

=> Every communication system has limited bandwidth that limits the signal speed. Noise imposes a second limitation on information transmission as it can also cause signal distortion. It is unavoidable. At any temperature above absolute zero, thermal energy causes microscopic particles to exhibit random motion. The random motion of charged particles such as electrons, generates random currents or voltage called thermal noise. Thermal noise exists in every communication system.

Due to various reasons, every communication system supports transmission at certain limited frequency bands only. Bandwidth of a channel is the range of frequencies that it can transmit with reasonable fidelity. So, the increase in bandwidth means the increase in number of frequencies. So, the transmission speed also increase. SNR is defined as the ratio of signal power to noise power. As we have no control on noise, increasing the signal power, we are able to increase SNR. That means reducing the signal effect.

$$C = B \log_2 (1 + SNR) \text{ bits/sec}$$

But, we cannot increase the signal power and channel bandwidth ultimately. Because they are exchangeable, which describes that increase in bandwidth causes decreases in signal power & vice-versa. Similarly equipment like receiver, encoder, transmitter, and other devices used for communication have its own noise, frequency and limitations;

Hence, noise, bandwidth and equipment limitations are limitation of communication system.

5. State shannon's channel capacity theorem. Given, an AWGN channel with 4 KHZ bandwidth & the noise power spectrum density of 10^{-12} W/Hz. The signal power required at the receiver is 0.1 mW calculate the capacity of the channel. 2016/F

(7)

Sol^n,

=> Noise power spectrum density (N_0) $= 10^{-12}$ W/HZ X2

Bandwidth (B) = 4 KHZ = 4000 Hz.

signal power transmit ($\bar{p}$) = 0.1 mW

= 0.0001 W

capacity of channel (c) = ?

We have,

$$C_{awgn} = B \log \left(1 + \frac{\bar{P}}{N_0 \cdot B}\right) \text{ bits/s.}$$

$$= B \log_2(1+SNR) \text{ bits/s}$$

$$SNR = \frac{P}{N_0 \cdot B}$$

$$C_{awgn} = 4000 \log_2\left(1 + \frac{0.0001}{2\times10^{-12} \times 4000}\right)$$

$$= 4000 \log_2 (12500) = 4000 \times 13.6096$$

$$\therefore C_{awgn} = 54438.562 \text{ bps}$$

$$= 54.44 \text{ kbps}$$

7) Theoretical limitation of shannon's channel capacity Theorem:

(i) The noise in channel tends to zero, $SNR \to \infty$, subsequently, $C \to \infty$, it means that noiseless channel has infinite bandwidth. This kind of channel is called ideal channel.

(ii) As the Bandwidth of the channel tends to infinite, the channel capacity reaches an upper limit C_{max}. This is because, noise power is proportional to the Bandwidth & as the bandwidth is increased, the noise power also increases correspondingly.

$$C = B \log_2\left(1 + \frac{S}{nB}\right)$$

n = power spectrum density
S = signal power
ie,

$$C = \left(\frac{S}{n} \cdot \frac{n}{S}\right) B \log_2 \left(1 + \frac{S}{n \cdot B}\right)$$

$$= \frac{S}{n} \log_2 \left(1 + \frac{S}{nB}\right)^{(nB)/S}$$

let, $Y = \frac{S}{nB}$ & considering limit,

$$\lim_{Y \to 0} (1+Y)^{1/Y} = e$$

$$\lim_{B \to \infty} C = C_{max} = \frac{S}{n} \log_2 e = 1.44 \left(\frac{S}{n}\right)$$

$$\therefore C_{max} = 1.44 \times \frac{S}{n}$$

Also,

$$\frac{C}{B} = \log_2 (1 + SNR)$$

$$\Rightarrow SNR = 2^{(C/B)} - 1$$

(5) U. Compares BASK, BFSK, BPSK for variables Bandwidth, bitrate and performance. 2014/S, 2017/F

⇒

	BASK	BFSK	BPSK
(i)	Bandwidth = $2f_b$	$4f_b$	$2f_b$
(ii)	suitable upto 100 bits/sec	upto about 1200 bits/se	Bitrate is the no. of digital bits sent/second. It has 1 bit/symbol. High bit rate.
(iii)	BASK performance in presence of noise is poor.	BFSK is relatively simple, low performance form of digital modulation.	It is very robust transmission method but consumes significant bandwidth.
iv)	simple system	moderately complex system	very complex system.
v)	Noise Immunity is Low	High	High.
vi	probability of error is high	low	low.

95

v) Differentiate between coherent and non-coherent digital modulation techniques. Explain DPSK system. 2016/F.
(7)

Coherent digital modulation	Non-coherent digital modulation.
(i) Better performance on AWGN channel, slow flat fading.	Better performance in fast frequently selective fading.
(ii) more complex than non-coherent demodulation & harder to implement.	Less complex than coherent detection and easier to implement but has worse performance.
(iii) expensive	Relatively less expensive.
iv). In coherent modulation technique, process received signal with a local carrier of same frequency & phase.	Here, no requirement of reference wave.
v) It is those technique which employ coherent detection. In coherent detection, the local carrier generated at the receiver is phase locked with the carrier at the Transmitter. Thus, the detection is done by correlating received noisy signal & locally generated carrier. The coherent detection is also known as synchronous detection.	It is those technique in which the detection process doesnot need receiver carrier to be phase locked with transmitter ckt. But, the drawback of such system is that error probability increases.

DPSK : 2015/S, 2016/F

A DPSK system may be viewed as the non-coherent version of of the PSK. It eliminates the need for coherent reference signal at the receiver by combining two basic operations at the transmitter.

(I) Differential encoding of the input binary wave &
(II) phase shift keying.

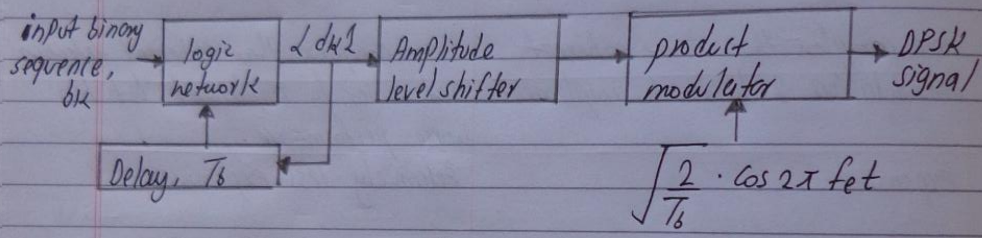


Fig: DPSK Transmitter

To send symbol '1' we phase advance the current signal waveform by 180° & to send symbol '0' we leave the phase of the current signal waveform unchanged.

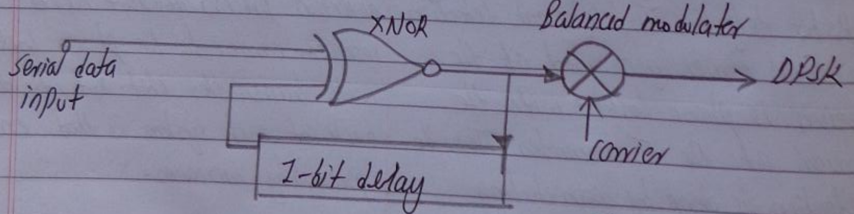


Fig: DPSK modulator.

The differential encoding process at the transmitter input starts with an arbitrary first bit serving as reference & thereafter the differentially encoded sequence $\{d_k\}$ is generated by using the logical eq^n,

$$d_k = d_{k-1} \oplus b_k.$$

w) Determine minimum Bandwidth for a BPSK modulator with carrier
(5) frequency of 50 MHZ & input bit rate of 500 kbps. 2017/F, 2014/S

Sol^n,

⇒ minimum Bandwidth (B) = ?

carrier frequency (f_c) = 50 MHZ

input bit rate (f_b) = 500 kbps

we have,

$$B = 2 f_a$$

$$= 2 \times \frac{f_b}{2}$$

$$= f_b$$

$$\therefore B = 500 \text{ kbps}$$

Hence, minimum double sided Nyquist Bandwidth = 500 kbps.

X. A carrier wave of frequency 91 MHZ is frequency modulated by a sine wave of amplitude 10V & 15 KHZ. The frequency sensitivity
(8) of the modulator is 3 kHZ/V. 2016/F

(i) Determine the approximate bandwidth of FM wave using carson's rule.

(ii) Repeat part (i), assuming that the amplitude of the modulation wave doubled and frequency of the modulating wave is halved.

Sol^n,

⇒ carrier wave frequency, f_c = 91 MHZ = 91000 kHZ

modulating frequency, f_m = 15 kHZ

modulating message signal amplitude, V_m = 10 V

k_f = 3 kHZ /V

we know,

$$\text{modulation index } (m_f) = \frac{k_f V_m}{f_m}$$

$$\therefore m_f = \frac{3 \times 10}{15} = 2$$

(i) Bandwidth, $BW = 2\Delta f + 2f_m$
$= 2m_f \cdot f_m + 2f_m$
$= 2f_m(1 + m_f)$
$= 2 \times 15(1 + 2)$
$= 90\ KHZ$

(ii) amplitude double, $= V_m' = 20V$
frequency of modulating wave is Halved $= f_m' = 7.5\ kHz$
Hence,

$$m_f' = \frac{k_f \cdot V_m'}{f_m'}$$

$$= \frac{3 \times 20}{7.5}$$

$$\therefore m_f' = 8$$

So,
Bandwidth $(BW) = 2\Delta f' + 2f_m'$
$= 2f_m'(1 + m_f')$
$= 2 \times 7.5(1 + 8)$
$\therefore BW = 135\ KHZ$

Y. Compare :

DSB-SC	SSB	VSB
(i) power : medium	less	High
(ii) Bandwidth : $2f_m$	f_m	$f_m < BW < 2f_m$
(iii) used for : Radio telecommunication	Radio communication	Television.
iv) carrier suppression takes		

	place completely	– completely	No suppression.
v)	Transmission efficiency: moderate	maximum	moderate.
vi.	sideband suppression: – No	one sideband completely	one sideband partially
vii	more complex to modulate & demodulate.	more difficult to modulate & demodulate	easier to implement.

2) The rms value of carrier voltage is 80 V. After amplitude modulation by a sinusoidal audio signal, the rms value of the carrier voltage becomes 96 volts, determine modulation factor.

Sol^n,

=> carrier voltage $(V_c) = 80\ V$

Total modulated voltage $(V_T) = 96\ V$

modulation index $(\mu) = \sqrt{\left[\left(\frac{V_T}{V_c}\right)^2 - 1\right] \times 2}$

$$= \sqrt{\left[\left(\frac{96}{80}\right)^2 - 1\right] \times 2}$$

$$= \sqrt{0.44 \times 2}$$

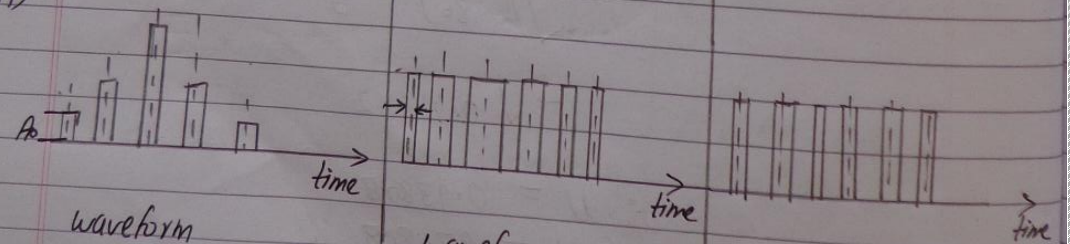
$$\therefore \mu = 0.93808$$

$$\& \quad \%\ \mu = 93.808\ \%$$

2015/F

2) A. Compare:

pulse amplitude modulation (PAM)	pulse width modulation PWM / PDM	pulse position modulation, PPM
(i) Amplitude of the pulse is proportional to amplitude of modulating signal.	width of the pulse is proportional to amplitude of modulating signal.	The relative position of the pulse is propotional to the amplitude of modulating signal.
ii) Bandwidth of the transmission channel depends on width of the pulse.	– depends upon the rise time of the pulse.	– depends on rising time of the pulse.
iii) Noise, interference is high. complex system.	Noise, interference is minimum.	minimum.
iv) Instantaneous power of transmitter varies.	Instantaneous power of transmitter varies.	– Remains constant.
v. similar to amplitude modulation.	similar to frequency modulation.	similar to phase modulation.
vi) waveform	waveform	waveform.



B) Signal to Quantization Noise Ratio for linear quantization:

We know, that in a PCM system for linear quantization the signal to noise ratio is given as,

$$\therefore \frac{S}{N} = \frac{\text{normalized signal power}}{\text{Normalized Noise ''}}$$

But, normalized noise power is $= \frac{\Delta^2}{12}$

Hence

$$\therefore \frac{S}{N} = \frac{\text{normalized signal power}}{(\Delta^2/12)}$$

We know that the number of bits 'v' and quantization levels are,

$$q = 2^v,$$

Also,

$$\text{Step size, } \Delta = 2\frac{x_{max}}{q}$$

$$\therefore \Delta = 2 \cdot \frac{x_{max}}{2^v}$$

$$\text{or } \frac{S}{N} = \frac{\text{normalized signal power}}{\left(2\frac{x_{max}}{2^v}\right)^2 \cdot \frac{1}{12}}$$

let, normalized signal power be denoted as 'P,

$$\text{then, } \frac{S}{N} = \frac{P}{\frac{4x^2_{max}}{2^{2v}} \times \frac{1}{12}} = \frac{3P}{x^2_{max}} \times 2^{2v}$$

Hence, signal to quantization noise ratio is,

$$\therefore \frac{S}{N} = \left(\frac{3P}{x^2_{max}}\right) \cdot 2^{2v}$$

This expression shows that signal to noise power ratio of quantizer increases exponentially with increasing bits per sample.

let, input $x(t)$ is normalized, ie,

$x_{max} = 1$,

Signal to quantization noise ratio will be,

$$\therefore \frac{S}{N} = 3 \times P \times 2^{2v}$$

Also, if destination signal power 'P' is normalized, ie,

$P \leq 1$,

Then, signal to noise ratio will be,

$$\frac{S}{N} \leq 3 \times 2^{2v}.$$

$$\therefore \left(\frac{S}{N}\right) dB = 10 \log_{10} \left(\frac{S}{N}\right) dB \leq 10 \log_{10} [3 \times 2^{2v}] \leq (4.8 + 6v) dB$$

c) A compact disc (CD) records audio signals digitally by PCM. Assume audio signal's bandwidth to be 15 kHz, if signals are sampled at a rate of 20% above Nyquist rate for practical reasons & the samples are quantized into 65,536 levels, determine bits/sec required to encode the signal and minimum bandwidth required to transmit encoded signal.

sol^n,

$\Rightarrow$ Given, $f_m = 15$ kHz,

$f_s = 1.2 \times 2 f_m = 2.4 \times 15 \text{ kHz} = 36$ kHz

$q = 65,536$

Signaling rate (r) :

$$q = 2^v$$

$$v = \log_2 q$$

$$v = \frac{\log_{10} (65536)}{\log_{10} 2} = 16$$

Now, signaling rate, $r = v \cdot f_s = 16 \times 36$ kHz

$= 576$ kb/sec

(ii) minimum bandwidth,

$$B.W = \frac{1}{2}\text{ (signaling rate) } = \frac{576}{2} = 288 \text{ KHZ.}$$

D) An analog waveform with bandwidth 15 KHZ is to quantized with 200 levels and transmitted via binary PCM signal. Find rate of transmission and bandwidth required. If 20 such signals are to be multiplexed, find the bandwidth required.

Sol^n

=> $f_m = 15$ KHZ

$Q = 200$

$f_s = 2 f_m = 2 \times 15 \times 10^3 = 30 \times 10^3$ HZ

$Q = 2^N$

No. of bits / sample, $N = \log_2 Q$

$$= \frac{\log_{10} 200}{\log_{10} 2}$$

$\therefore N = 8$

we know that rate of transmission $= N.f_s$

$= 8 \times 30 \times 10^3 = 240$ kbps

Transmission Bandwidth $= \frac{1}{2} \times N \times f_s$

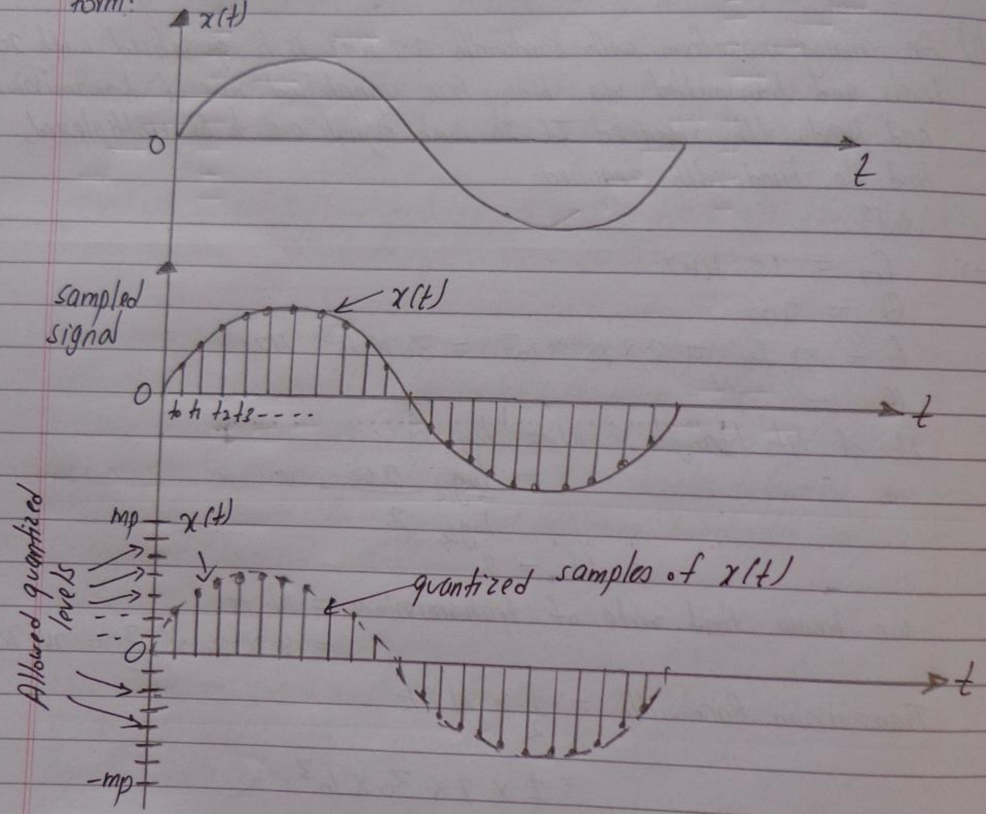
$$= \frac{1}{2} \times 8 \times 30 \times 10^3$$

$\therefore BW = 120$ KHZ

E) Explain the concept of quantization and classify it in details.

=> Quantization is a process of approximation. The samples are

approximated or rounded off to the nearest quantization level.
In communication system, sometimes, it happens that we are available with analog signal however, we have to transmit a digital signal for a particular application. In such cases we have to convert a continuous time signal in the form of digits. ie, digital form:



we get sample of the signal according to sampling theorem. For this purpose, we mark time instants $t_0, t_1, t_2, \ldots$ & so on, at equal time intervals along the time axis. This means that it no longer is a continuous function of time, but rather, it is a discrete time signal. However, since the magnitude of each sample can take any value in a continuous range, the signal of sampled is still an analog. This difficulty is neatly resolved by a process known as quantization. In quantization, the total amplitude range which the signal may occupy is divided into a number of standard levels. Amplitude of the signal $x(t)$ lie in the range $(-m_p, m_p)$ which is partitioned into L intervals, each of magnitude, $\Delta V = \frac{2m_p}{L}$. Now, each sample is approximated to the nearest quantized level. since, each sample is now approximated to one of the L numbers, hence the information is digitized.

Classification of Quantization process:

Quantization:

- Uniform quantization
  - Mid-tread type
  - Mid-rise type
- Non-uniform quantization

(i) Uniform quantizer:

It is that type of quantizer in which the step size remains same throughout the input range.

(ii) Non-uniform Quantizer:

It is that type of quantizer in which the 'step size' varies according to the input signal values.

Types of uniform quantizer:

(i) symmetric quantizer of the midtread type
(ii) Symmetric " " " midrise "

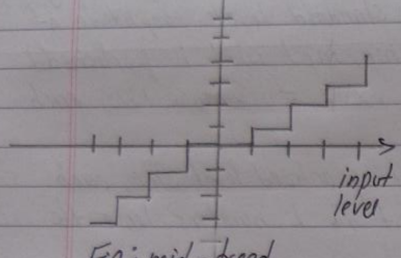


Fig: mid-tread

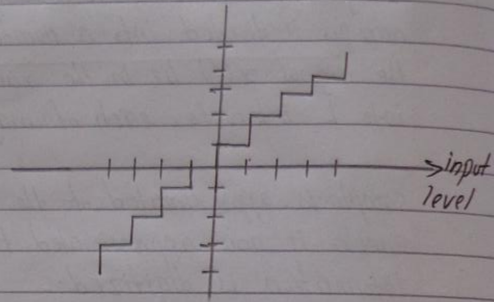


Fig: mid-rise

In midrise type, origin lies in the middle of rising part of the stair case like graph.
In midtread type, origin lies in the middle of a tread of the staircase like graph.

F). Compare Am and ASK

parameter of comparision	Am	ASK
variable characteristics of the carrier ⟶	amplitude	amplitude.
	modulating signal is analog	digital
No. of sidebands produced	Two	Two
Band width	$2f_m$	$(1+r)R$
Noise immunity	poor	poor
Application	Radio broadcasting.	Data transmision at low bit rate.

G. Compare :

BPSK	DPSK
i) variable characteristics is phase.	phase.
ii) Bandwidth : f_b	f_b.
iii) probability of error : low	Higher than BPSK
iv) Noise effect is low	" " "
v) Demodulation technique : synchronous	synchronous.
vi) system complexity : lower than DPSK	Higher than BPSK.
vii) Requirement of synchronous carrier is required.	Not required.

H)

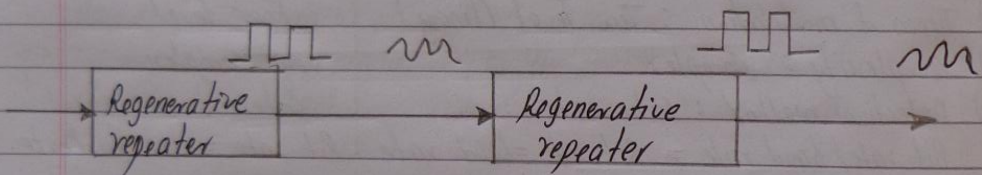
BPSK	QPSK
i) variable characteristics is phase	phase
ii) Types of modulation : Two level (binary)	Four level
iii) Complexity : Complex	very complex.
iv) Detection method : Coherent	Coherent.
v) Bit rate/baud rate Bit rate = baud rate	Bit rate = 2 X baud rate.
vi) Types of representation: A binary bit is represented by one phase state.	A group of two binary bits is represented by one phase state.

I) write short notes on : Regenerative Repeater :

Regenerative repeaters are used at regularly spaced intervals along a digital transmission line to detect the incoming digital signal & regenerate new clean pulses for further transmission along the line.

This process periodically eliminates and thereby combacts, the accumulation of noise and signal distortion along the transmission path. If the pulses are transmitted at a rate of R_b pulses per second, we require the periodic timing information, - the clock signal at R_b HZ - to sample the incoming pulses at a repeater. This timing information can be extracted from the received signal itself if the line code is chosen properly.

In digital systems, the 'repeater' detects and regenerates a clean signal (noise free) along the transmission channel. If an error occurs in the detector, this error is propagated forward. Regenerative repeaters are used on wired line, fiber & wireless transmission systems. It consists of demodulators / detectors and transmitters.



Advantages:

- improvement in overall error rate performance.
- Ease of amplification at Baseband.

J) A TV signal having a Bandwidth of 4.2 MHZ transmitted using PCM system. Given that no. of quantization level is 512, find,

(i) Code word length
(ii) Transmission Bandwidth
(iii) Final bit rate
(iv) output signal to quantization noise ratio.

⇒ Given, Bandwidth = 4.2 MHZ = 4.2×10^6 Hz
No. of quantization level, L = 512,

(i) Code word length (b) = ?
we know,

$$L = 2^b$$

$$\therefore b = \log_2(L) = \log_2(512) = 9 \text{ length.}$$

(ii) Bandwidth, $\geq b \cdot f_m = 9 \times 4.2 \times 10^6 \geq 37.8$ MHZ

(iii) Final bit rate $= r = b \cdot f_s = 9 \times B.W$

$$= 2 \times b \cdot f_m$$

$$= 37.8 \times 2$$

$$= 75.6 \times 10^6 \text{ bps}$$

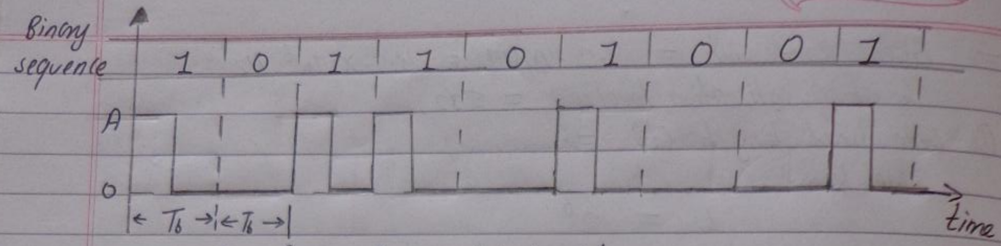
iv) SQNR $\leq (4.8 + 6b) \leq 4.8 + 6 \times 9 \leq 58.8$ dB

k. Various PAM Formats for line codes:

(i) Manchester format
(ii) polarity quaternary NRZ format
(iii) Non-return to zero bipolar format.
(iv) NRZ and RZ polar format.
v) Non-return to zero (NRZ) and return to zero (RZ) unipolar format.

(i) Unipolar RZ:

The waveform has zero value when symbol '0' is transmitted and waveform has 'A' volts when 1 is transmitted. In RZ form, the 'A' volts is present for $T_b/2$ period if symbol 1 is transmitted & for remaining $T_b/2$ waveform returns to zero value, ie, for unipolar RZ form, we have,



ie, $x(t) = \begin{cases} A & \text{for } 0 \le t \le T_b/2 \\ 0 & \text{for } T_b/2 \le t < T_b \end{cases}$

(ii) <u>Unipolar NRZ :</u>

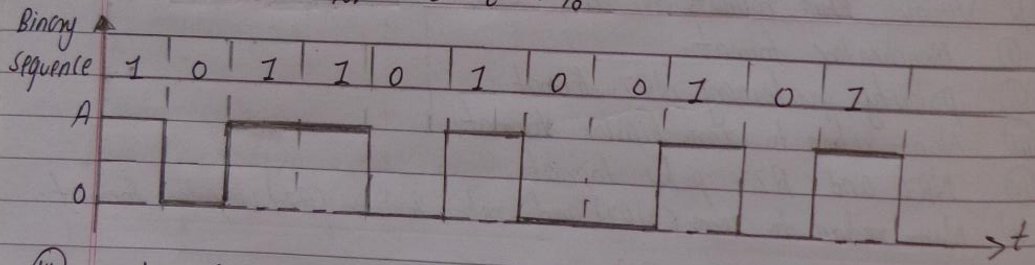
A unipolar NRZ format is shown below. When symbol 1 is to be transmitted, the signal has 'A' volts for full duration. When symbol 0 is to be transmitted, the signal has zero volts, (ie, no signal for complete symbol duration.

If symbol 1 is transmitted, we have,

$x(t) = A \quad \text{for} \quad 0 \le t < T_b$

If symbol 0 is transmitted, we have,

$x(t) = 0 \quad \text{for} \quad 0 \le t < T_b$



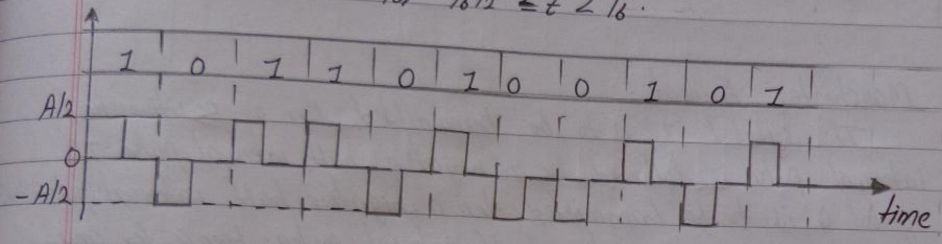
(iii) <u>polar RZ :</u>

In the polar format, symbol 1 is represented by positive voltage polarity whereas symbol 0 is represented by negative voltage polarity. Because this is RZ format, the pulse is transmitted only for half duration. Thus, for polar RZ, if symbol 1 is transmitted, then,

$$x(t) = \begin{cases} A/2 & \text{for } 0 \le t < T_b/2 \\ 0 & \text{for } T_b/2 \le t < T_b \end{cases}$$

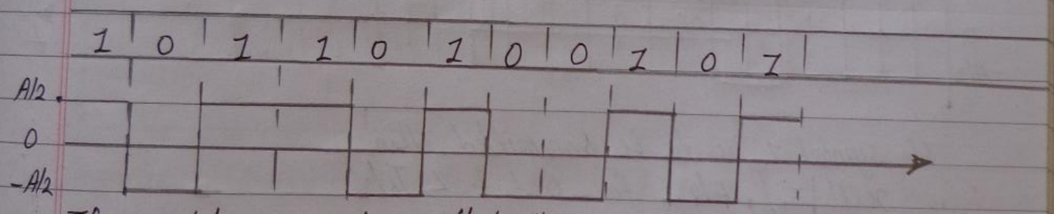
If symbol 0 is transmitted, then,

$$x(t) = \begin{cases} -A/2 & \text{for } 0 \le t < T_b/2 \\ 0 & \text{for } T_b/2 \le t < T_b \end{cases}$$



iv) Polar NRZ :

In polar NRZ format, symbol 1 is represented by positive polarity whereas symbol 0 is represented by negative polarity. These polarities are maintained over the complete pulse duration.



If symbol 1 is transmitted, then,

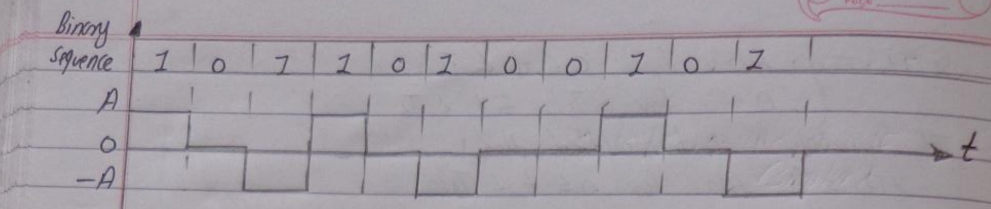
$x(t) = +A/2$ for $0 \le t < T_b$

If symbol 0 is transmitted, then,

$x(t) = -A/2$ for $0 \le t < T_b$.

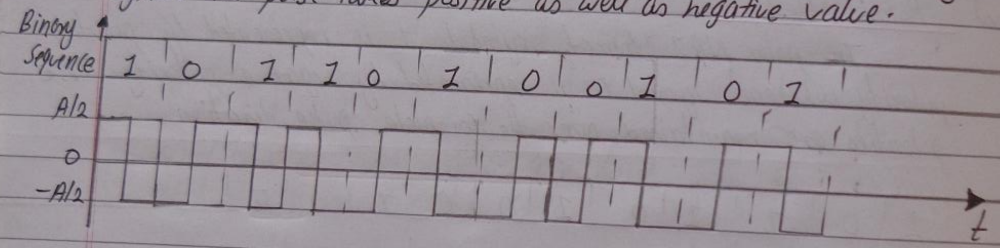
v) Bipolar NRZ [alternate mark inversion] AMI :

In this format, the successive 1 are represented by pulses with alternate polarity and 0's are represented by no pulses.



v) Manchester Format:

If symbol '1' is to be transmitted, then a positive half interval pulse is followed by a negative half interval pulse. If symbol 0 is to be transmitted, then a negative half interval pulse is followed by a positive half interval pulse. Hence, for any symbol the pulse takes positive as well as negative value.



If symbol 1 is to be transmitted, then,

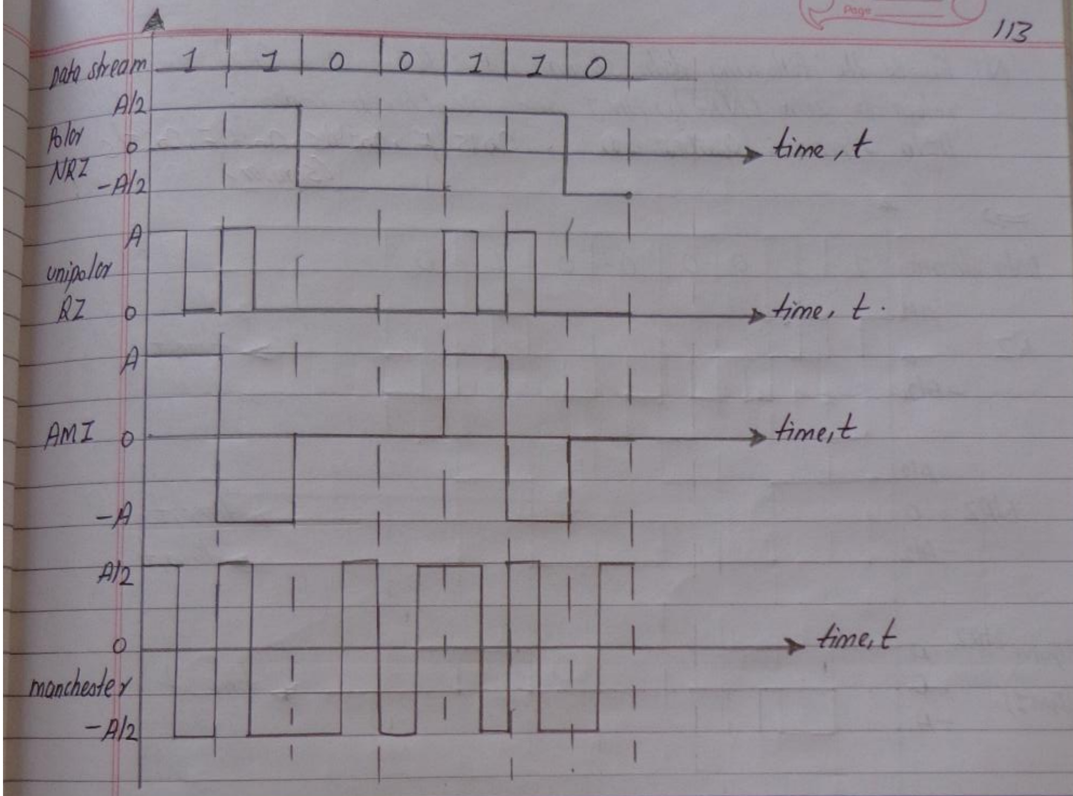
$$\therefore \quad x(t) = \begin{cases} A/2 & \text{for } 0 \le t < T_b/2 \\ -A/2 & \text{for } T_b/2 \le t < T_b \end{cases}$$

If symbol 0 is to be transmitted, then,

$$\therefore \quad x(t) = \begin{cases} -A/2 & \text{for } 0 \le t < T_b \\ A/2 & \text{for } T_b/2 \le t < T_b \end{cases}$$

1) Draw the following data formats for the bit stream 1100110
i) polar NRZ
ii) unipolar NRZ
iii) AMI
iv) Manchester

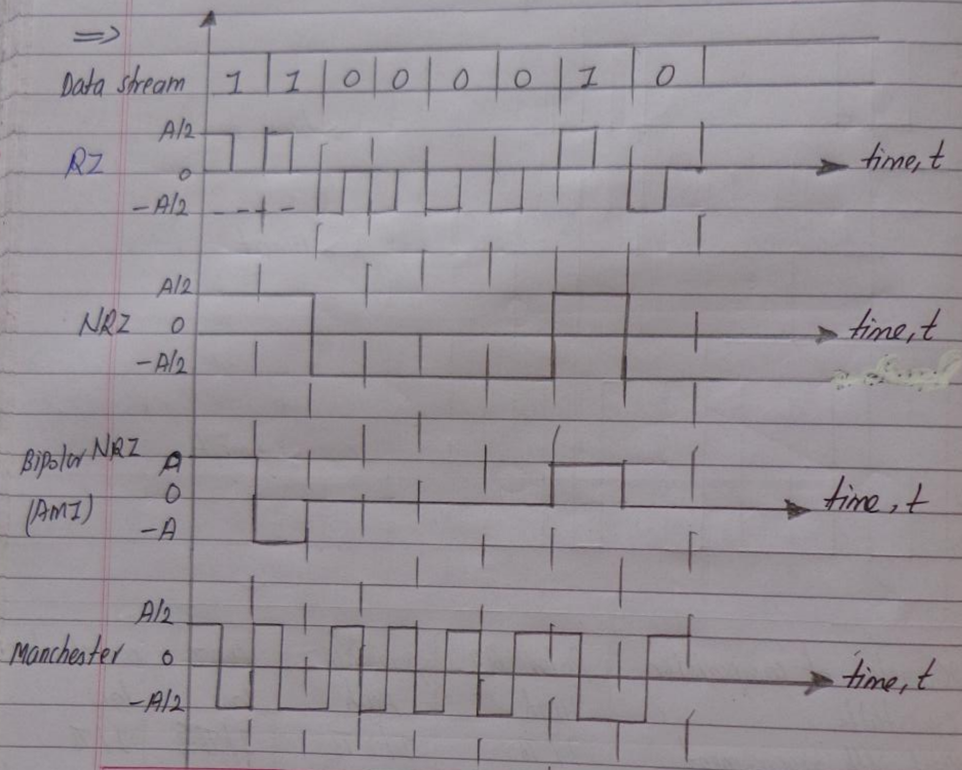
113



M) parameter of comparision	Polar RZ	polar NRZ	AMI	manchester
① Crosstalk	High	High	low	low
ⓘⓘ Bandwidth requirement	1/Tb	1/2 Tb	1/2 Tb	1/Tb
ⓘⓘⓘ synchronizing capability	poor	poor	very good	very good
iv) Noise immunity	low	low	High	High
v) signalling rate	1/Tb	1/Tb	1/Tb	1/Tb
vi) Transmission of DC Component	Yes	Yes	No	No

N. Encode the following data stream into: Return zero (RZ), non-return to zero (NRZ), AMI and Manchester codes:
Data stream : 11000010 . 2015/F, 2014/S, 2016/F, 2015/S (similar)

⟹



O. A voice grade channel of the telephone network has a bandwidth of 3.4 kHz. Determine the information capacity of the telephone channel for a signal to noise ratio of 30 dB.

Sol^n,

⟹ Bandwidth (B) = 3.4 kHz

Channel capacity (C) = ?

signal to noise-ratio $(S/N) = 30$ dB

or $\left[\frac{S}{N}\right]_{dB} = 10 \log_{10}\left[\frac{S}{N}\right]$

or $30 = 10 \log_{10}\left[\frac{S}{N}\right]$

or $[S/N] = \text{antilog}(3) = 1000$

Now,

$C = B \log_2\left[1 + \frac{S}{N}\right]$

$= 3.4 \times 10^3 \log_2 [1 + 1000]$

$= 33888.58$ bits/sec

$\therefore C = 33.89$ kbps

P. A channel has a bandwidth of 8 kHz, what is channel capacity if signal to noise ratio being 31. For same channel capacity, if signal to noise ratio is increased to 61, then, what will be new channel capacity?

soln,

$\Rightarrow$ Bandwidth $(B) = 8$ kHz

signal to Noise ratio $(S/N) = 31$

Using shannon Hartley theorem,

$C = B \log_2 [1 + S/N]$

$= 8 \times 10^3 \log_2 [1 + 31]$

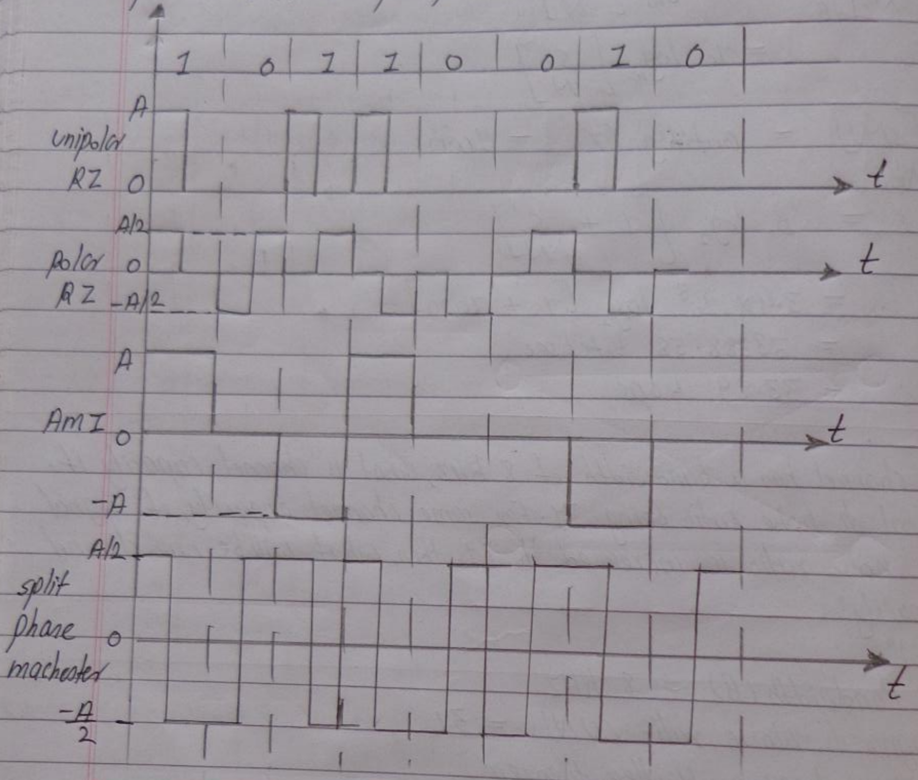
$\therefore C = 40 \times 10^3$ bits/sec

The new Bandwidth when $S/N = 61$, assuming channel capacity C to be constant.

$40 \times 10^3 = B \log_2 (1 + 61)$

or $B = \frac{40 \times 10^3}{5.954} = 6.71$ kHz

Q. Encode the given data stream = 10110010 into unipolar RZ, polar RZ, AMI, split phase manchester.



R) Bandwidth of the input to pulse code modulator is restricted to 4 kHz. The input varies from −3.8 V to +3.8 V. & has the average power of 30 mW, the required signal to quantization noise power ratio is 20 dB. The modulator produces binary output. Assume uniform quantization. Calculate the no. of bits required per sample.

soln,

Sol^n;

$\Rightarrow$ Given, $\left(\frac{S}{N_q}\right)_{dB} = 20\ dB$

or $10 \log \left(\frac{S}{N_q}\right)_0 = 20\ dB$

or $\left(\frac{S}{N_q}\right)_0 = \text{antilog}\left(\frac{20}{10}\right) = 10^2$

$\therefore$ $(S/N_q)_0 = 100$

Quantization step size, $\Delta = \frac{2A}{L}$

where,

$L = 2^n$, n = number of binary digits,

Average quantizing power:

$N_q = \overline{(q_e)^2} = \frac{\Delta^2}{12} = \frac{A^2}{3L^2}$

or $\left(\frac{S}{N_q}\right)_0 = \frac{\text{average signal power}}{\text{Average quantizing power}}$

or $100 = \frac{30 \times 10^{-3}}{A^2/3L^2}$

or $L = \sqrt{\frac{30 \times 10^{-3}}{3 \times 100 \times 3.8^2}} = 126.67$

or $2^n = 128 \Rightarrow 2^n = 2^7$

$\therefore$ $n = 7$

Hence, No. of bits required/sample $= 7$.

5) Derive an expression for signal to quantization noise ratio for a PCM system which employs linear (ie. uniform) quantization technique. Given that input to the PCM system is a sinusoidal signal.

$\Rightarrow$ let, us assume that the modulating signal is a sinusoidal signal (voltage), having peak amplitude equal to A_m. Also, let the signal cover the complete excursion of representation levels. Then, power of this signal is,

$$P = \frac{V^2}{R}$$

V = rms value ie, $V = \left[\frac{A_m}{\sqrt{2}}\right]^2$

we have, $P = \frac{A_m^2}{2} \cdot \frac{1}{R}$

In case, when $R=1$, the power P is normalized, ie,
Normalized power, $P = \frac{A_m^2}{2}$ with $R=1$,

we know, signal to quantization noise ratio is,

$$\frac{S}{N} = \frac{3P}{x^2_{max}} \times 2^{2V}$$

or, $P = \frac{A_m^2}{2}$

and, $x_{max} = A_m$

so,

$$\frac{S}{N} = 3 \times \frac{A_m^2/2 \times 2^{2V}}{A_m^2}$$

or $\frac{S}{N} = \frac{3}{2} \times 2^{2V}$

or, $\frac{S}{N} = 1.5 \times 2^{2V}$

Expressing in dB,

$$\left(\frac{S}{N}\right) dB = 10 \log_{10}\left(\frac{S}{N}\right) = 10 \log_{10} (1.5 \times 2^{2V})$$

$$= 10 \log_{10} (1.5) + 10 \log_{10} (2^{2V})$$

or $\left(\frac{S}{N}\right) dB = 1.76 + 2V \times 10 \times 0.3$

Hence,

$\therefore \left(\frac{S}{N}\right) dB$ in PCM $= \left(\frac{S}{N}\right) dB = 1.8 + 6V$ for sinusoidal signal.

7) The signal having bandwidth equal to 3.5 KHZ is sampled, quantized & coded by a PCM system. The coded signal is then transmitted over a transmission channel of supporting a transmission rate of 50 kbps. Find the maximum signal to noise ratio that can be obtained by this system. The input signal has peak to peak value of 4 volts & rms voltage value of 0.2 V.

Soln,

=> maximum frequency of the signal, $f_m = 3.5$ KHZ

sampling " , $f_s \geq 2f_m \geq 2 \times 3.5 \geq 7$ KHZ

we know, signaling rate is,

$$r \geq V \cdot f_s$$

$$50 \times 10^3 \geq r \times 7 \times 10^3$$

$$\therefore V \leq 7.142 \text{ bits} \cong 8 \text{ bits}.$$

The rms value of signal is 0.2 V, Normalized signal power is,

Normalized signal power, $P = \frac{(0.2)^2}{1}$

ie, $P = 0.04$ W

maximum signal to noise ratio is,

$$\frac{S}{N} = \frac{3P \cdot 2^{2V}}{x^2_{max}}$$

or $\left(\frac{S}{N}\right) = \frac{3 \times 0.04 \times 2^{2 \times 8}}{4} = 1996.08 \cong 33$ dB

Hence, $\left(\frac{S}{N}\right)_{max} = 33$ dB.

iv) **Probability of Error:**

It defines average probability of error that can occur in a communication system. Let, $P(0)$ be the probability of occurrence of '0' and $P(1)$ be the probability of occurrence of '1'. Then, the error is of two types.

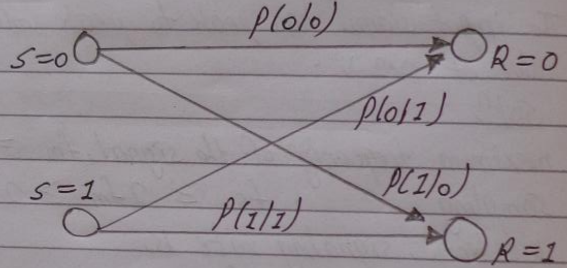
$P(0/1)$: ie, transmitted as '1' but received as '0'

$P(1/0)$: ie, " " '0' " " " '1'

The total probability of error, P_e is,

$$P_e = P(1/0) \cdot P(0) + P(0/1) \cdot P(1)$$

It is measured on the basis of following functions,



Error function:

$$erf(u) = \frac{2}{\sqrt{\pi}} \int_0^u \exp(-z^2)\, dz$$

Complementary error function,

$$erfc(u) = \frac{2}{\sqrt{\pi}} \int_u^\infty \exp(-z^2)\, dz$$

Q-function, $$Q(u) = \frac{1}{\sqrt{2\pi}} \int_u^\infty \exp\left(-\frac{x^2}{2}\right) dx$$

v) Binary data is transmitted over a microwave link at a rate of 10^6 bits/sec and the PSD of noise at the receiver input is

10^{-10} watts/HZ. Find the average carrier power required to maintain an average probability of error $Pe < 10^{-4}$ for coherent binary FSK. What will be the required channel Bandwidth?

Soln,

$\Rightarrow$ Bit rate $= 10^6$ bits/sec

$\frac{N_0}{2} = 10^{-10}$ W/HZ

or $N_0 = 2 \times 10^{-10}$ W/HZ

Also,

$Pe < 10^{-4}$,

$P_s = ?$

we know, error probability of FSK with coherent detection is,

$$Pe = \frac{1}{2} \text{erfc} \left[\frac{0.6\, E_b}{N_0} \right]^{1/2}$$

But, $E_b = P_s \cdot T_b$

$$\text{or } Pe = \frac{1}{2} \text{erfc} \left[0.6\, P_s \frac{T_b}{N_0} \right]^{1/2}$$

But, $T_b = \frac{1}{\text{bitrate}}$

$$\text{or } 10^{-4} = \frac{1}{2} \text{erfc} \left[\frac{0.6\, P_s}{2 \times 10^{-10} \times 10^6} \right]^{1/2}$$

or $2 \times 10^{-4} = \text{erfc}\, [3000\, P_s]^{0.5}$

But,

$1 - \text{erfc}(u) = \text{erf}(u)$

or $1 - 2 \times 10^{-4} = 1 - \text{erfc}\, [3000\, P_s]^{1/2}$

or $0.9988 = \text{erf}\, [3000\, P_s]^{0.5}$

we have,

$0.9998 = \text{erf}\, [2.8]$

or $3000\ P_s = 2.8$

Hence,

$\therefore\ P_s = \frac{2.8}{3000} = 0.933\ mW$

w) Binary data is transmitted at a rate of 10^6 bits/second over a chanel having a bandwidth of 3 MHz. Assume that the noise PSD at the receiver is $N_0/2, = 10^{-10}$ watt/Hz. Find the average carrier power required at the receiver input for coherent PSK and DPSK signalling schemes to maintain a probability of error $P_e = 10^{-4}$.

sol^n,

⇒ ① PSK system.

We know that the average error probability of FSK system,

$P_e = \frac{1}{2} \text{erfc} \sqrt{\frac{E}{N_0}}$

But, $P_e = 10^{-4}$ (Given)

Hence,

$10^{-4} = \frac{1}{2} \text{erfc} \sqrt{\frac{E}{N_0}}$

or $2 \times 10^{-4} = \frac{1}{2}\ \text{erfc} \sqrt{\frac{E}{N_0}}$

or, $1 - 2\times 10^{-4} = 1 - \text{erfc} \sqrt{\frac{E}{N_0}}$ $[\because \text{erf}(u) = 1 - \text{erfc}(u)]$

or, $0.9998 = \text{erfc} \sqrt{\frac{E}{N_0}}$

We have, $\text{erf}\ (2.6) \cong 0.9998$

Thus, $\sqrt{\frac{E}{N_0}} = 2.6$

or $\frac{E}{N_o} = 6.76$

Therefore, $E = 6.76\ N_o$

But, $N_o = 2 \times 10^{-10}$

Therefore, $E = 6.76 \times 2 \times 10^{-10}$

$= 1.352 \times 10^{-9}$ J

But, $E = PT$

and, $T = \frac{1}{\text{bit rate}} = \frac{1}{10^6}$

Hence, $P = \frac{E}{T} = 1.325 \times 10^{-9} \times 10^6 = 1.352$ mW.

(ii) DPSK system:

$P_e = \frac{1}{2} e^{-(E_b/N_o)}$

or $10^{-4} = \frac{1}{2} e^{-(E_b/N_o)}$

Thus,

$\frac{-N_b}{N_o} = -8.5171$

or $\frac{E_b}{N_o} = 8.5171$

or $E_b = 8.5171\ N_o$

$= 8.5171 \times 2 \times 10^{-10}$

$= 1.7 \times 10^{-9}$ J

Hence, power $= P = \frac{E_b}{T_b} = 1.7 \times 10^{-9} \times 10^{+6}$

$\therefore P = 1.7$ mW

X. Explain briefly the generation of BPSK with mathematical expression and waveforms used. 2016/s

⟹ BPSK signal can be generated by applying carrier signal to a balanced modulator. The binary data signal is converted to a NRZ bipolar signal by an NRZ encoder. Here, the bipolar signal $b(t)$ is applied as a modulating signal to the Balanced modulator.

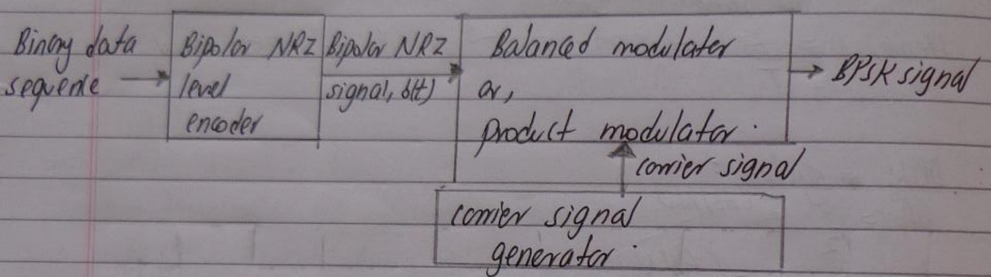


Fig: Generation of BPSK.

Input digital signal	Bipolar NRZ signal $b(t)$	BPSK output
0	$b(t) = -1$	$-\sqrt{2P}\cos\omega_c t$
1	$b(t) = +1$	$+\sqrt{2P}\cos\omega_c t$

$P = \frac{E_b}{T_b}$, E_b = signal energy , T_b = bit duration.

Bandwidth = $2f_b$.

In a coherent BPSK system, the pair of signals $s_1(t)$ and $s_2(t)$ are used to represent binary symbol 1 and 0 as,

$s_1(t) = \sqrt{\frac{2E_b}{T_b}}\cos 2\pi f_c t$ --- for symbol 1

$$s_2(t) = \sqrt{\frac{2E_b}{T_b}} \cos(2\pi f_c t + \pi) = -\sqrt{\frac{2E_b}{T_b}} \cos 2\pi f_c t \quad \text{for '0'}$$

In case of BPSK, there is only one basic function of unit energy which is,

$$\phi_1(t) = \sqrt{\frac{2}{T_b}} \cos 2\pi f_c t \quad , \quad 0 \le t \le T_b$$

Hence,

$$s_1(t) = \sqrt{E_b}\ \phi_1(t) \quad 0 \le t \le T_b \text{ for symbol } 1.$$
$$s_2(t) = -\sqrt{E_b}\ \phi_1(t) \quad 0 \le t \le T_b \text{ for " } 0.$$

The phase changes occurs in the carrier only at zero crossing. The BPSK waveform has full cycles of sinusoidal carrier.

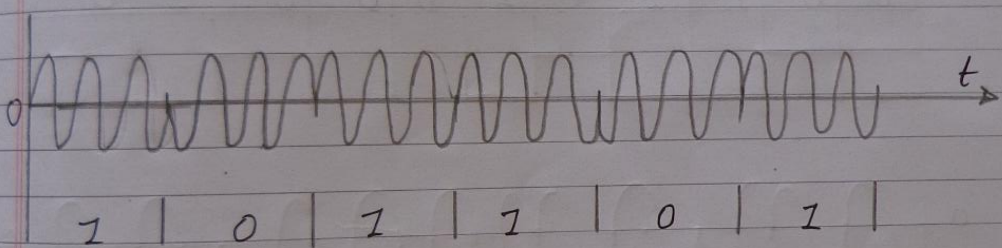


Fig: BPSK waveform

Y) Explain DPSK system. Encode binary sequence 0011010001 0 into differentially encoded sequence & draw DPSK waveform. 2015 fs

⇒ A DPSK system can be viewed as the non coherent version of the PSK. To send symbol 1, we phase advance the current signal waveform by 180° and to send symbol '0', we leave the phase of the current signal waveform unchanged.

dk is generated by using the logical eq^n,

$d_k = d_{k-1} \oplus b_k$

where, b_k = input binary digit at time kT_b

d_{k-1} = previous value of differentially encoded digit.

input binary sequence	0	0	1	1	0	1	0	0	0	1	0
d_{k-1}	0	0	0	1	0	0	1	1	1	1	0
d_k	0	0	1	0	0	1	1	1	1	0	0

For symbol '0' to phase shift by 180° and symbol '1' to leave the phase, we take,

$$d_k = \overline{d_{k-1} \oplus b_k}$$

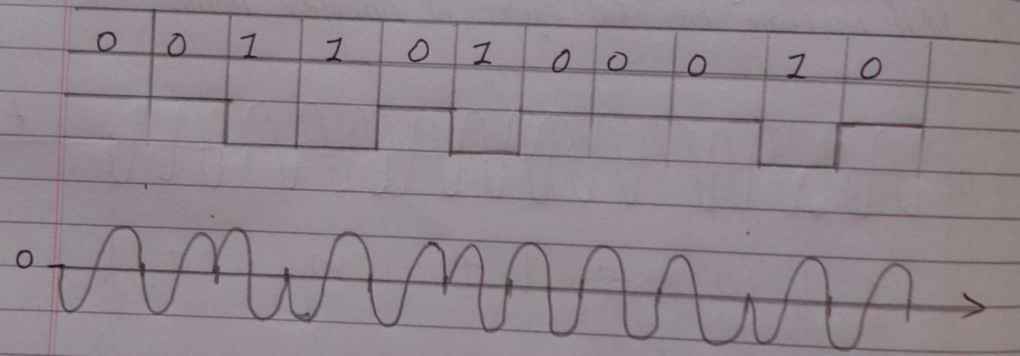


Fig: DPSK waveform.

4) Data Communication

1) A) Data communication refers to the transmission of digital data between two or more computers and a computer network or data network is a telecommunications network that allows computer to exchange data.

Electronic transmission of information that has been encoded digitally with some standard from one network to other network (systems) via certain medium is known as data communication.

B) Serial and parallel data transmission:

In telecommunication and data transmission, serial communication is the process of sending data one bit at a time, sequentially, over a communication channel or computer bus. This is in contrast to parallel communication, where several bits are sent as a whole, on a link with several parallel channels.

Serial communication is used for all long-haul communication and most computer networks, where the cost of cable & synchronization difficulties make parallel communication impractical.

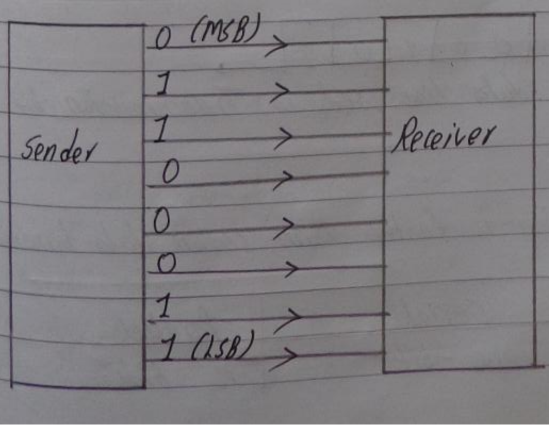


Fig: Parallel data transmission.

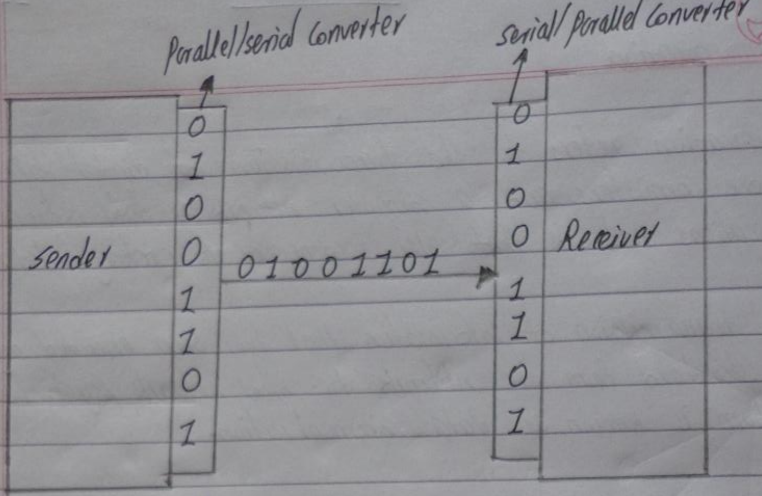
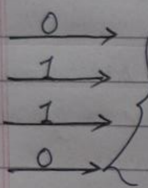
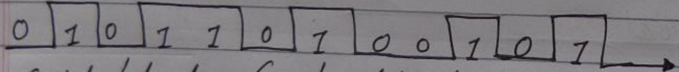


Fig: Serial transmission

In parallel transmission, all the bits of a byte are transmitted simultaneously on separate wires. It is a method of conveying multiple binary digits (bits) simultaneously



parallel data lines (multi parallel lines for data transmission, bits are sent parallel in the same time.)



serial data line (single data line used in Transmission bit by bit at a time.)

Parallel data transmission is faster than serial data transmission.

	Factor	Serial	Parallel
1	No. of bits transmitted	one -bit	n bits.

at one clock.		
(ii) No. of lines required to transmit n bits	one line	n lines.
(iii) speed of data transfer	slow	Fast
iv) Cost of transmission	low as one line is needed.	High as n lines are needed.
v) Data rate	slower.	Faster
vi cable	use less no. of wires.	use large no. of wires.
vii cable length	use long shield cables protected from EMI	can't use lengthy cables EMI limits data rate
viii Communication error.	only one bit is communicated at a time. hence, less chance of error.	Bits get corrupted due to capacitance effects between cable wires.
ix. Applications	Long distance communication	short distance communication like computer to printer.
x. Communication modes	can be Full duplex.	can be Half duplex.

c(i) Data Communication Topologies:

Network topologies may be physical or logical with respect their functionality. In general, physical topology relates to a core network whereas logical topology relates to basic network. It describes the layout or appearance of a network.
Eg: Star, Bus, ring, mesh and Hybrid.

Types :

① **Star Topology :**

Each computer is connected to a central device (hub) by a separate cable.

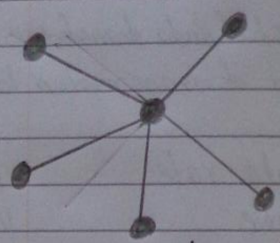


Fig : Star network

Advantages :

- Easy to add new stations
- Easy to monitor and troubleshoot.
- Can accomodate different wiring.

Disadvantages :

- Failure of Hub cripples attached stations.
- More cable required and more expensive.

⑪ **Bus - Network Topology :**

A Bus network is a network topology in which nodes are directly connected to a common linear or branched half duplex link called a Bus.

Advantages :

i) Easy to connect a computer or peripheral to a linear Bus.
ii) Requires less cable length than a star topology.

Disadvantages :

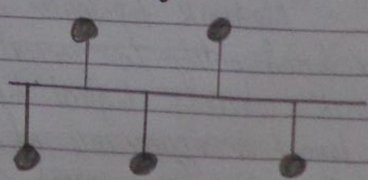


Fig : Bus

i) Entire network shuts down if there is a break in the main cable.

ii). Terminators are required at both ends of the backbone cable.

iii) Difficult to identify the problem if the entire network shuts down.

iv) Not meant to be used as a stand-alone solution in a large building.

(III) Ring network Topology:

It is a network topology in which each node connects to exactly two other nodes, forming a single continuous pathways for signals through each node. Data travels from node to node, with each node along the way handling every packet.

Rings are normally implemented using twisted pair or fiber-optic cable.

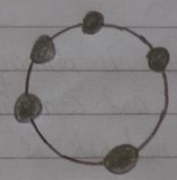


Fig: Ring network

Advantages:

- All stations have equal access.
- growth of system has minimal impact on performance.

Disadvantages:

- most expensive topology.
- Failure of one computer may impact others.
- Complex.

(IV) Tree topology:

A tree topology combines characteristics of linear bus and star topologies. It allows the expansion of an existing network and enables to configure a network to meet their needs.

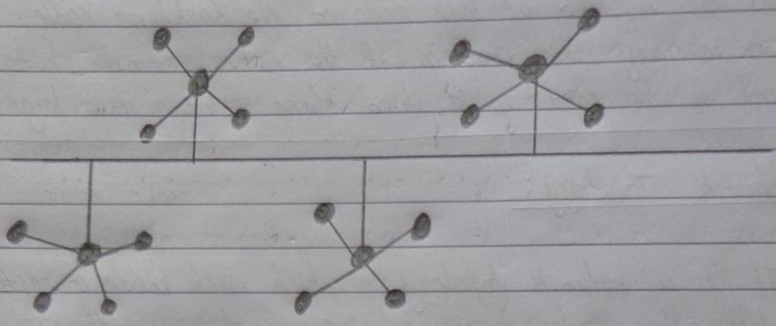


Fig: Tree topology.

Advantages:

- If one segment is damaged, other segments are not affected.
- error detection and correction is easy.
- expansion of network is possible and easy.

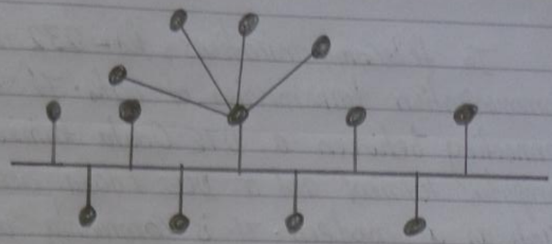
Disadvantages:

- scalability of the network depends on the type of cable used.
- maintenance becomes difficult.
- It relies heavily on main Bus cable, if it breaks whole network is cripped.

v) Hybrid Topology:

It is an integration of two or more different topologies to form a resultant topology which has many advantages of all constituent basic topologies rather than having characteristics of one specific topology.

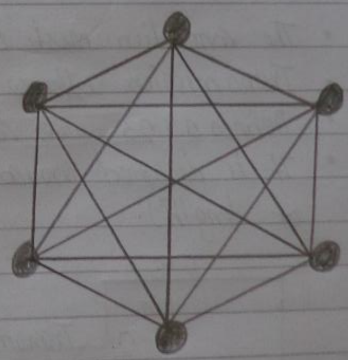
Advantages:
- Reliable
- scalable
- Flexible
- Effective



Disadvantages:
- complexity of design.
- costly Hub
- costly infrastructure

vi mesh Topology:
Each and every node of the network is interconnected.

Advantages:



- It has multiple lines, so if one route is blocked then other routes can be used for communication.
- performance is not affected with heavy load of data transmission.
- ensures data privacy or security
- expansion and modification can be done without disrupting other nodes.

Disadvantages:
- overall cost is high compared to other.
- setup and maintenance is very difficult.
- leads to redundancy of many of the network connections.

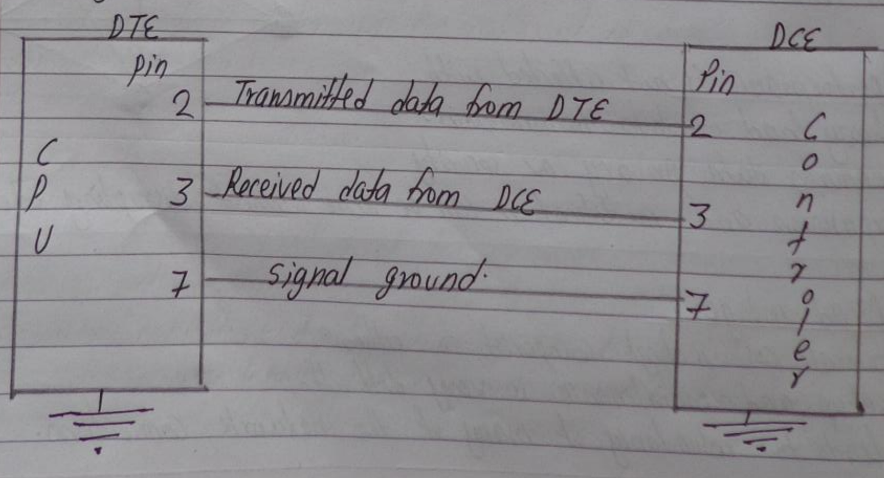
D. Serial interface – Rs –232 :

In telecommunications, Rs–232 is a standard for serial communication transmission of data. It formally defines the signal connecting between a DTE (data terminal equipment) such as a computer terminal and a DCE (data circuit terminating equipment) such as a modem. It is commonly used in computer serial ports.

The standard defines the electrical characteristics and timing of signals, the meaning of signals and the physical size and pinout of connectors.

Rs–232 is a standard by which two serial devices communicate:

- The connection must be no longer than 50 feet
- Transmission voltages are –15 V and + 15 V.
- Defines a 25 wire cable with a DB 25 s/95 connector.
- It is designed around transmission of characters (of 7 bits of length).



It is a 25-pin connector, each pin has its own functions:

Pin 1 : protective ground
pin 2 : Transmit data
pin 3 : Receive data.
pin 4 : Request to send
pin 5 : clear to send
pin 20 : data terminal ready.
pin 10 : Test pin
pin 12 : data carrier detect
pin 15 : Transmit clock
pin 17 : Receive "
pin 24 : external "
pin 16 : Receive data.

Disadvantages:

- Increases power consumption.
- 25-way connector, too large.
- Unbalanced transmission.
- Complicates power supply design.
- Low speed for long distance.
- Requires transceiver chips which add to system cost.

E) Parallel Interface: 2015/F

- Transfers data between two devices at eight or more bits at a time.
- Also referred to as serial by word transmission.
- Eg: Centronics parallel interface.

Generally, parallel needs eight wires, one for each bit of the data so that whole 8 bits are send at the same time. The interface utilizing 25 pin connector and is extra of a printer version.

serial interface needs only one wire and each of eight bits is send one at a time. on a pc, parallel interface is frequently called printer port and was initially for a centronics printer. A serial interface of a Pc was initially RS-232 for connection modems.

A parallel interface provides a multiline data channel in which bits are sent across multiple conductors simultaneously. The bits must stay in synchronization as they cross the wire, so the parallel interfaces are limited in distance.

Several technologies implement in parallel interfaces are :

(i) HIPPI (High performance parallel interface)
(ii) SCSI (small computer system ")
(iii) PCI Bus.
(iv) OFDM (orthogonal frequency division multiplexing).

F) Telephone network :

- uses circuit switching
- Telephone network had its beginnings in late 1800s
- The entire network was originally an analog system using analog signals to transmit voices.

- Telephone network is made up of three major components : local loops, trunks and switching offices.

Local loops :

- A twisted pair cable that connects the subscriber telephone to the nearest end office or local central office.
- The local loop, when used for voice, has a bandwidth of 4 KHZ.

Trunks :

- Transmission is usually through optical fibres or satellite links.
- Handles hundreds or thousands of connection through multiplexing.
- A transmission media that handle the communication between offices.

Switching offices :

- Connects several local loops or trunks.

G) Direct distance Dialing (DDD) :

- DDD is a telecommunication term for a network provided service feature in which a call originator may, without operator assistance, call originator may, without operator assistance, call any other user outside the local calling area or extended service area.

- Also known as national wide dialing.

- Successful operation of the DDD program depends primarily on three factors :
  - A standard nationwide numbering plan.
  - A method of charging the customer.

- A fundamental plan for automatic toll switching.

- DDD also extends beyond the boundaries of national public telephone network, in which case it is called international direct distance dialing (IDDD).

H) Dedicated line service :

- A point to point, hardwire connection between two service locations.

- shared resources: the telephone network or the internet.
- In computer network and telecommunications, a dedicated line is a communications cable or other facility dedicated to a specific applications, in contrast with a shared resources such as a telephone network or the internet.

I. Data modems: 2014/S

A modem (modulator - demodulator) is a network hardware device that modulates one or more carrier wave signals to encode digital information for transmission and demodulates signals to decode the transmitted information. The goal is to produce a signal that can be transmitted easily and decoded to reproduce the original digital data.

Need for modem :

To interface computers, computer networks and other digital terminal equipment with analog communication lines and radio channel.

Types of modems :
(i) Half Duplex
(ii) Full Duplex
(iii) 2 wire modem
(iv) asynchronous and synchronous modems.

Synchronous modems :
- Clocking information is recovered at the receiver
- use PSK or QAM modulation technique.
- Used for mostly medium and high speed applications.

Asynchronous modems :
- No clocking information is sent.
- Mostly use ASK / FSK
- Restricted to use for low speed applications.

eg: Bell systems 202 T/S modems, uses FSK
- 202 T - full duplex, four wire operation
- 202 S - half " , " " "

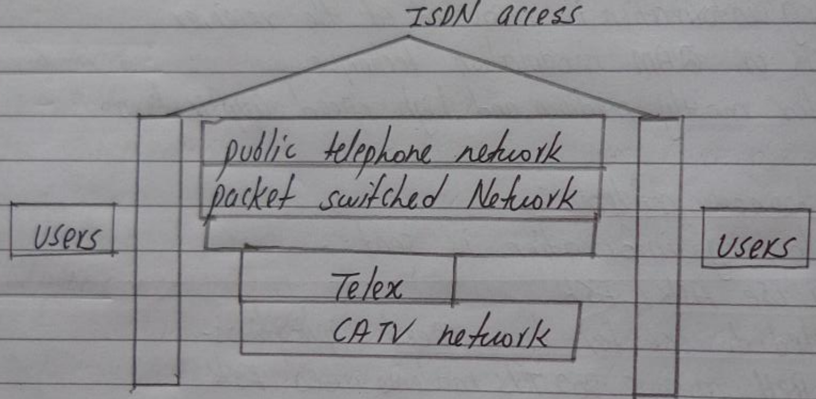
J. ISDN : 2015/F, 2016/F, 2014/S

Integrated service digital Network (ISDN) is a set of communication standards for simultaneous digital transmission of voice, video, data and other network services over the traditional circuits of the public switched telephone network.

ISDN is an all digital communications line that allows for the transmission of voice, data, video and graphics at very high speeds, over standard communication lines.

- ISDN provides a single, common interface with which to access digital communications services that are required by varying devices, while remaining transparent to the user.

- ISDN is not restricted to public telephone network alone; it may be transmitted via packet switched networks, telex, CATV network etc.



- It offers circuit switched connections (for either voice or data) and packet-switched connections (for data) in increments of 64 kilobits/s.

- With ISDN, voice and data are carried by bearer channels (B channel) occupying a bandwidth of 64 kb/s. Some switches limit B channels to a capacity of 56 kb/s.

- A data channel (D channel) handles signalling at 16 kb/s or 64 kb/s, depending on the service type.

- Two types:
  - BRI (Basic Rate interface)
    2B + D, 128 Kbps, for home uses
  - PRI (primary rate interface)
    23B + D channels, 1.544 Mbps for large business.

Advantages of ISDN:

- speed:
  BRI ISDN, using a channel aggregation protocol such as Bonding or multilink – PPP, supports an uncompressed data transfer speed of 128 kb/s, plus bandwidth for overhead and signalling. PRI transfers at an even higher speed upto 1920 kb/s.

- multiple devices:
  ISDN allows multiple devices to share a single line. It is possible to combine many different digital data sources and have the information routed to the proper destination.

- Signalling / Fast call setup:
  Istead of the phone company sending a ring voltage signal to ring the bell in phone, it sends a digital packet on a separate channel (out of band signal).

K. Packet Format:

Most network packets are split into three parts;

- Header
- payload

- Trailer

- Header :
  - Contains instructions about the data carried by the packet
  - length of packet
  - synchronization
  - packet number
  - protocol
  - originating address and destination address

- Payload :
  - Body or data of a packet
  - If a packet is fixed length, then the payload may be padded with blank information to make it right size.

- Trailer :
  - sometimes called footer, typically contains a couples of bits that tell the receiving device that it has reached the end of the packet.
  - It may also have some types of error checking (CRC)

1) Data communication protocol: 2016/F, 2016/S

In telecommunication, a communications protocol is a system of digital rules for data exchange within or between computers.

- Communicating systems use well defined formats for exchanging messages.

- Some network protocols :
  - open systems interconnections (OSI)
  - File transfer protocol (FTP)
  - Transmission control protocol / internet protocol (TCP/IP)

Basic requirements of protocols :

In general much of the following should be addressed.

- Data Formats for data exchange.
- Address " " " "
- Address mapping
- Routing
- Detection of transmission errors.
- Acknowledgements
- loss of information timeouts and retries
- Direction of information flow
- sequence control
- Flow control

M) **Ethernet :**

Ethernet is the name of the most commonly used LAN today. A Lan (local area network) is a network of computers that covers a small area like a room, office, a building. It is used in contrast with WAN (wide area network) which spans for much larger geographical areas. Ethernet is a network protocol that controls how data is transmitted over a LAN. Technically, it is referred to as the IEEE 802.3 protocol.

- most popular packet - switched LAN technology
- Bandwidths 10 Mbps, 100 Mbps, 1 Gbs
- max Bus length : 2500 m
  - 500 m segments with 4 repeaters

- Bus and star topologies are used to connect hosts.
  - Host attach to network via ethernet transceiver or hub or switch.
  - Hubs are used to faciliate shared connections.
  - All the hosts on an Ethernet are competing for access to the medium.

problem : Distributed algorithm that provides fair access.

- Ethernet uses MAC addresses that are 48 bits in length & expressed as twelve hexadecimal digits.

- on an ethernet network, when one device sends data it can open a communication pathway to the other device by using the destination MAC address.

  FF : FF : FF : FF : FF : FF (Broad cast).

- Ethernet standard defines both the layer 2 protocols and the layer 1 technologies.

- Types of Ethernet LANs :
  - 10 Base - T
  - Fast Ethernet
  - Gigabit "

- 10 Gbps ethernet
- wireless ethernet

• Ethernet uses baseband signalling.

N. Error detection and correction : 2017/F, 2015/S, 2015/F, 2016/F, 2014/S

Networks must be able to transfer data from one device to another with complete accuracy. While the transmission data can be corrupted, for reliable communication errors must be detected and corrected.

Types of errors :

• Single bit error :
  only one bit in the data unit has changed.

• Burst error :
  two or more bits in the data unit has changed.

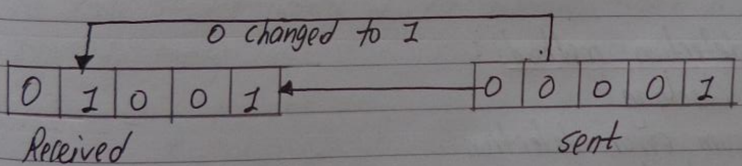


Fig: single bit error.

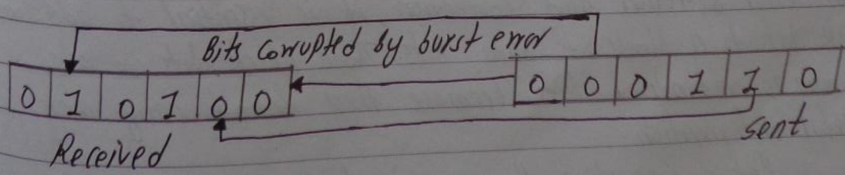
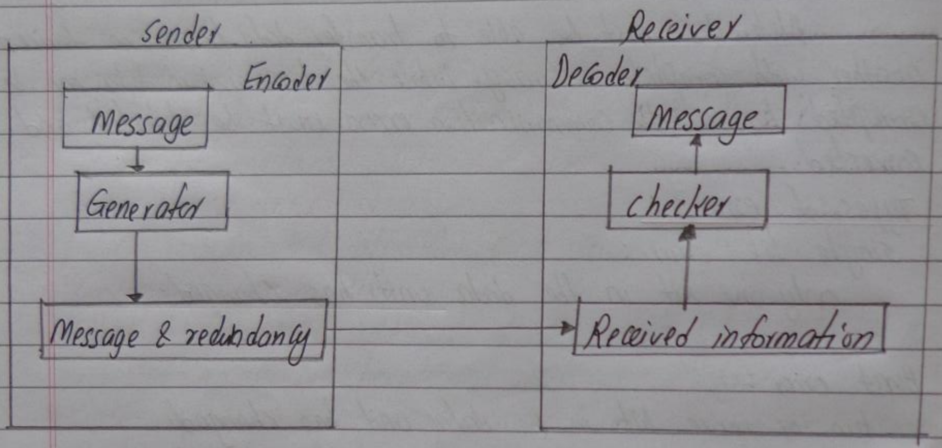


Fig: Burst error.

Error detection :

Error detecting code is to include only enough redundancy to allow the receiver to deduce than an error occurred, but not which error, and have it request a re-transmission.
Error detection uses the concept of redundancy, which means adding extra bits for detecting error at the destination.



Error detection methods :

(I) parity checking
(II) check sum error detection
(III) cyclic redundancy check (CRC)

For error detection and correction, it is essential to add some check bits to a block of data bits. These check bits are also known as redundant bits because they donot carry any useful information.
Before correcting the error, introduced in data bits, it is essential to first detect them.

① parity checking:

In this method, an additional bit called as parity bit is added to each data word. This additional bit is so chosen that the resultant word will have either an even parity or odd parity. If even parity is to be used then a parity bit is added to make the total number of 1 bits even. similarly, for odd parity the total number of 1 bits is made odd by adding the parity bit.

P	data word						
0	1	0	0	1	0	1	1

→ even parity as total number of 1 bit's in the data word is 4 ie, even. hence, parity bit P = 0

P	data word						
1	0	0	1	0	0	1	1

→ even parity. Total number of 1 bit's in the data word is 3 originally. Add a parity bit P of 1 to make the total number of 1 bits to be 4 ie, even.

error detection:

The parity checking at the receiver can detect the presence of an error if the parity of the received signal is different from the expected parity.

	P	message bits						Parity	Receiver's decision
Transmitted code:	0	1	0	0	1	0	1	even	Correct word
Received code with one error	0	0	0	0	1	0	1	odd	Correct word
Received code with three errors	0	0	1	0	0	0	1	odd	Correct word

error / errors

The receiver detects the presence of error if the number of error is odd ie, 1, 3, 5, 7 . . .

Failure of parity checking methods:

If the number of errors introduced in the transmitted code is two or any even number then the parity of the received code word will not change. It will remain same ie, even and the receiver will fail to detect the presence of errors.

Limitation of parity checking:

(i) It is not suitable for detection of multiple errors (2, 4, 6 etc)

(ii) It cannot reveal the location of errorneous bit. It cannot correct the error either.

(II) Checksum Error detection:

The errors usually occur in bursts. The parity check method is not useful in detecting the errors under such conditions. The check sum error detection method can be used successfully in detecting such errors. A checksum is transmitted along with every block of data types. The carries of the MSB are igonered while finding out the checksum byte.

The advantage of this method over the simple parity checking method is that the data bits are mixed up due to the 8-bit addition. Hence, checksum represents the overall data block. In checksum, hence, there is 255 to 1 chance of detecting random errors.

(III) Cyclic Redundancy check (CRC):

- CRC error detection method treats packet of data to be transmitted as a large polynomial.

- Transmitter:
  Using polynomial arithemetic, divides polynomial by a given generating polynomial.

- Quotient is discarded.
  Remainder is attached to the end of message.

- message with the remainder is transmitted to the receiver.
- Receiver divides the message and remainder by same generating polynomial.

- If a remainder not equal to zero results → error during transmission.
- " " " Of zero results → error during transmission.

quotient → not used

divisor) dividend → dividend : Content of frame

divisor → generator polynomial

Remainder → fixed size used for checksum

The CRC is calculated by performing a modulo 2 division of the data by a generator polynomial and recording the remainder after division.

Eg: A Codeword is received as 1100 1001 01011. Check whether there are errors in the received codeword, if the divisor is 10101. (The divisor corresponds to the generator polynomial).

⇒ Data word = 1100100101011
Divisor = 10101

```
              1 1 1 1 1 0 0 0 0 1
Divisor  10101 ) 1 1 0 0 1 0 0 1 0 1 0 1 1   ← Received codeword
              ⊕ 1 0 1 0 1
                0 1 1 0 0 0
              ⊕  1 0 1 0 1
                 0 1 1 1 1 1
mod-2         ⊕   1 0 1 0 1
additions        0 1 0 1 0 0
              ⊕    1 0 1 0 1
                   0 0 0 0 1 1 0 1 1
                           1 0 1 0 1
                           0 1 1 1 0   ← Remainder
```

The non-zero remainder shows that there are errors in the received codeword.

CRC cannot detect all types of errors. The probability of error detection and the types of detectable errors depend on the choice of divisor.

Hamming codes :

Hamming codes are linear block codes. The family (n,k) Hamming codes for $q \geq 3$ is defined by,

- Block diagram : $n = 2^q - 1$
- no. of message bits : $k = 2^q - 1 - q$
- no. of parity bits : $(n-k) = q$

  where $q \geq 3$, ie, minimum number of parity bit is 3
- minimum distance, $d_{min} = 3$

  code rate or code efficiency $= \frac{k}{n} = \frac{2^q - q - 1}{2^q - 1} = 1 - \frac{q}{2^q - 1}$

  if $q >> 1$, then code rate, $r \cong 1$

← code word length, $n = 2^q - 1$ →

$k = 2^q - 1 - q$	$q = (n-k)$

← message bits → ← parity bits →

Fig: code word structure of Hamming code.

Error detection and correction capabilities of Hamming codes:

- Number of errors that can be detected per word = 2

  minimum distance, $d_{min} = 3$

- no. of errors that can be corrected per word = 1

  since, $d_{min} \geq (2t+1)$, therefore, $3 \geq (2t+1)$ or $t \leq 1$

  Thus, with $d_{min} = 3$, it is possible to detect upto 2 errors and it is possible to correct upto only 1 error.

Q. What are the advantages and disadvantages of ISDN? What benefits does it offer to user, network provider & manufacturer? 2016F, 2014S

⇒ Advantages:

i) The basic advantage of ISDN is to facilitate the user with multiple digital channels. These channels can operate concurrently through the same one copper wire pair.

ii) ISDN takes only two second to launch a connection while other modem takes 30 to 60 seconds for establishment.

iii) The digital signals broadcasting transversely the telephone lines.

iv) It provides high data rate because of digital scheme which is 56 kbps.

v) ISDN network lines are able to switch manifold devices on the single line such as faxes, computers, cash registers and many other devices. These all devices can work together and directly be connected to a single line.

Disadvantages:

i) very costly than the other typical telephone system.

ii) ISDN requires specialized digital devices.

Typical ISDN applications:

* Internet access and on-line service
* Tele commuting
* Remote office routing
* Video - Conferencing
* Disaster Recovery
* PC-to-PC screen sharing and collaboration.

What makes ISDN unique is that each B channel is a separate communication circuit. That means that just one ISDN line can support simultaneous two-way communication for two devices, such as a computer and a telephone or a computer and a video camera for teleconferencing.

P) What services does the PTN offers? Define the terms : instrument, Dial switch, local loop and Trunk circuit. 2016/s

=> PTN is an integrated transport network technology featuring both packet and transmission. It has become a mainstream solution for IP based mobile backhaul networks in 3G/LTE system due to several key technologies.

- Local loop :

In telephony, a local loop is the wired connection from a telephone company's end office to customer's houses or small business. It is also referred as "last mile". The connection is usually of twisted pair copper wire.

The local loop or local tail or subscriber line is the physical link or circuit that connects from the demarcation point of the customer premises to the edge of a common carrier or telecommunications services provider's network.

- Trunk circuit :

A trunk is a communication line or link designed to carry multiple signals simultaneously to provide network access between two points. It typically connect switching centers in a communications systems.

An interconnecting transmission channel between a switching machine in one location and a switching machine in an

154

adjacent node.

- Dial switch:

A disc on a telephone that is rotated a fixed distance for each number called.

- Instrument:

Devices that communicate, denotes, detects, indicates, measures, observes, records or signals a quantity or phenomenon or controls or manipulates other device.

Q. Why are synchronous modems required for medium and high speed applications? 2016/S

⟹ Synchronous modems use timing to determine where data begins and ends. Uses periodic synchronization bits to synchronize modems. It is faster than asynchronous modems and provides function such as error checking. It uses PSK and QAM modulation techniques.

– Used for mostly medium and high speed applications upto 57.6 kbps

For medium speed:

- QPSK for 2.4 kbps (eg: Bell system 201 C)
- 8-PSK for 4.8 kbps (eg: " " 208 A)
- Both are duplex (full), 4 wire systems.

For High speed:

- 16 QAM - for a 9.6 kbps (eg: Bell system 209 A)

- Full duplex, four wire transmission.

Since the synchronous modem are used for medium and high speed modems, these modems contain the following additional hardware:

- Clock recovery
- equalizers
- Scramblers and descrambler circuit.

Q) What do you mean by packet switching? Explain the packet data transmission with an example of ethernet packet format. 2015 fs

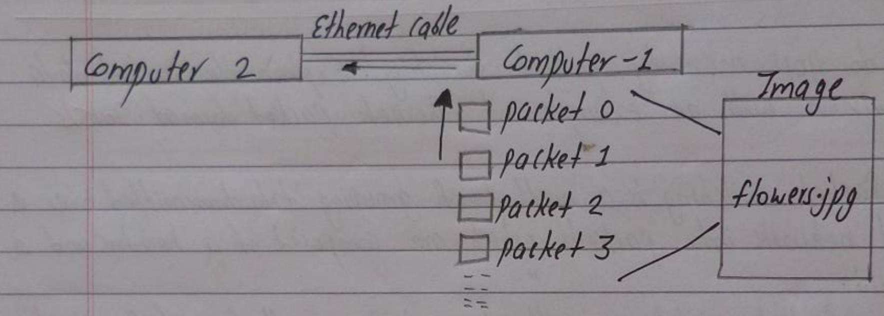
⟹ Packet switching is a method of grouping data transmitted over a digital network into packets which are composed of a header and a payload.

Packet switching is the transmission method used for most computer network because the data transported by these networks is fundamentally burstly in character and can tolerate latency (due to lost or dropped packets). The transmission bandwidth needed varies greatly in time, from relatively low traffic because of background services such as name resolution services, to periods of high bandwidth usage during activities such as file transfer. The Internet is the prime example of a packet switched network based on the TCP/IP protocol suite.

Packets – Data from Here to There:

- Eg: To send an image file between ethernet connected computers
- This is 'one hop' LAN case
- eg: 50 KB image.jpg
- 50,000 bytes

- How to send the image.jpg on the wire?
- Use packets
- Divide bytes of image.jpg into packets
- say each packet is 1500 bytes.
- Then image.jpg divides into about 32 packets.
- Network transmits one packet at a time.



For transmission, the 50 KB of the image is divided into packets. The packet is the natural unit of transmission within networking.

By sending a large file in several small chunks over a network, packet switching minimizes the impacts of data transmission errors. Statistical multiplexing is used to enable devices to share these circuits.

5) Explain serial and parallel interface in data communication.
2015/F

⇒

	serial interface	parallel interface.
①	Data is transmitted bit after bit in a single line.	Data is transmitted simultaneously through group of lines (Bus).

157

(ii)	Data congestion takes place.	No data congestion.
(iii)	serial interface consists of an I2C bus, SPI bus or synchronous serial control and data lines.	Parallel interface consists of 8 data pins and 3 control lines. The control lines are Enable, register select and Read/write.
iv)	Low speed transmission.	High speed transmission.
v)	Implementation of serial links is not an easy task.	parallel data links are easily implemented in Hardware.
vi)	Bandwidth of serial wire is much higher.	lower.
vii)	less expensive	more expensive.
viii)	serial interface is more flexible to upgrade, without changing the Hardware.	parallel data transfer mechanism rely on hardware resources and hence not flexible to upgrades.
ix.	serial Communication work effectively even at high frequencies.	Parallel buses are hard to run at high frequencies.
x.	Uses serial data transmission.	Uses parallel data transmission.

5 Hours

5) optical Fiber Communication

2015/F

1)A Fiber optic communication is a method of transmitting information from one place to another by sending pulses of light through an optical fibre. The light forms an electromagnetic carrier wave that is modulated to carry information.

An optical fibre is a dielectric wave guide that operates at optical frequencies. This fibre wave guide is normally cylindrical in form.

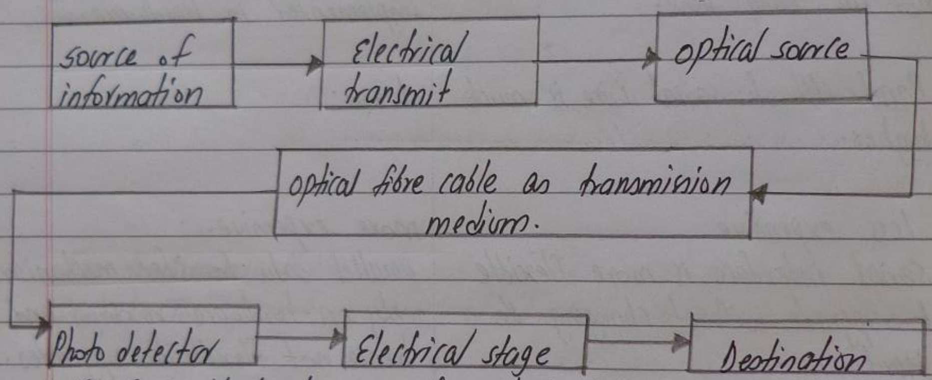


Fig: Basic block diagram of optical fiber communication.

Applications:

(i) As fibres are very flexible, they are used in flexible digital cameras.
(ii) used in mechanical imaging i.e. for inspection of mechanical welds in pipes and engines of rockets, space shuttles, etc.
(iii) medical imaging such as endoscope and laparoscopes.
(iv) used under sea communication.
v) used in military applications such as ships, aircrafts, tanks.

vi) Nuclear testing applications use optical fiber phase sensors and transducers.

vii) used in public utility organizations like TV transmission, railways.
viii) used in LAN systems of offices, industrial plants, colleges.
ix) used in telecommunications such as voice telephones, video phones, data networks, message services.

Advantages :

① Enormous potential Bandwidth :
The optical carrier frequency range yields the far greater potential transmission bandwidth than metallic cable system.

② Small size and less weight :
It has very small diameters which are often no greater than the diameter of a human hair. Hence, even when such fibres are covered with protective coatings, they are far smaller & much lighter than corresponding copper cables.

③ Electrical isolation :
optical fibers which are isolated from glass or sometimes a plastic polymer are electrical insulators and hence, unlike their metallic counterparts, they donot exhibit earth loop and interface problems.

iv) Signal security :
Signal sniffing cannot be carried out until physical insertion of detecting instrument, which bit difficult. widely applicable in

military, danking & other secure networking.

v) Low Transmission loss:
- efforts are made in the field of fiber development.
- glass fiber cable is efficient than plastic.

vi) Ruggedness and Flexibility:
- Flexibility in uses, manufactured for high tensile strength.
- can be bent till certain diameter w/o prominent loss.

vii) low cost and availability:
Since the fibers are made of silica which is available in abundance, hence there is no shortage of material & optical fibers offer the potential for low cost communication.

viii) Reliability:
Quality is not affected by external radiations. Further, due to its negligible attenuation and dispersion, optical fiber communication has high reliability. All the above factors also tend to reduce the expenditure on its maintenance.

Disadvantages:

(i) only economical when the bandwidth is fully utilized.
(ii) High cost of installation.
iii) Need more expensive optical transmitter and receivers.
iv) more difficult and expensive to splice than wires.
v) Tapping is not possible. specialized equipment is needed to tap a fiber.

vi) Communication is not totally in optical domain, so repeated electric optical-electrical conversion is needed.

vii) The splicing and testing equipments are very expensive as compared to copper equipments.

viii) It requires more protection around the cable compared to copper.

ix) Fibers can be broken or have transmission losses when wrapped around curves of only a few cm radius.

x. Transmission on optical fiber requires repeating at distance intervals.

B) Ray theory for optical fiber communication:

The simplest way to view light in fiber optics is by ray theory. In this theory, the light is treated as a simple ray, shown by a line. An arrow on the line shows the direction of propagation.
speed of light in vacuum = 300000 km/s
However, speed of light in medium is more slowly, $v = c/\mu$
The ratio of the velocity of the light, c, in vacuum to the velocity of light in the medium, v, is the refractive index, μ.

$$\mu = \frac{c}{v}$$

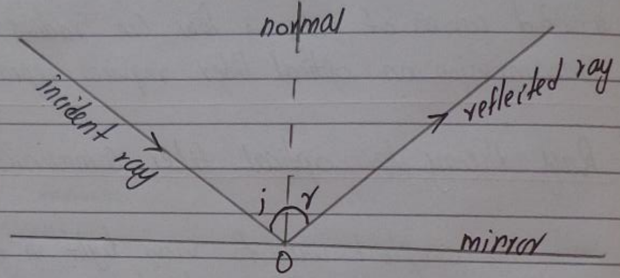
The optical energy in the form of wave, follows narrows paths called rays. Rays are used to describe optical effects geometrically, ray theory is called geometrical optics. When a ray is incident on the interface between two dielectrics of differing refractive indices (eg: glass-air), refraction occurs.

Light rays in homogenous media are straight lines.

Laws of Reflection:

Reflection from a mirror or at the boundary between two media of different refractive index ; angle of reflection equals to the angle of incidence, $\theta_r = \theta_i$

The law of reflection states that the incident ray, the normal to the surface of the mirror and the reflected ray all lie in the same plane.



Snell's law of Refraction:

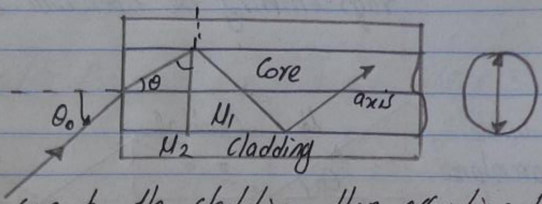
It states that the ratio of the sines of angle of incidence and refraction is equivalent to the ratio of phase velocities in the two media or equivalent to the reciprocal of the ratio of the indices of refraction.

$$\frac{\sin\theta_1}{\sin\theta_2} = \frac{v_1}{v_2} = \frac{\lambda_1}{\lambda_2} = \frac{\mu_2}{\mu_1}$$

with each θ as the angle measured from the normal of the boundary. v as the velocity of light in respective medium.

The angle at which total internal reflection occurs is called the critical angle of incidence. Ray theory gives an intuitive feel for light containment and light pulse spreading. It is accurate for multimode fibres.

For the meridional ray,



Snell's law at the fiber enter,
$\mu_a \sin\theta_0 = \mu_{core} \sin\theta$

If the ray is refracted from the core to the cladding, then according to Snell's law,

$$\sin\theta_i = \frac{\mu_{cladding}}{\mu_{core}} \cdot \sin\theta_r$$

$$\text{if } \sin\theta_i > \frac{\mu_{cladding}}{\mu_{core}} =$$

then, $\sin\theta_c \leq 1$, there is no tunneling from core to cladding.
Since, $\theta + \theta_i = 90°$, we have,

$$\sin\theta_0 = \mu_{core} \cdot \sin\theta$$
$$= \mu_{core} \cdot \cos\theta_i$$
$$= \mu_{core} \cdot \sqrt{1 - \sin^2\theta_i} < \sqrt{\mu_{core}^2 - \mu_{cladding}^2}$$

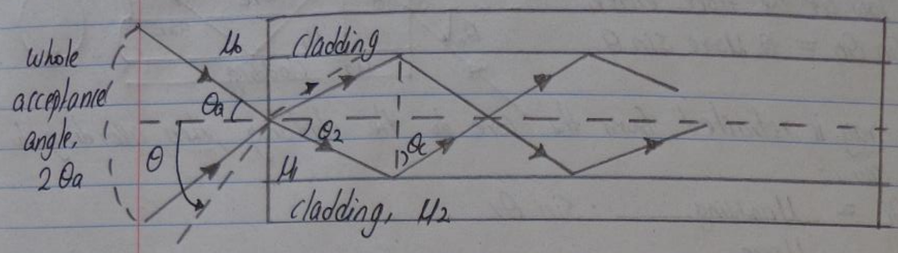
Hence,
Total internal reflection occurs if,

$$\therefore \sin\theta_0 < \sqrt{\mu_{core}^2 - \mu_{cladding}^2} = \mu_{core}\sqrt{1 - \left(\frac{\mu_{cladding}}{\mu_{core}}\right)^2}$$

c) Numerical Aperture and acceptance angle : 2015/S

The ray which passes through axis of the fiber core is called meridional ray.
Acceptance angle, θ_a is the maximum angle over which light

rays entering the fiber will be guided along its core.



The sine of that acceptable angle is called the numerical aperture and it is essentially determined by the refractive index contrast between core and cladding of the fiber, assuming that incident beam comes from air to vacuum.

Acceptance angle, $\theta a = \sin^{-1}\left(\sqrt{\mu_1^2 - \mu_2^2}\right)$
μ_1 = refractive index of glass fiber core
μ_2 = " " " quartz fiber cladding.

Numerical aperture (NA) is used to describe the light gathering or light collecting ability of an optical fiber. In optics, the numerical aperture (NA) of an optical system is a dimensionless number that characterizes the range of angles over which the system can accept or emit light.

$\therefore$ $NA = \sin\theta a \left(\sqrt{\mu_1^2 - \mu_2^2}\right)$
$\therefore$ Full acceptance angle = $2\ \theta a$.

D. Fiber Types, properties and Applications :

Based on the refractive index profile, there are two types of fibers :

- step index fibers
- Graded " "

Based on the modes, there are two types of fibers

- single mode fibers
- multimode fibers.

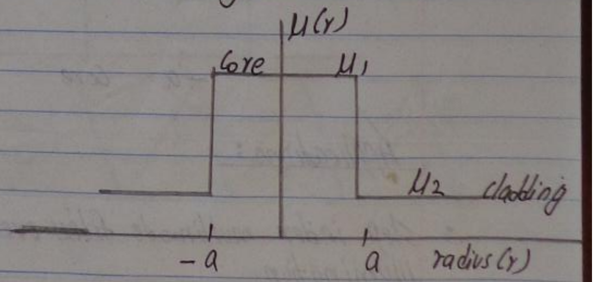
(I) step index fibers :

This is called so because the refractive index of the fibre step up as we move from the cladding to the core and this type of fibre allows single mode to propagate at a time due to very small diameter of its core.

$\mu_{cladding} < \mu_{core}$.

ie,

$$\mu(r) = \begin{cases} \mu_1 & \text{for } r < a \quad (\text{core}) \\ \mu_2 & \text{for } r \geq a \quad (\text{cladding}) \end{cases}$$



(II) Graded Index Fibers :

Graded index fibers donot have a constant refractive index in the core but a decreasing core index $\mu(r)$ with radial distance from a maximum value of μ_1 at the axis to a constant value μ_2 beyond the core radius 'a' in the cladding.

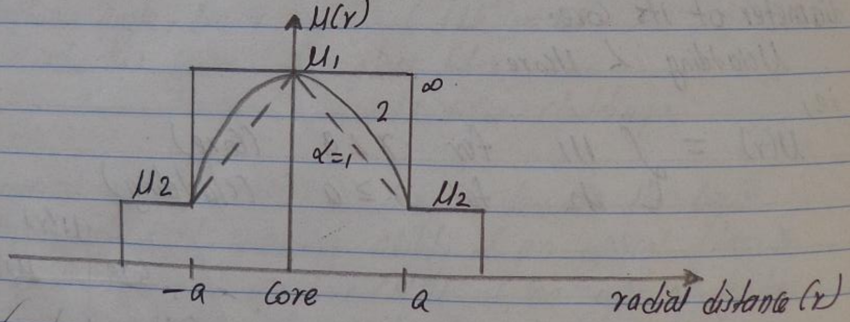
The index variation (r);

$$\mu(r) = \begin{cases} \mu_1 (1 - 2\Delta (r/a)^{\alpha})^{1/2} & r < a \text{ (core)} \\ \mu_1 (1 - 2\Delta)^{1/2} = \mu_2 & r \geq a \text{ (cladding)} \end{cases}$$

where,

Δ = relative refractive index difference.

α = profile parameter which gives characteristics refractive index profile of the fiber core.



Applications:

- Step index multimode fibers are mostly used for imaging and illumination.
- Graded index multimode fibers are used for data communication.

and network carrying signals for typically no more than a couple of km.

(iii) Single mode Fibers : 2014/s

- carries light pulses along single path. only the lowest order mode (fundamental mode) can propagate in the fiber and all higher order modes are under cut off condition (non-propagating).

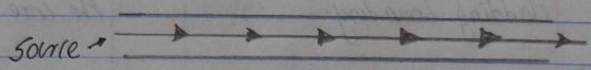
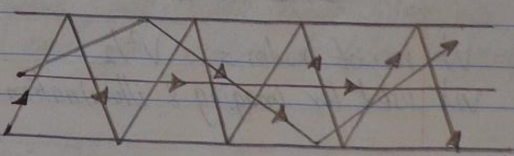


Fig: single mode.

- only one path is available
- core diameter is small , 8-10 μm
- No dispersion
- Higher Bandwidth
- Fabrication is costly and difficult.
- Used for long haul communication.
- supports one mode of propagation.
- optical source - LASER

(iv) multimode Fiber : 2014/s



- more than one path is available.
- High dispersion.
- Lower Bandwidth.
- Core diameter is higher.
- Fabrication is less difficult and not costly.

- used for short distance communication.

E)

Step index Fiber	Graded index Fiber
(i) The refractive index of the core is uniform throughout & undergoes an abrupt change at the core cladding boundary.	The refractive index of the core is made to vary gradually such that it is maximum at the center of the core.
(ii) Diameter of the core is about 50 – 200 µm in the case of multi-mode fiber & 10 µm in case of single mode fiber.	Diameter of the core is about 50 µm in the case of multi-mode fiber.
(iii) path of light propagation is zig-zag in manner.	path of light is helical in manner.
iv) Attenuation is more for multimode step index fiber but for single mode it is very less.	Attenuation is less.
v) no. of modes = $V^2/2$	no. of modes = $V^2/4$
vi) used for imaging & illumination.	used for data communication & networks over moderate distance.
vii) modal dispersion is found.	solves the problem of modal dispersion.

F). Single mode Fiber	multimode Fiber
(I) Core radius is small.	Core radius is large.
(II) supports one mode of propagation.	supports hundreds of modes.
III optical source - LASER.	optical source - LED.
iv. Launching of optical power into fiber is difficult as the core radius is small.	The launching of optical power into fiber is easier as the core radius is large.
v) Application: submarine cable system.	Application: Telephone links.
vi) cheaper	costly.
vii) Higher bandwidth.	limited bandwidth.
viii) Donot exhibit dispersion.	limited by modal dispersion.
ix. High cost connector.	low cost connectors.

G. <u>Mode propagation in SI and GI multimode fibers:</u>

Mode is the one which describes the nature of propagation of electromagnetic waves in a wave guide. single mode fibers can propagate only the fundamental mode and multimode fibers can propagate hundreds of modes.

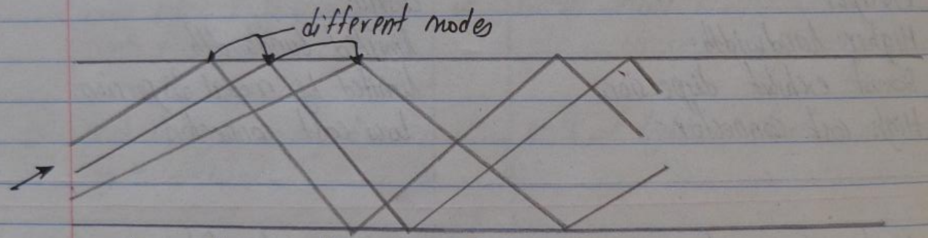
Propagation eq^n: $\Delta E = \mu_0 \epsilon_0 \mu^2 \frac{\partial^2 E}{\partial t^2}$

$= -\mu_0 \epsilon_0 \mu^2 \omega^2 E$

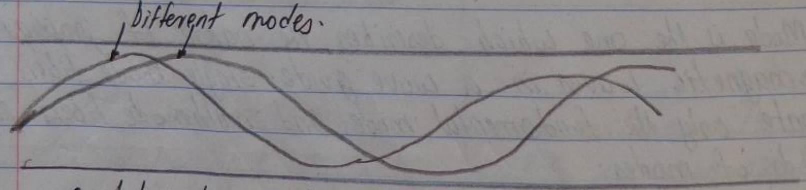
An index value, v, defined as the normalized frequency is used to determine how many different guided modes a fiber can support. In order to find a mode propagation constant and cut off frequencies of various modes of the optical fiber, V-number is most important and given by,

$$V = \frac{2\pi a}{\lambda}\sqrt{\mu_1^2 - \mu_2^2} = \frac{2\pi a}{\lambda} \cdot NA$$

a = radius of core
λ = optical free space wavelength.
μ_1, μ_2 = refractive indices of the core and cladding.



step index multimode Fiber.



graded index multimode Fiber.

H) Index profile of optical Fiber: 2014/S

The radial distribution of the fiber refractive index is called the index profile. In the case of a slab waveguide, the transverse refractive index is called the index profile. The index profile determines guiding properties of the fiber or slab waveguide. In general, the core region has a higher index than the cladding region. However, the index profile can have regions where the index is lower than the cladding value. Modern fiber or slab waveguide designs are based on index profile that assure proper operation within a range of wavelengths.

Constant index profile, $n(x) = \text{Constant}$

Linear " " , $n(x) = n(0) + x \cdot \frac{n(w) - n(0)}{w}$

Parabolic index profile :

$$n(x) = [n(w) - n(0)] \cdot \left(\frac{x}{w}\right)^2 + n(0)$$

where, $n(0), n(w)$ is the refractive index at $x = 0$ and $x = w$.

I. Attenuation, Dispersion, Bend loss in optical Fiber : 2015 S

Attenuation is defined as the ratio of optical output power to the input power in the fiber of length L.

$$\alpha = 10 \log_{10} \left(\frac{P_i}{P_o}\right)$$

P_i = input power

P_o = output "

α = attenuation constant

Attenuation means loss of light energy as the light pulse travels from one end of the cable to the other. Also known as signal or power loss. It decides the number of repeaters required between transmitter and receiver. Attenuation is directly proportional to the length of the cable. Attenuation in optical fiber take place due to elements like couplers, splices, connector and fiber itself. Modern fiber material is very pure, but there is still some attenuation.

Bending losses : Radiative loss :

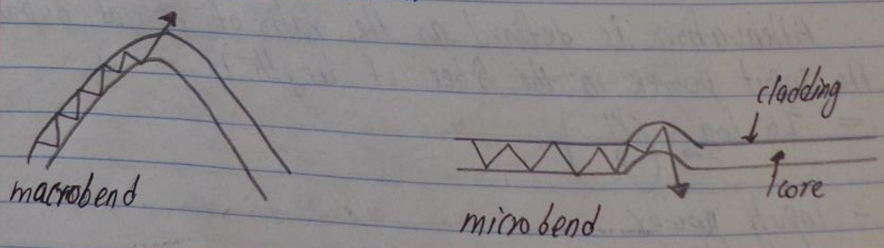
The loss which exists when an optical fiber undergoes bending is called bending losses. There are two types of bending:

(i) **macroscopic bending :**

Bending in which complete fiber undergoes bends which causes certain modes not to be reflected and hence, causes loss to the cladding.

(ii) **microscopic Bending :**

Either the core or cladding undergoes slight bends at its surfaces. It cause light to be reflected at angles when there is no further reflection.



Microbending is a much more critical feature and can be a major cause of cabling attenuation. Macro bending losses are called large curvature radiation losses.

Dispersion :

As a pulse travels down a fiber, dispersion causes pulse spreading. This limits the distance and the bit rate of data on an optical fiber. As an optical signal travels along the fiber, it becomes increasingly distorted. This distortion is a sequence of intermodal and intramodal dispersion.

Intermodal dispersion :

pulse broadening due to intermodal dispersion results from the propagation delay difference between modes within a multimode fiber.

Intramodal dispersion : /chromatic dispersion :

It is the pulse spreading that occurs within a single mode.

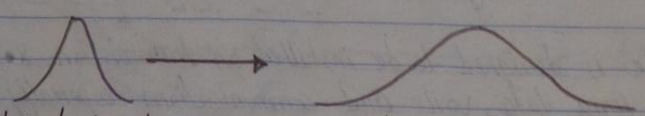
- material dispersion
- waveguide "

material dispersion : /spectral dispersion :

wavelength based effect caused by glass of which fiber is made.

waveguide dispersion :

occurs due to change in speed of wave propagating through waveguide.



input initial pulse — pulse due to dispersion.

2) A. Optical Ground wire (OPGW) system : 2017/F, 2016/F, 2014/S 2015/F

OPGW is an overhead grounding wire to effect grounding of overhead transmission lines, in which optical fibres are generated to provide communication systems functions. OPGW's enables long distance, high quality data transmission as well as video transmission without being affected by electromagnetic fields in any way, so that it is utilized as a transmission line for remote of unattended power plants and substation in addition to communications between power plants.

OPGW is intended for the transmission of communication. An OPGW is composed of different number of fibres, one is intended for a multiplexer, two pair for relay and on possible one for own LAN.

Properties :

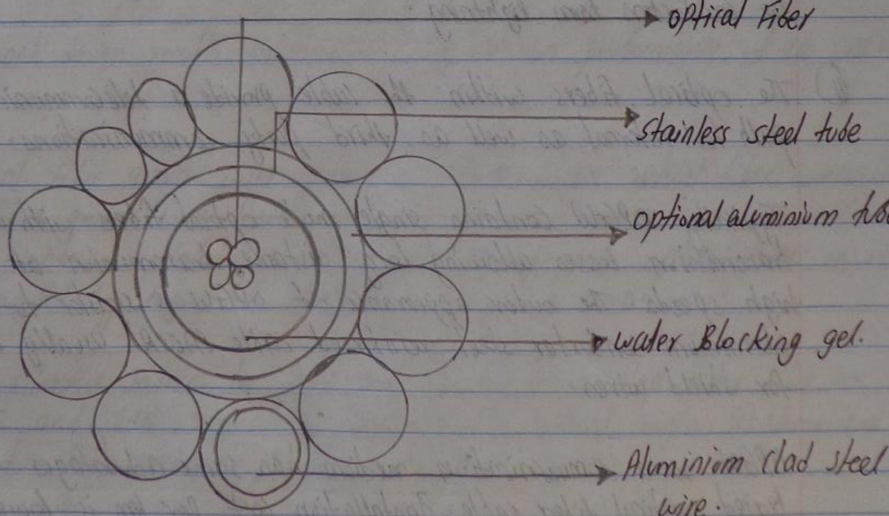
- It can be designed to achieve the desired tensile strength and sag characteristics.
- It can also be used for conducting fault currents & lightning currents without damage.

The cable is designed to be installed on transmission & distribution lines to carry data, voice and communications, especially in lightning waveform monitoring system, maintenance data information system, power line protection system, power line operation system.

Constructions : 2015/F, 2014/S

① Central loose tube type:

The fibers is placed loosely in a sealed and water resistant stainless steel tube filled with water blocking gel. This tube provides protection to the fiber during installation and operation under severe environmental conditions. Aluminium layer over the tube is optional. The stainless optical tube is located at the center of cable protected by single or multiple layers of aluminium clad steel & aluminium alloy wires. This type of constructions can accommodate upto 48 fibers in a cable. Despite such a high fiber count in a single tube, each optical fiber is clearly distinguishable utilizing a fiber identification system consisting of coloring and the number of ring marks on it. This compact design features high mechanical strength and fault current rating within a smaller diameter. The smaller diameter also results in excellent sag tension performance.



(II) multi loose tube type:

The fiber is placed loosely in a sealed & water resistant stainless steel tube filled with water blocking gel. Two or three stainless steel optical tubes are helically stranded in the inner layer of a multiple layer cable.

The multiloose tube type is designed mostly for very high fiber count requirement over 48 with the maximum fiber count reaching 144. The multi loose tube type can meet the requirement of huge cross and large current capacity.

OPGW cable combines the function of grounding & communications:

(I) The conductive cable links the tower to the earth ground, shielding the conductors from lightning.

(II) The optical fibers within the cable provide a telecommunication path for internal as well as third party communications.

Typically OPGW contains single mode optical fibers with low transmission losses allowing long distance transmission at high speeds. The outer appearance of OPGW is similar to aluminium conductor steel reinforced cable (ACSR) usually used for shield wires.

OPGW as a communication medium has some advantages over buried optical fiber cable. Installation cost per km is lower than a buried cable. Effectively, the optical circuits are protected from accidental contact by the high voltage cables below. (and by the elevation of the OPGW from ground).

Applications : 2014/S, 2017/F

(i) OPGW is a dual functioning cable performing the duties of a ground wire and also providing a patch for the transmission of voice, video or data signals. The fibers are protected from environmental conditions like lightning, short circuit, loading to ensure reliability and longevity.

Properties :

(i) High load, long span capability.

(ii) Unique design has maximum allowable tension to control fiber strain.

(iii) Compact design results in excellent sag tension performance of the cable.

Optical Fibre ground wire system or OPGW is an optical fiber composite overhead ground wire that is used in overhead power lines. The OPGW cable contains a tabular structure with one or more optical fibers in it, surrounded by layers of steel and aluminium wire.

B) The refractive indices for core and cladding for a step index fibre are 1.52 and 1.41 respectively. Calculate (i) critical angle (ii) Numerical aperture (iii) maximum incidence angle.

sol^n,

Sol^n,

$\Rightarrow$ $\mu_1 = \mu_{core} = 1.52$

$\mu_2 = \mu_{clad} = 1.41$

(i) critical angle $(\theta_c) = \sin^{-1}\left(\frac{\mu_2}{\mu_1}\right)$

or $\theta_c = \sin^{-1}\left(\frac{1.41}{1.52}\right)$

$\therefore$ $\theta_c = 68.068°$

(ii) Numerical aperture (NA) $= \sqrt{\mu_1^2 - \mu_2^2}$

$= \sqrt{(1.52)^2 - (1.51)^2}$

$\therefore NA = 0.568$

(iii) maximum incidence angle, $(\theta_o) = \sin^{-1}\left(\sqrt{\mu_1^2 - \mu_2^2}\right)$

$= \sin^{-1}(NA)$

$= \sin^{-1}(0.568)$

$\therefore$ $\theta_o = 34.6°$

c) An optical fiber has refractive index of core & cladding is 1.514 and 1.48 resp. calculate the acceptance angle and fractional index change.

Sol^n,

$\Rightarrow$ $\mu_1 = 1.514$

$\mu_2 = 1.48$

$\phi_{in}(max) = ?$

$\Delta = ?$

We have,

(i) acceptange angle, $\phi_{in(max)} = \sin^{-1}(\mu_1^2 - \mu_2^2)^{1/2}$

$$= \sin^{-1}(1.514^2 - 1.48^2)^{1/2}$$
$$= \sin^{-1}(0.316)$$

$\therefore \phi_{in(max)} = 18^\circ\ 42'$

(ii) $\Delta = \dfrac{\mu_1^2 - \mu_2^2}{2\mu_1^2}$

$$= \frac{(1.514)^2 - (1.48)^2}{2 \times (1.514)^2}$$

$\therefore \Delta = 0.02220$

D). Consider a multimode fiber that has a core refractive index of 1.48 and a core cladding index difference of 3 percent ($\Delta = 0.03$). Find,

(i) Numerical Apertre

(ii) acceptance angle 2016/F

(iii) Critical angle

Soln,

⇒ Core refractive index (μ_1) = 1.48

Cladding " " (μ_2) = ?

Core-Cladding index differenc (Δ) = 0.003

We know,

(i) $\Delta = \dfrac{\mu_1^2 - \mu_2^2}{2\mu_1^2} \approx \dfrac{\mu_1 - \mu_2}{\mu_1} = \dfrac{\Delta\mu}{\mu_1}$

or $0.03 = \dfrac{1.48^2 - \mu_2^2}{2 \times 1.48^2}$

or $0.13142 = 2.19040 - \mu_2^2$

$\therefore \mu_2 = 1.4349$

(i) critical angle, $(\theta_c) = \sin^{-1}\left(\frac{\mu_2}{\mu_1}\right)$

$= \sin^{-1}\left(\frac{1.434}{1.48}\right)$

$= 75.677°$

(ii) acceptange angle :

$\phi_{in(max)} = \sin^{-1}(\mu_1^2 - \mu_2^2)^{1/2}$

$= \sin^{-1}(1.48^2 - 1.434^2)^{1/2}$

$= \sin^{-1}(0.36612)$

$\therefore \phi_{in(max)} = 21.4765°$

(iii) Numerical aperture (NA) $= \sin\phi_{in} = \mu_1^2 - \mu_2^2$

$\therefore NA = 0.36612$

E. Draw the simplified fiber optic communication link and briefly explain the working of each components used in order to carry the information to the destination. 2016(s)

⟹ Copy Block diagram from page No : 158

Main Basic elements of fiber optic communications system are,

(I) Compact light source
(II) photo detector
(III) low loss optical fiber.

(I) Compact Light source:

Depending on the applications like local area networks and the long haul communication systems, the light source requirement vary. The requirement of the sources include power, speed, spectral line width, noise, ruggedness, cost and so on. Two components are used as light sources: Light emitting diodes and LASER diodes.

For longer distances and high data rate transmission, laser diodes are preferred due to high power, high speed and narrower spectral line width characteristics.

(II) Low loss optical fiber:

Optical fiber is a cable which is also known as cylindrical dielectric waveguide made of low loss material. A fiber optic cable consists of four parts : Core, Cladding, Buffer, Jacket.

(III) Photo detectors:

It converts back light signal to an electrical signal. Two types of photo detectors are mainly used for optical receiver in optical communication system: PN Photo diode and avalanche photo diode. Depending on the applications wavelength, the material composition of these devices vary. These materials include germanium, silicon etc.

5 Hrs

182

6) Satellite Communications

1) A. In general terms, a satellite is a smaller object that revolves around a larger object in space. For example, moon is a natural satellite of earth. The communication refers to the exchange (sharing) of information between two or more entities, through any medium or channel. In other words, it is nothing but sending, receiving and processing of information.

If the communication takes place between any two earth stations through a satellite, then it is called as satellite communication. In this communication, electromagnetic waves are used as carrier signals. These signals carry the information such as voice, audio, video or any other data between ground and space and vice-versa.

Soviet union had launched the world's first artificial satellite named, sputnik 1 in 1957;

B. Need of satellite Communication:

The following two kinds of propagation are used earlier for communication upto some distance:

- Ground wave propagation:
  - Is suitable for frequencies upto 30 MHz. This method of communication makes use of the troposphere conditions of the earth.

- Sky wave propagation:
  - suitable bandwidth for this type of communication is

broadly between 30-40 MHZ and it makes use of the ionosphere properties of the earth.

The maximum hop or the station distance is limited to 1500 km only in both ground wave propagation and sky wave propagation, satellite communication overcomes this limitation. In this method, satellites provide communication for long distances, which is well beyond the line of sight.

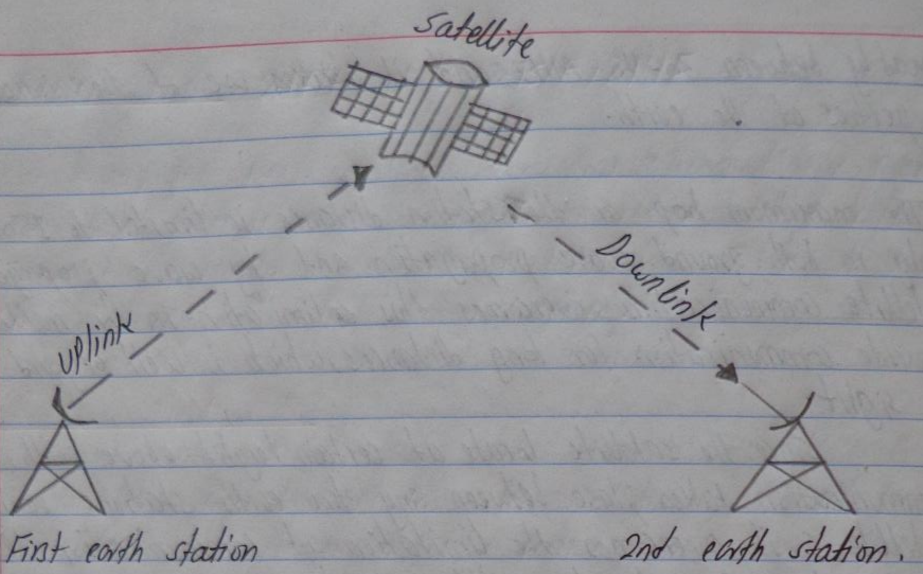
Since the satellite locate at certain height above earth, the communication takes place between any two earth stations easily via satellite. So, it overcomes the limitation of communication between two earth stations due to earth's curvature.

C. How a satellite works ??

A communication satellite is nothing but a microwave repeater station in space. It is helpful in telecommunications, radio & television along with internet applications.

A repeater is a circuit which increases the strength of received signal and then transmits it. But this repeater works as a transponder. That means, it changes the frequency band of the transmitted signal from the received one.

The frequency with which the signal is sent into the space is called as uplink frequency. Similarly, the frequency with which the signal is sent by a transponder is called as downlink frequency.



The transmission of signal from first earth station to satellite through a channel is called as uplink. similarly, the transmission of signal from satellite to second earth station through a channel is called as down link.

The process of satellite communications begin at an Earth station. Here, an installation is designed to transmit & receive signals from a satellite in an orbit around the Earth. Earth stations send the information to satellites in the form of high powered, high frequency (GHz range) signals.

The satellite receive and retransmit the signals back to earth where they are received by other earth stations in the coverage area of the satellite. Satellite's footprint is the area which receives a signal of useful strength from the satellite.

D. Advantages:

i) can reach over large geographical area.
ii) Flexible
iii) Broadcast possibilities.
iv) Easy to install new circuits.
v) capable of handling very high bandwidth.
vi) User has control over own network.
vii) Niche applications.
viii) circuit costs independent of distance.
ix) 1 for N multipoint standby possibilities.
x. Terrestrial network "By-pass".
xi Mobile applications (especially "fill-in").
xii High quality of Transmission.

E. Disadvantages:

i) Congestion of frequencies and orbits.
ii) Interference and propagation delay.
iii) launching satellite into orbit is costly
iv. Impossibility to repair and maintain.
v) satellite bandwidth is gradually becoming used up.
vi) poor spatial resolution in the polar regions.
vii) Expensive ground stations required
viii limited orbital space (geosynchronous)
ix. Terrestrial break even distance expanding.
x. Network problem.

F. Kepler's Law : 2017/F, 2015/S, 2015/F, 2016/F

Kepler (1571 - 1630) formulated three laws that changed the whole satellite communication theory and observations. There are popularly known as Kepler's laws. These are helpful to visualize the motion through space.

① Kepler's First law :

Kepler's first law states that the path followed by a satellite around its primary (the earth) will be an ellipse. This ellipse has two focal points (focii) F_1 and F_2. as shown in the figure below.

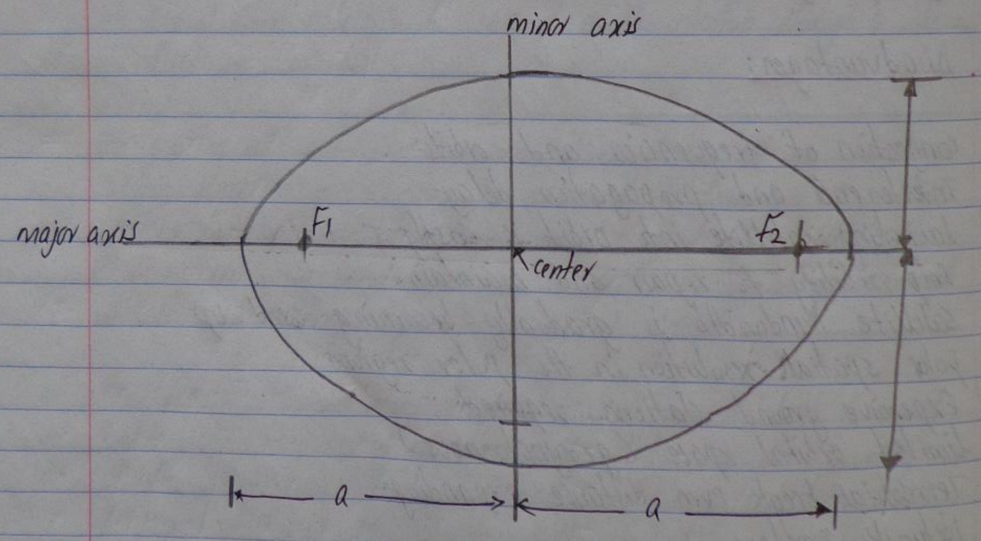


Fig: Kepler's 1st law

Center of mass of the Earth will always present at one of the two focii of the ellipse.

If the distance from the center of the object to a point on its elliptical path is considered, then the farthest point of an ellipse from the center is called as apogee and the shortest point of an ellipse from the center is called perigee.

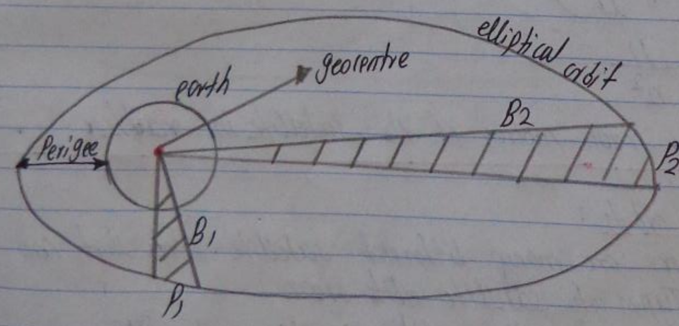
Eccentricity 'e' of this system can be written as,

$$e = \frac{\sqrt{a^2 - b^2}}{a}$$

a = length of semi-major axis
b = " " " -minor axis of ellipse.

(ii) Kepler's second law:

- states that for equal intervals of time, the area covered by the satellite will be same with respect to center of mass of earth.



Assume that the satellite covers p_1 and p_2 distances in the same time interval, then the areas B_1 and B_2 covered by the satellite at those two instances are equal.

(III) Kepler's third law :

-states that the square of the periodic time of an elliptical orbit is proportional to the cube of its semi-major axis length. mathematically,

$$T^2 \propto a^3$$

$$\Rightarrow T^2 = \left(\frac{4\pi^2}{\mu}\right) a^3$$

where, $\frac{4\pi^2}{\mu}$ = proportionality constant.

μ = kepler's constant $= 3.986005 \times 10^{14}\ m^3/sec^2$

$$\text{or } 1 = \left(\frac{2\pi}{T}\right)^2 \left(\frac{a^3}{\mu}\right)$$

$$\text{or } 1 = n^2 \left(\frac{a^3}{\mu}\right)$$

$$\Rightarrow a^3 = \frac{\mu}{n^2}$$

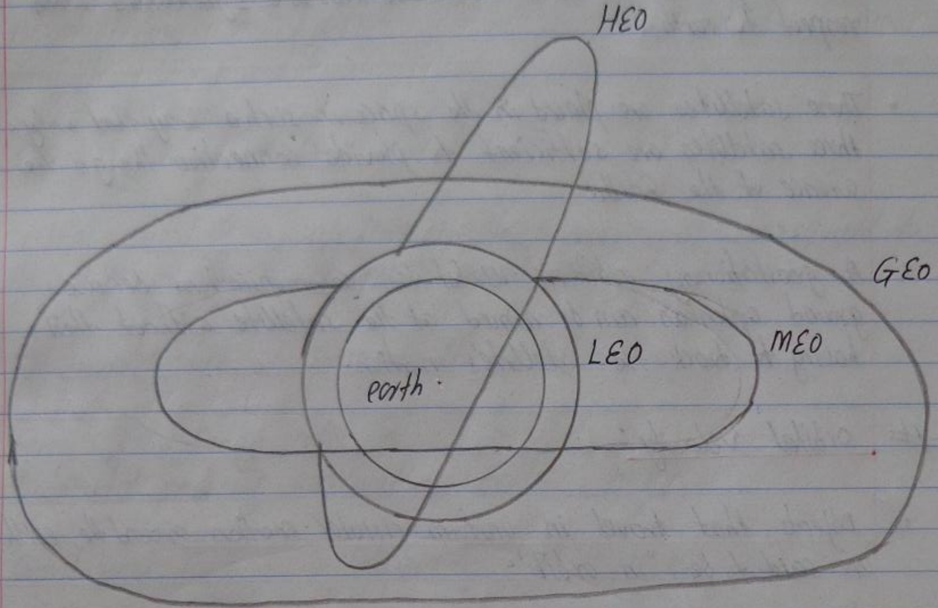
where, n = mean motion of the satellite in rad/sec.

G. Satellite orbits :

There are many different satellite orbits that can be used. Types of satellite orbits are :

- LEO : Low earth orbit - 100 - 1500 KM
- MEO : medium " " - 5000 - 10000 km
- GEO : Geostationary " " - 36000 km.
- HEO : Highly elliptical orbit. -

The ones that receive the most attention are the geostationary orbit used as they are stationary above a particular point on the Earth. The orbit that is chosen for a satellite depends upon its application.



H. Geostationary satellite :

A geostationary satellite is an earth-orbiting satellite, placed at an altitude of approximately 35800 km directly over the equator, that revolves in the same direction the earth rotates (west to east). At this altitude, one orbit takes 24 hours, the same length of time as the earth requires to rotate once on its axis.

190

Geostationary or geosynchronous earth orbit (GEO) : 2015/s

- A satellite in a geostationary orbit appears to be stationary with respect to the earth. GEO satellites are synchronous with respect to earth.
- These satellites are placed in the space in such a way that only three satellites are sufficient to provide connection through the surface of the Earth.
- A geostationary orbit is useful for communications because ground antennas can be aimed at the satellite without their having to track the satellite's motion.

Orbital velocity :

- Objects that travel in uniform circular motion around the earth are said to be "in orbit".
- velocity of this orbit depends on the distance from the object to the center of the earth.
- velocity has to be just right, so that the distance to the center of the Earth is always the same.

$$v = \sqrt{\frac{G \cdot m_E}{r}}$$

v = velocity of an object (m/s.

r = distance from the objects to the center of the Earth.
G = Universal gravitational constant $= 6.674 \times 10^{-11}\ Nm^2/kg^2$
m_E = mass of Earth $= 6 \times 10^{24}$ kg

Or $v = \sqrt{\frac{GM}{r}} = \sqrt{g \cdot \frac{R^2}{r}}$

$g = 9.8$ m/s
R_e = radius of earth

- orbital speed of a satellite is the minimum speed required to put the satellite into a given orbit around earth.
- It is independent of mass of the satellite.
- For a geostationary orbit,
  Centripetal producing acceleration, $F_c = \frac{mv^2}{r}$

  Gravitational Force on satellite, $F_g = \frac{G m \cdot M_e}{r^2}$

or, $\frac{mv^2}{r} = \frac{G \cdot m M_e}{r^2}$

$\therefore\ v = \left(\sqrt{\frac{GM_e}{R}}\right)$ m/s

Round trip time delay of GSS : 2014/S, 2015/F, 2015/S

- Round trip time delay also known as latency is the time taken for information to pass from the original to the destination & potentially for the response to return.

- Round time delay in satellites is very much more obvious because of the very long distances that the signals must travel into space and back.
- The distance travelled depends on the location of the satellite and thus its orbit.
- Round trip time delay: $\geq 2 \times \dfrac{\text{distance}}{\text{velocity of light}}$
- calculating a transmission's RTT is advantageous because it allow users and operators to identify how long a transmission will take to complete and how fast a network can operate.

Characteristics of a Geostationary satellite orbit:

- Eccentricity (e) = 0
- inclination of the orbital plane (i) = 0°
- period (T) = 23 hour 56 min 4 sec.
- semi major axis (a) = 42164 km
- satellite altitude (R) = 35786 km
- " velocity (V_s) = 3075 m/s

Advantages: 2015/F

- No ground station tracking required.
- No inter satellite handoff, permanently in view.

- 3 satellites give full earth coverage.
- Almost no Doppler shift, yields reduced complexity receivers.
- High covering area.
- No problem with frequency changes.
- One ground segment is enough for the satellite monitoring.

Disadvantages : 2015/F

- 35786 km orbits imply long transmission latencies
- Weak received signal.
- poor coverage at high lattitudes (>77 degrees).
- Large free space loss.
- Transmission delay of the order 250 ms

I) Why are uplink frequencies greater than downlink frequencies in satellite communication? 2014/S

⟹ The signals have to cross the atmosphere which presents a great deal of attenuation. The higher the frequency, the more is the signal loss and more power is needed for reliable transmission.

A satellite is a light weight device which cannot support high-power transmitters on it; so, it transmits at a low frequency (higher the frequency, higher is the transmitter power to accomodate losses) as compared to the stationary earth station which can afford to use very high power transmitters. This is compensated by using highly sensitive receiver circuits on the earth station which is in the line of sight of the satellite.

J) Footprint : 2015/s, 2017/s

- The area of the Earth covered by the microwave radiation from a satellite dish (transponder) is called the satellite footprint.

- The size of the footprint depends on the location of satellite in its orbit, the shape and size of beam produces by its transponder and the distance from the earth.

- The footprint can be seen as;
  - The area in which a broadcast signal from a particular satellite can be received.
  - The area of the surface of the earth within a satellite's transponder (transmitter or sensor) field of view.

- The signal power at the center of the footprint is maximum. The power decreases as we move from the footprint center. The boundary of the footprint is the location where the power reaches a predefined threshold.

- satellite beam their signals in a straight path to the earth. The satellite focus these microwaves signals onto the specified portion's of the earth's surface to most effectively use the limited power of their transponders. These focused signals create unique beam pattern called footprints.

Types of Footprints:

- Global Beam footprint
- Zone " "
- Hemispheric " "

- Geo satellites have no coverage on the poles.

K. Transponder: 2015/s, 2016/s

- In a Communication satellite, a transponder gathers signals over a range of uplink frequencies and retransmits them on a different set of downlink frequencies to receivers on Earth.

- Transmitter responder.
- Known as satellite repeater.
- A transpoder is typically composed of:
  - An input band limiting device (input band pass filter)
  - an " low noise amplifier.
  - Frequency translator
  - An output band pass filter.
  - A power amplifier.

- Transpoder performs the functions of both transmitter and receiver (responder) in a satellite. Hence, the word transponder is obtained by the combining few letters of two words, Transmitter & Responder.

Transponder performs mainly two functions. Those are amplifying the

the received input signal and translates the frequency of it. In general, different frequency values are chosen for both uplink and down link in order to avoid the interference between the transmitted and received signals.

Two types of Transponder:

- Re-generative Transponder
- Transparent (amplifying) Transponder.

(i) Regenerative Transponder:

- A regenerative transponder is a complete transreceiver including a demodulator and modulator.
- The uplink signal is down-converter to IF and then demodulated to baseband.
- These pulses are then remodulated into a downlink carrier up-converted and amplified before retransmission to the receiving earth station.
- The up and down links are decoupled and hence the interference and noise of two links donot add.

(ii) Amplifying Transponder:

- It consists of three main components.
  - Low noise amplifier
  - Frequency converter
  - High power amplifier (HPA) with relevant filtering.

- In these transducers, the uplink noise and interference are amplified at the transponder and transmitted to the receiving earth station. Hence, the uplink and downlink noise and interference are added at the earth station Receiver.

L) Satellite system up link and downlink:

- Power is the limitation at the satellite and power is directly proportional to frequency.
- At earth station, we can generate as much power as we need, but this is not possible at satellite.
- For less attenuation, and better signal to noise ratio, lower frequency is more suitable for down link and higher frequency is commonly used for uplink.
- RF amplifiers are most efficient at lower frequencies.
- So, uplink frequencies are higher than downlink frequencies.

M. Perform the brief comparision between satellites orbit (LEO, MEO, GEO) based on the points: 2016/S
Height, RTT, Technology used, Footprint, launching cost.

⟹

Features	LEO	MEO	GEO
(i) Height in KM	100-3000	3000-36000	36000
(ii) RTT	20-25 ms	110-130 ms	250-270 ms

(iii) Transmitter power	small	medium	large.
(iv) Antenna size	small	medium	large.
v) satellite cost	minimum	medium	maximum
vi) Satellite life	3-7 years	10-15	10-15
vii) Trunking capabilities	low	medium	High.
viii) Application	internet, data, voice, paging	24 GPS	satellite TV
ix. orbital period	90 min	6 hrs	24 hrs.
x. propagation loss	Least	High	Highest
xi No. of satellites	40-80	8-20	3

N. What are the different satellite orbits that can be used? Explain each orbits in brief. 2016/F

⇒ make figure from page 189 and explain above answer.

O) Satellite system parameters :

The Basic carrier to noise (C/N) relationship in a system establishes the transmission performance of RF portion of the system.

- The downlink thermal carrier to noise ratio is,

$C/N = C - 10 \log (KTB)$

C = Received power in dbW
B = noise bandwidth in HZ
T = absolute temperature of receiving system in °K
K = Boltzman's constant $= 1.38 \times 10^{-23}$ W/°K/HZ

Link equation:

$(C/N) = EIRP - L + G - 10 \log (KTB)$

where,
EIRP = equivalent isotropically radiated power (dBW)
L = Transmission loss
G = gain of received antenna.

$EIRP = 10 \log P_T + G_T$

- P_T = antenna input power
- G_T = Transmit antenna gain

$G = 10 \log \eta + 20 \log f + 20 \log d + 20.4$ dB

where,
η = antenna efficiency. (0.55 - 0.78)
d = diameter of antenna
f = operating frequency.

Transmission losses:

$L = L_{free\ space} + L_{rain} + L_{tracking} + L_{atmospheric}$.

equivalent noise temperature:

$T_e = T_0 (NF - 1)$

T_e = eqv. noise temperature (K)
T_o = temperature of environment
NF = noise factor.

Transmit power and Bit energy:

$$E_b = P_t \cdot T_b$$

E_b = energy of a single bit
P_t = total carrier power
f_b = bit rate
T_b = time of a single bit.

P. Applications:

(I) Weather Forecasting

(II) Radio and TV Broadcast

(III) Navigation.

iv) Military

v. Telemedicine

vi) Data communication

vii) Remote sensing

viii) GPS

ix) Tele Conference

x). Standard time

xi. Connecting remote areas.

4 Hours

4) Mobile communication

201

1)A. Frequency Reuse : 2015/s, 2014/s

- The process in which the same set of frequencies (channels) can be allocated to more than one cell, provided the cells are separated by sufficient distance.
- Used by service providers to improve the efficiency of a cellular network and to serve millions of subscribers using a limited radio spectrum.
- After covering a certain distance a radio wave gets attenuated & the signal falls below a point where it can no longer be used or cause any interference.
- A transmitter transmitting in a specific frequency range will have only a limited coverage area. Beyond this coverage area, that frequency can be reused by another transmitter. The entire network coverage area is divided into cell based on the principle of frequency reuse.
- cell = Basic geographical unit of a cellular network; is a area around an antenna where a specific frequency range is used.
- cluster = A group of adjacent cells, usually 7 cells; no frequency reuse is done within a cluster.
The frequency spectrum is divided into sub bands and each sub-band is used within one cell of the cluster. In heavy traffic zones, cells are smaller, while in isolated zones, cells are larger.

- The design process of selecting and allocating channel groups for all of the cellular base stations within a system is called frequency reuse or frequency planning.

- cell labeled with same letters use the same set of frequencies.
- cell shapes = circle, square, Hexagon, Triangle.

- Frequency reuse factor $= \frac{1}{N}$; N = cluster size

Typically, N = 4, 7 or 12.

- Frequency reuse has become essential due to :
- limited frequency spectrum available for cellular mobile communications.
- Tremendous growth in the number of mobile users.

- cells in a cluster use unique frequency channels. However, different clusters can use the same set of frequencies.

B) Interference : 2015/S

- It is the major limiting factor in the performance of cellular radio systems.
- It is a major bottleneck in increasing capacity & often responsible for dropped calls.
- Interference is more severe in urban areas, due to the greater RF noise floor and the large number of base stations & mobiles.

- Sources of interference includes another mobile in the same cell, a call in progress in a neighbouring cell, other base stations operating in the same frequency band or any noncellular systems which leaks energy into the cellular frequency band.

Two major types of interference:

(i) Co-channel interference.
(ii) Adjacent channel "

(i) Adjacent channel Interference:

It is due to imperfect receiver filters, allowing nearby frequencies to leak into pass band. It can be minimized by careful filtering and channel assignments. Channels are assigned such that the frequency separations between channels are maximized.

- Interference resulting from signals which are adjacent in frequency to the desired signal is called adjacent channel interference.

(ii) Co-channel interference: (CCI)

- Depends on R = cell radius, D = distance to base station of nearest co-channel cell,

$$D = R\sqrt{3N}$$

- Independent of the transmit power.
- Can't remove by increasing power.
- By increasing the ratio $\frac{D}{R}$, we reduce CCI
- Co-channel frequency reuse ratio as $Q = \frac{D}{R}$, then, for

Hexagonal cells, $Q = \sqrt{3K}$.

- By reducing Q,
  - cluster size K is reduced.
  - CCI is increased
  - system traffic capacity is increased.
- Q increases, → interference decreases
- cochannel interference is due to simultaneous transmission of two or more on the same channel. Cell sectoring is said to reduce this type of interference.

c. <u>cell splitting and sectoring:</u> 2014/S, 2017/F

- cell splitting is the process of subdividing a congested cell into smaller cells (micro cells), each with its own base station and a corresponding reduction in antenna height and transmitter power.
- cell splitting increases the capacity of a cellular system since it increases the number of times that channels are reused.
- The increased number of cells increases the number of clusters over the coverage area, which in turn increases the number of channels and thus capacity, in the coverage area.

Limitations:

- Handoffs are more frequent.
- channel assignments become difficult.
- All cells are not split simultaneously so special care have to be taken for proper allocation of problem.

To overcome some limitations like cochannel interference, cell sectoring is done.

- Sectoring is another way to increase the capacity.
- In this method, directional antennas are used to improve the signal to interference ratio so that the number of cells in a cluster can be reduced to increase the capacity.
- By using directional antennas, a given cell will receive interference and transmit with only a fraction of available co-channel cells.
- The further improvement in SIR is achieved by downtilting the sector antennas.

Advantages:

(i) Better S/I ratio
(ii) Reduces interference.
(iii) Reduces cluster size, more freedom in assigning channels.

Limitations:

i) Loss of Traffic.

(ii) increased number of Handoffs.
(iii) Decrease in Trunk efficiency.
iv) Increased number of antennas per base station.

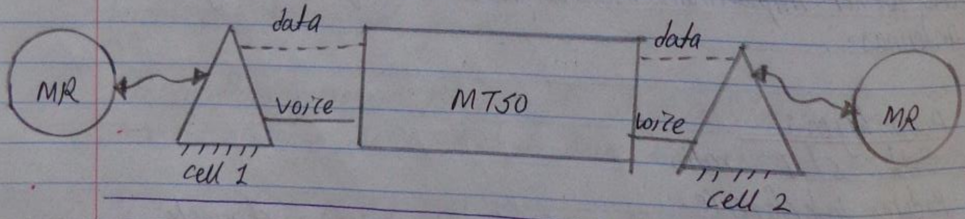
D. Basic cellular system: 2014/s

A basic cellular system consists of three parts:

i) Mobile units.

ii) cell units/site
Also known as Base Transceiver station. It contains a control unit, radio cabinets, antennas, data terminal and power plant.

iii) MSC:
Mobile switching center, also known as mobile telecommunication switching office (MTSO). It is central coordinating element for all cell sites and contains the cellular processor and a cellular switch.



E) Setting up a call process:

- When powered on, the phone doesnot have a frequency / time slot assigned to it yet, so it scans for the control channel of the BTS and picks the strongest signal.

- Then it sends a message (including its identification number) to the BTS to indicate its presence.
- The BTS sends an acknowledgement message back to the cell phone.
- The phone then registers with the BTS and informs the BTS of its exact location.
- After the phone is registered to the BTS, the BTS assigns a channel to the phone and the phone is ready to receive or make cells.

F. Making a call process : 2015/F

- The subscriber dials the receiver's number and sends it to the BTS.
- The BTS sends to its BSC the ID, location and number of the caller and also the number of the receiver.
- The BSC forwards this information to its MSC.
- The MSC routes the call to the receiver's MSC which is then sent to the receiver's BSC and then to its BTS.
- The communication with the receiver's cell phone is established.

G. Receiving a call process:

- When the receiver's phone is in an ideal state it listens for the control channel of its BTS.

- If there is an incoming call the BSC and BTS sends a message to the cells in the area where the receiver's phone is located.

- The phone monitors its message and compares the number from the message with its own.

- If the numbers matches the cell phone sends an acknowledgement to the BTS.

- After authentication, the communication is established between the caller and the Receiver.

H). Hand-OFF : / Handover :

Hand off is the process of changing the channel associated with current connection while a call is in progress. Hand off is often initiated either by crossing a cell boundary or by a deterioration in quality of signal in current channel.

- Normally induced by the quality of the ongoing communication channel parameters: Received signal strength (RSS), signal-to-noise ratio (SNR) and Bit error rate (BER).

- Handoffs are triggered either by the B.S or the mobile station itself.

- It is similar to an initial call request.

- The handoff has the priority over a new call to avoid call cut off in the mid conversation.

- In reality, a fraction of total channels can be reserved for handoff requests in each cell.

- The handoff must be successful as infrequent as possible and unnoticeable to the user.

- Types of Handoff:

(i) Hard Handoff:

- Mobile user is passed between disjoint towers that assign different frequency or adapt different air interface technology.

(ii) Soft Handoff:

- Mobile user communicates to two towers simultaneously and the signal is treated as a multipath signal.
- make-before-break
- Generally used in CDMA systems.
- The signal of the best of all connected channels is utilized.

Advantages of soft handoff:

- Reduces number of call drops.
- increase the overall capacity.
- mobile transmit power is reduced.
- voice quality near the cell boundaries are improved.

- Handoff must be performed :
  - successfully
  - infrequently
  - imperceptile

Types of Handoff	Classification	
Horizontal	Intra cell soft	Intercell Hard
vertical	Downward soft	upward Hard.

- Soft handover = make-before-break
- Hard handover = Break-before-make.

- Δ = Handoff threshold.

- Δ too large :
  - Too many handoffs
  - Burden for msc

- Δ too small :
  - More call losses
  - before call is lost
  - To complete Handoff.
  - Insufficient time.

2) GSM services: 2015/F, 2016/F

A).

- Global system for Mobile communication.

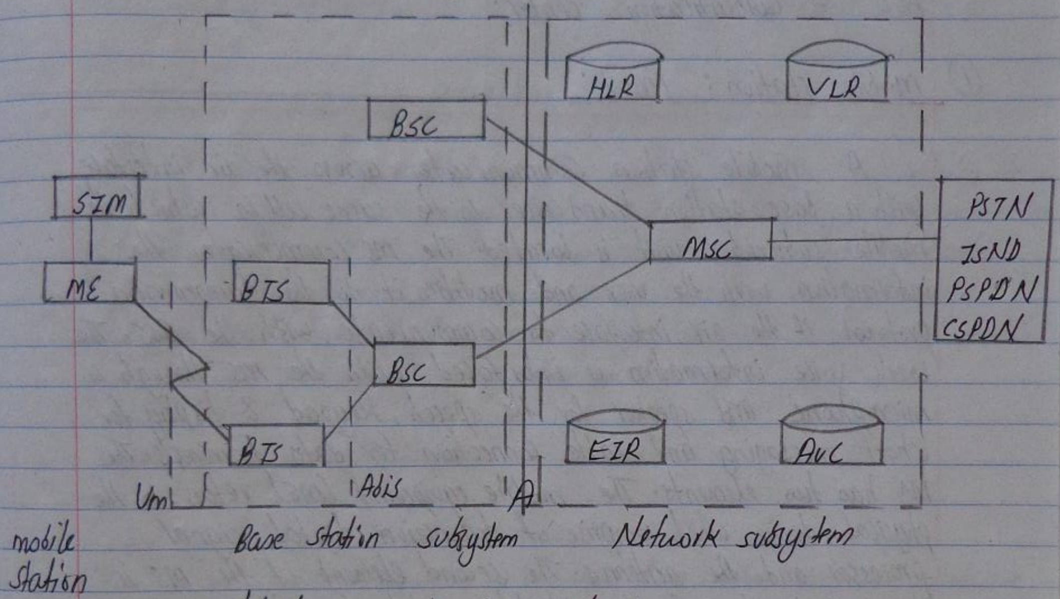


Fig: Architecture of the GSM network.

- Consists of three major subsystems:
  - Mobile station (MS)
  - Base station subsystem (BSS)
  - Network and switching subsystem (NSS)

SIM = subscriber Identify Module.
ME = mobile equipment
BTS = base Transreceiver station.

HLR = Home location register
VLR = Visitor " "
EIR = Equipment identity Register
AuC = authentication center.

(i) 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 protocol 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 & display for short messaging and cable connection for data terminals. The MS has two elements. The mobile equipment (ME) refers to the physical device which comprise of transceiver, digital signal processor 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 memory of 32 KB.

(ii) 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 100 m to 35 km, 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 element in the BSS – the Base Transceiver subsystem (BTS) and Base station controller (BSC). The interface that connects a BTS to a BSC is called the A-bis interface.

(III) 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 Network (PSTN or ISND or data network). The NSS controls handoffs between cells in different BSS, authenticates users and validates their accounts, and includes functions for enabling worldwide roaming of mobile subscribers. In particular switching subsystem consists of,

- MSC
- HLR
- VLR
- AuC
- EIR
- IWF

The NSS has one hardware, mobile switching center and four software database elements: HLR, VLR, AuC and EIR.

2016/F

8) Differentiate conventional and cellular mobile radio system.

	Conventional system	cellular mobile radio system.
(I)	No frequency reuse	Frequency reuse.
(II)	used before 1980's	used after 1980's
(III)	Bulky equipment.	Hand portable.
iv.	High antenna height.	Low antenna height.
v.	Low capacity	High capacity.

vi. High transmitted power.
vii. Poor service performance.
viii. limited service capability.
ix. inefficient frequency spectrum utilization.

c) What is cellular system capacity? 2016 S

⇒ cellular system solves the problem of spectral congestion. It offers high capacity in limited spectrum. High capacity is achieved by limited the coverage area of each B.S to a small geographical area called cell.

- Replaces high power transmitter with low power transmitters.
- several approaches for increasing cellular system capacity are,
  - cell clustering
  - sectoring of cells
  - cell splitting
  - Frequency reuse
  - Reduction of adjacent cell interference & cochannel interference.

New + Probable Questions solutions:

1) A. Back - off loss :

High power amplifiers used in Earth station transmitters and the travelling wave tubes typically used in the satellite transponders are non-linear devices; their gain (output power versus input power) is dependent on input signal level. It can be seen that as the input power is reduced by 4 dB, the output power is reduced by only 1 dB. There is an obvious power compression.

To reduce the amount of intermodulation distortion caused by the non-linear amplification of the HPA, the input power must be reduced (backed off) by several dB. This allows the HPA to operate in a more linear region. The amount the output level is backed off from rated level is equivalent to a loss and is appropriately called Back-off loss. (L_{bo}).

Thus,

$$EIRP = P_t - L_{bo} - L_{bf} + A_t$$

where,

A_t = transmit antenna gain (dB)

L_{bf} = total branching and feeder loss in dB.

L_{bo} = Back-off losses of HPA in dB

P_t = actual power output of the transmitter in dBW

Hence,

$$L_{bo} = EIRP + L_{bf} - A_t - P_t$$

To operate as efficiently as possible, a power amplifier should be operated as close as possible to saturation.

The saturated output power is designated (P_o) or simply P_t. The output power of a typical satellite earth station transmitter is much higher than the output power from a terrestrial microwave power amplifier.

Most modern satellite systems use either phase shift keying (PSK) or quadrature amplitude modulation (QAM) rather than conventional, frequency modulation (FM).

B. Noise - Temperature :

With terrestrial microwave systems, the noise introduced in a receiver or a component within a receiver was commonly specified by the parameter noise figure. In satellite communication systems, it is often necessary to differentiate or measure noise in increments as small as a tenth or a hundredth of a decibel. Noise figure, in its standard form, is inadequate for such precise calculations. Hence, it is common to use environmental temperature (T) and equivalent noise temperature (Te) when evaluating the performance of a satellite system.

Total noise power is,

$$N = KTB \quad \text{—— (I)}$$

or

$$T = \frac{N}{KB} \quad \text{—— (II)}$$

where, N = total power (watt)

K = Boltzmann's constant (J/K)

B = Bandwidth (Hz)

T = temperature of the environment (K)

For,

$$F = 1 + \frac{T_e}{T} \quad \text{—— (III)}$$

where, T_e = equivalent noise temperature (K)
F = noise factor (unitless)
T = temperature of the environment (K).

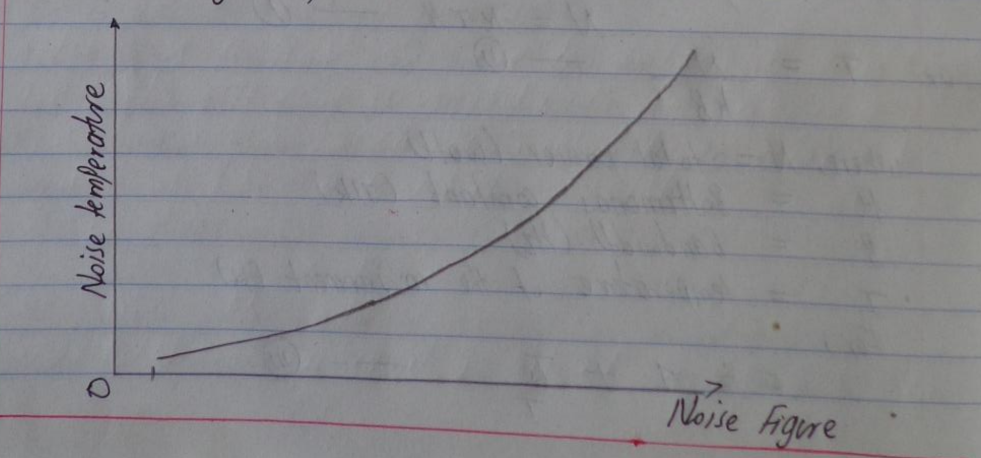
Hence,
$$T_e = T(F-1)$$

Typically, equivalent noise temperatures of the receivers used in satellite transponders are about 1000 K. For Earth station receivers, T_e values are between 20 K and 1000 K.

In dB,
$$T_e(\text{dBK}) = 10 \log T_e$$

For an equivalent noise temperature of 100K,
$$T_e = 10 \log 100$$
$$= 20 \text{ dBK}$$

Equivalent noise temperature is a hypothetical value that can be calculated but cannot be measured. It is often used rather than noise figure because it is a more accurate method of expressing the noise contributed by a device or a receiver when evaluating its performance.



C. Noise Density:

Noise density (N_0) is the power normalized to a 1-Hz bandwidth or the noise power present in a 1-Hz bandwidth. Mathematically,

$$N_0 = \frac{N}{B}$$

$$= \frac{K\,T_e \cdot B}{B}$$

$$= K \cdot T_e$$

where,

N_0 = noise density (watt/Hz).

$$1\ W/Hz = \frac{1\ \text{joule/sec}}{1\ \text{cycle/sec}} = \frac{1\ J}{1\ \text{cycle}}$$

N = total noise power (watt)

B = Bandwidth (Hz)

K = Boltzmann's constant (J/K)

T_e = equivalent noise temperature (K)

Expressing as a log with 1 W/Hz as the reference,

$$\therefore N_0\,(\text{dBW/Hz}) = 10 \log N - 10 \log B$$

$$= 10 \log K + 10 \log T_e$$

D. Carrier to Noise Density Ratio:

C/N_0 is the average wideband carrier power to noise density ratio. The wideband carrier power is the combined power of the carrier and its associated sidebands. The noise density is the thermal noise present in a normalized 1-HZ Bandwidth. The carrier-to-noise density may also be written as a function of noise temperatre. mathematically,

$$\frac{C}{N_0} = \frac{C}{K T_e}$$

Expressing in dB,

$$\frac{C}{N_0}(dB) = C(dBW) - N_0(dBW)$$

E. Energy Bit-to-Noise density Ratio:

Eb/No is one of the most important and most often used parameters when evaluating a digital radio system. The Eb/No ratio is a convenient way to compare digital systems that use different transmission rates, modulation schemes or encoding techniques.

$$\frac{E_b}{N_0} = \frac{C/f_b}{N/B} = \frac{CB}{N f_b} \quad \text{— (1)}$$

Eb/No is a convenient term used for digital system calculations and performance comparisions, but in the real world, it is more convenient to measure the wideband carrier power-to noise density ratio and convert it to Eb/No.
Thus,

$$\frac{E_b}{N_0} = \frac{C}{N} \times \frac{B}{f_b}$$

The Eb/No ratio is called product of the carrier to noise

noise ratio (C/N) and the noise band-width - to bit rate ratio. (B/f_b).

$$\therefore \frac{E_b}{N_0}(dB) = \frac{C}{N}(dB) + \frac{B}{f_b}(dB)$$

The energy per bit (E_b) will remain constant as long as the total wide band carrier power (C) and the transmission rate (bps) remains unchanged. Also, the noise density (N_0) will remain constant as long as the noise temperature remains constant.

For a given carrier power, bit rate, and noise temperature, the E_b/N_0 ratio will remain constant regardless of the encoding technique, modulation scheme or Bandwidth.

The minimum E_b/N_0 equals the minimum C/N when the receiver noise bandwidth equals the bit rate which for BPSK also equals the minimum Nyquist bandwidth.

F. Gain to equivalent Noise temperature Ratio:

Gain to equivalent noise temperature ratio (G/Te) is a figure of merit used to represent the quality of a satellite or earth station receiver. The G/Te ratio is the ratio of the receiver antenna gain (G) to the equivalent system noise temperature (Te) of the receiver. G/Te is expressed as,

$$\frac{G}{T_e} = G - 10 \log (T_s)$$

where,

G = receive antenna gain (dB)

T_s = operating or system temperature (K)

and, $T_a + T_r = T_s$.
where,
T_a = antenna temperature
T_r = receiver effective input noise temperature.

Antenna gain is a unitless value whereas temperature has the unit of degree kelvin.

G. For an equivalent noise bandwidth of 10 MHZ, and a total noise power of 0.0276 pw, calculate the noise density & equivalent noise temperature.

Soln,

=> We know,

$$N_0 = \frac{N}{B} = \frac{276 \times 10^{-16}\ W}{10 \times 10^{6}\ HZ} = 276 \times 10^{-23}\ W/HZ$$

$$N_0 = 10 \log(276 \times 10^{-23}) = -205.6\ dBW/HZ.$$

or
$$N_0 = 10 \log(276 \times 10^{-16}) - 10 \log 10\ MHZ$$
$$= -135.6\ dBW - 70\ dBW$$
$$= -205.6\ dBW$$

Also,

$$T_e = \frac{N_0}{K}$$
$$= \frac{276 \times 10^{-23}\ J/cycle}{1.38 \times 10^{-23}\ J/K}$$
$$= 200\ K/cycle.$$

Now

$$\log T_e = 10 \log 200 = 23 \text{ dBk}$$
$$= N_0 - 10 \log k$$
$$= N_0 - 10 \log(1.38 \times 10^{-23})$$
$$= -205.6 \text{ dBW} - (-228.6 \text{ dBW/k})$$
$$= 23 \text{ dB k}.$$

H) Effective Isotropic Radiated power (EIRP):

The EIRP is the transmit power of a hypothetical antenna radiating equally in all directions (like a light bulb) so as to have the same power flux density over the coverage area as the actual antenna.

The power flux density of the actual antenna is,

$$\phi = \frac{P}{S} = \frac{n^* P_{in}}{\Omega_A \cdot d^2}$$
$$= n^* \times \frac{4}{\Omega_A} \times \pi \times \frac{P_{in}}{4\pi d^2}$$
$$= G_t \cdot \frac{P_{in}}{4\pi d^2}$$

where, n^* = Antenna power loss efficiency

$P = n^* \cdot P_{in}$ = transmitted power.

S = total coverage area at distance d.

Ω_A = Antenna beam solid angle.

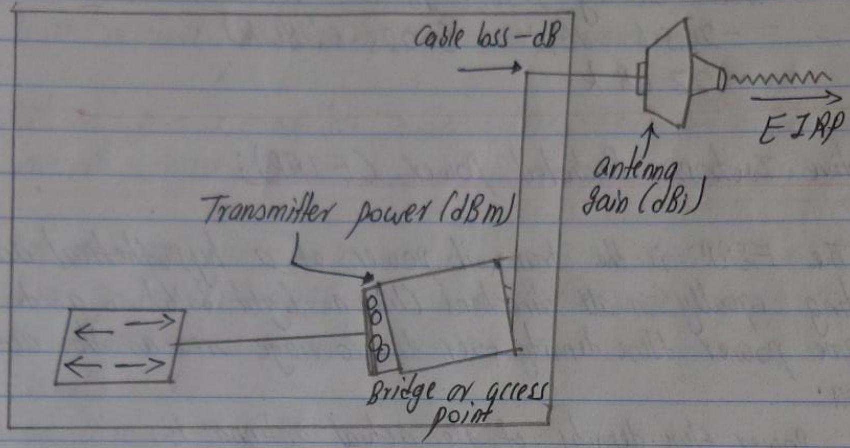
G_t = transmit gain.

By definition of EIRP,

$$\phi = \frac{EIRP}{4\pi d^2}$$

Hence, $EIRP = G_t \cdot P_{in}$.

The EIRP is the product of the antenna transmit gain and the power applied to the input terminal of the antenna.



Also, $EIRP = P_r \cdot A_t$

where,

P_r = power radiated power from antenna (W)

A_t = transmit antenna gain (W/W or unitless)

Expressing in log,

$\therefore EIRP(dBW) = P_r(dBW) + A_t(dBW)$

In respect to the transmitter output,

$$P_r = P_t + L_{bo} - L_{bf}$$

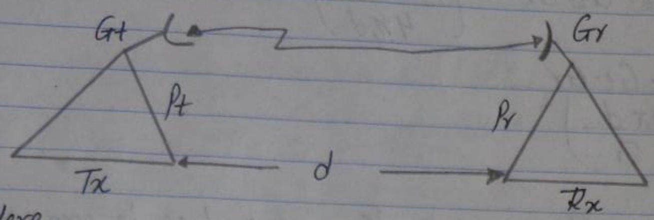
P_t = actual power output of transmitter (dBW)

L_{bo} = backoff losses of HPA (dB)

L_{bf} = total branching and feeder loss (dB)

A_t = transmit antenna gain (dB)

Proof:



Here,

$$P_r = |P_{fd}| \cdot A_d \quad \text{— (I)}$$

where, $P_{fd} = \frac{P_t \times G_t}{4\pi d^2}$

A_d = aperture of the receiving antenna.

So,

$$P_r = \frac{P_t \cdot G_t}{4\pi d^2} \cdot A_d \quad \text{— (II)}$$

Also,

$$G = \frac{\eta 4\pi A}{\lambda^2}$$

or $$A = \frac{G_r \lambda^2}{\eta 4\pi} \quad \text{— (III)}$$

or $$A_d = \frac{\eta G_r \lambda^2}{4\eta\pi} \qquad \text{where } A_d = \eta A$$

$$= \frac{G_r \lambda^2}{4\pi}$$

and,

$$P_r = \frac{P_t \cdot G_t}{4\pi d^2} \cdot A_d \qquad \text{From (II)}$$

$$= \frac{P_t \cdot G_t}{4\pi d^2} \cdot \frac{G_r \lambda^2}{4\pi}$$

$$= P_t \cdot G_t \cdot G_r \cdot \left(\frac{\lambda}{4\pi d}\right)^2$$

$$= \frac{P_t \cdot G_t \cdot G_r}{\left(\frac{4\pi d}{\lambda}\right)^2}$$

where,

$L_p = \left(\frac{4\pi d}{\lambda}\right)^2$ = path loss/free space loss ie, energy spread out as a EM wave travel away from Transmitting wave.

$$\therefore P_r = \frac{P_t \cdot G_t \cdot G_r}{L_p}$$

$$\therefore P_r = P_t \cdot G_t \cdot G_r \left(\frac{\lambda}{4\pi d}\right)^2$$

$$= \frac{EIRP}{\left(\frac{4\pi d}{\lambda}\right)^2} \cdot G_r$$

$$\therefore P_r = \frac{EIRP}{L_p} \cdot G_r$$

Expressing in dB,

$\therefore P_r (dB) = 10 \log (EIRP) + 10 \log (G_r) - 10 \log (L_p)$.

$\therefore P_r (dB) = (EIRP)_{dB} + (G_r)_{dB} - (L_p)_{dB}$.

G_t = transmitter gain

G_r = receiver gain.

P_r = power radiated from antenna.

P_t = transmitted power.

1. Figure of Footprint:

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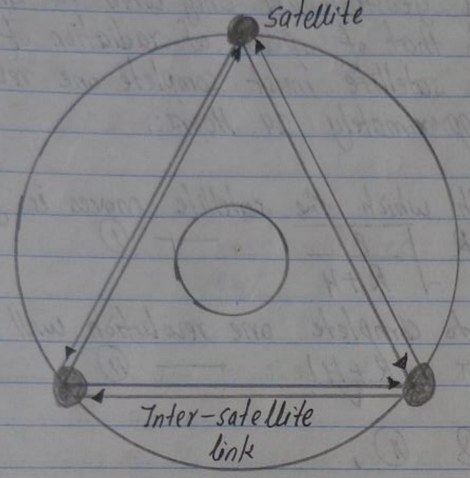

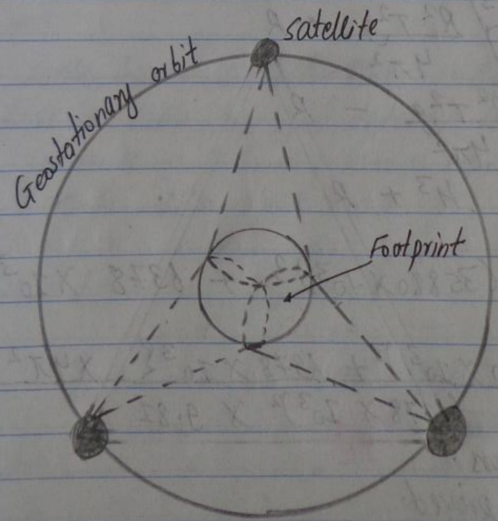

J. Prove that a satellite orbiting the earth at a height of 35880 km from the surface of the earth will be geostationary.

Soln,

=> A satellite is geostationary only when it moves with a velocity equal to that of speed of radiation of earth about its axis i.e. the satellite must complete one revolution around the earth in approximately 24 Hours.

Velocity v with which the satellite moves is given by,

$$V = R\sqrt{\frac{g}{R+H}} \quad \text{—— (i)}$$

Also, Time, T to complete one revolution will be,

$$T = 2\pi \frac{(R+H)}{V} \quad \text{—— (ii)}$$

From (i) & (ii),

$$\text{Height, } H = \sqrt[3]{\frac{R^2T^2g}{4\pi^2}} - R$$

or $$H^3 = \frac{R^2T^2g}{4\pi^2} - R$$

or $$\frac{R^2T^2g}{4\pi^2} = H^3 + R$$

$$= (35880 \times 10^3)^3 + 6378 \times 10^3$$

or $$T = \left[\frac{\{(35880 \times 10^3)^3 + 6378 \times 10^3\} \times 4\pi^2}{(6378 \times 10^3)^2 \times 9.81}\right]^{1/2}$$

$\therefore$ T = 24 Hours.

Hence, proved.

k. Determine the optical power received in dB, dBm and watts for a 50 km optical fiber link with the following design parameters: LED output $= 45$ mW, eight, 5 km sections of optical cable each with a loss of 0.2 dB/km, Four cable to cable connectors with loss of 1 dB each, two splices with loss 0.5 dB each, light source to fiber interface loss 1.5 dB, fiber to light detector loss of 1.6 dB and no losses due to cable bends. 2017/F.

Sol^n:

$\Rightarrow$ LED $\rightarrow$ Source, converted to dBm

$$P_{out} = 10 \log \left(\frac{45 \text{ mW}}{1 \text{ mW}} \right) = 16.5321 \text{ dBm}.$$

$$\text{Total cable loss} = 50 \text{ km} \times 0.2 \text{ db/km} = 10 \text{ dB}$$

$$\text{Total connector loss} = 4 \text{ connectors} \times 1 \text{ dB/connectors} = 4 \text{ dB}$$

Hence,

$\therefore$ total loss = cable loss + connector loss + light-source-to cable loss + cable to light detector loss + splices loss

$$= 10 + 4 + 1.5 + 1.6 + 2 \times 0.5$$

$$\therefore \quad P_t = 18.10 \text{ dB}$$

$$\therefore \text{Received power } (P_r) = P_{out} - P_t = \text{dBm} - \text{dB}$$

$$= 16.532 - 18.10$$

$$= -1.5680 \text{ dBm}$$

Now,

$$10 \log(P_r) = -1.5680$$

$$\text{or,} \quad P_r = \text{antilog}\left(-\frac{1.5680}{10}\right)$$

$\therefore P_r = 0.70 \text{ mwatt} = 0.00070 \text{ watt}$

$dBm = 10 \log \left(\frac{P}{1 mW} \right)$

$\therefore P$ in watts

$P_r = P_t - \text{losses}$ ← link budget analysis of optical fiber.
$= P_{in} (dBm) - A (dB)$

$\therefore P_r \text{ in } dB = -30 + dBm$
$= -30 - 1.5680$
$= -31.5680 \ dB$

Note:
$dB = dBm - 30$

$\text{watt} = 10^{\frac{dBm}{10} - 3}$

$mW = 10^{\frac{dBm}{10}}$

1. For a single mode optical cable with 0.25 dB/km loss, determine the optical power 100 km from a 0.1 mW source.
Sol^n,
⟹ We know,
$P = P_t \times 10^{\frac{-A \times L}{10}}$

$P_t = 0.1 \ mW$
$A = 0.25 \ dB/km$
$L = 100 \ km$

$\therefore P = 0.1 \times 10^{-\frac{0.25 \times 100}{10}}$
$= 0.316 \ \mu W$

Now,
$\text{Power in } dBm = 10 \log \left(\frac{P_{out}}{1 mW} \right)$
$= 10 \log \left(\frac{0.316 \times 10^{-6}}{0.001} \right)$
$= -35 \ dBm.$

Course : Communication system Engineering
Model Set-1
Programme : BE

1.a)Draw block diagram of communication system and explain each block. 5
b)A signal having a transmitted power of 5mw has been affected by a noise of power 0.02 µW. If the bandwidth of the signal is 3100Hz, then estimate the channel capacity. If the bandwidth is increased by a factor of 900Hz, what will be the effect on channel capacity? Clarify it. 6
C)Briefly discuss the needs of modulation. 4

2.A)perform the performance comparison between FM and AM. 5
B)What is amplitude modulation? Write short note on SSB modulation. 5
C)Show that any method used to demodulate DSB-SC wave can be used to demodulate an DSB-wave. 5

3.a)Describe the ASK,FSK and PSK technique with neat waveforms. 7
B)An amplitude modulated signal is given by 5[1- cos (2π * 103t)] cos (2π*107t)
Find: 5
[i] Amplitude and frequency of modulating signal.
[ii] Amplitude and frequency of carrier signal.
C)Encode the given data stream 100101101 into return to zero(RZ), non return to zero (NRZ) and AMI. 3

4.A)If the signal to interference ratio of 15dB is required for satisfactory forward channel performance of the cellular system, what is the frequency reuse factor and cluster size that should be used for maximum capacity? If the path loss exponent is (a) n = 4. (b) n = 3.Assume that there are six co-channel cells in the 1^{st} tier, all of them at the same distance from the mobile. Use suitable approximations. 7

B)What is OPGW system? How it is constructed and what are it's advantages and disadvantages? 8

5.A)How call is made in wireless communication.Explain with GSM system architecture block diagram. 7
B)Why is it necessary to develop and implement procedures for error control in communications circuit? Briefly explain error detection and error correction mechanism with examples. 8

6.A)Explain ISDN system.What benefits does it offer to user & network provider ? 7
B)State and explain Kepler's all three laws. List out the advantage and application of Geostationary Satellite. What is Transponders ? 8

7)Write short notes on (Any two): 5*2=10
A)Ethernet
B)DDD Network
C)Probability of error and bit error rate
D)Carson's rule