OUR GATE 2009 TOPPERS

CIVIL ENGINEERING TOPPERS

S.NO	STUDENT NAME	GATE H.T NO	RANK
01	SATHISH.T	CE 1620834	1
02	NETAJI PRASAD.B	CE6920096	2
03	A.BHASKER	CE1430312	3
04	MOHD.ATAULAH KHAN	CE1430072	5
05	CH.J.P.SREE RAM	CE6690202	10

MECHANICAL ENGINEERING TOPPERS

S.NO	STUDENT NAME	GATE H.T NO	RANK
01	DESAI HARDIKKUMAR .G.	ME1500007	7
02	K.VENKATACHALAPATHI	ME1500093	9

COMPUTER SCIENCE & ENGINEERING TOPPERS

S.NO	STUDENT NAME	GATE H.T NO	RANK
01	JITHIN VACHARY	CS7580245	2
02	VENKATASATYAKIRAN.D	CS1540009	10

ELECTRICAL ENGINEERING TOPPERS

S.NO	STUDENT NAME	GATE H.T NO	RANK
01	ANIL KUMAR A.	EE6240284	2 (Test series)
02	GIRISH KUMAR DASARI	EE6760582	7 (Test series)
01	V. NARASIMHAMURTHY	EE7640633	13
02	M. NAGARAJU REDDY	EE7630195	14
03	THOKARE NITHIN.D	EE1600530	18

ELECTRONICS &COMMUNICATION ENGINEERING TOPPERS

S.NO	STUDENT NAME	GATE H.T NO	RANK
01	M.YESHWANTH	EC1440615	9
02	SANDEEP SHARATH WAIKER	EC1460408	11
03	VENKATESH SASANAPURI	EC1490210	16

INSTRUMENTATION ENGINEERING TOPPERS

S.NO	STUDENT NAME	GATE H.T NO	RANK
01	MAHESH RAVI VARMA	IN1570571	2
02	MD ABDUL RAZAK	IN1570385	3
03	MAHESHWARI RUTHIKA	IN2300299	6

COMMUNICATION SYSTEMS
CONTENTS

for

GATE, DRDO, & IES

Y. V. Gopala Krishna Murthy
Managing Director

GATE SYLLABUS

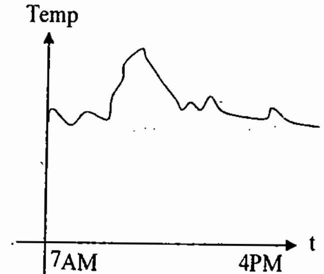
COMMUNICATION SYSTEMS

Random signals and noise: probability, random variables, probability density function, autocorrelation, power spectral density. Analog communication systems: amplitude and angle modulation and demodulation systems, spectral analysis of these operations, superheterodyne receivers; elements of hardware, realizations of analog communication systems; signal-to-noise ratio (SNR) calculations for amplitude modulation (AM) and frequency modulation (FM) for low noise conditions. Fundamentals of information theory and channel capacity theorem. Digital communication systems: pulse code modulation (PCM), differential pulse code modulation (DPCM), digital modulation schemes: amplitude, phase and frequency shift keying schemes (ASK, PSK, FSK), matched filter receivers, bandwidth consideration and probability of error calculations for these schemes. Basics of TDMA, FDMA and CDMA and GSM.

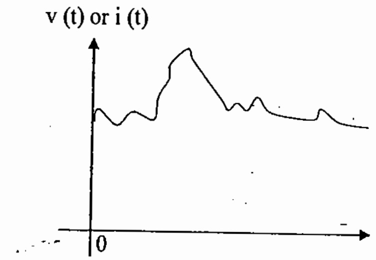
The main objective of Communication is to transfer information from one point to another point by using electrical signals

Voice -----	300-3.5KHz
Audio ----	20Hz-20KHz
Video ----	0 – 4.5 MHz

Signal is defined as a function of one or more variables and represents a physical quantity



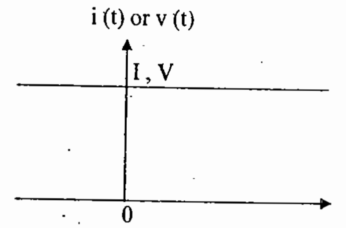
Non - electrical signal



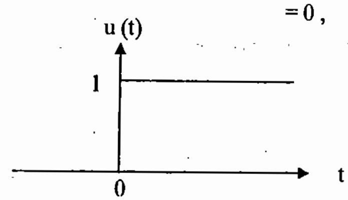
Electrical signal

Elementary Signals:-

1. D.C Signal:-With respect to time the signal voltage or current is constant for all 't'

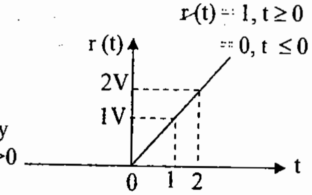


2. Unit Step Signal: - $u(t)=1, \quad t>0$
$\quad\quad = 0, \quad t<0$



3. Unit ramp Signal:- With respect to time the signal is increasing linearly with slope =1 for t >0

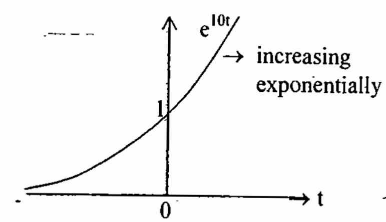
$r(t) = 1, t \geq 0$
$= 0, t \leq 0$



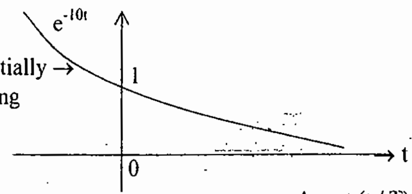
4. Exponential Signal:-

Assume $x(t)=e^{\alpha t}$

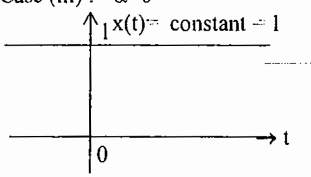
Case (i) $\alpha > 0$ ($\alpha = 10$)
$x(t) = e^{10t}$



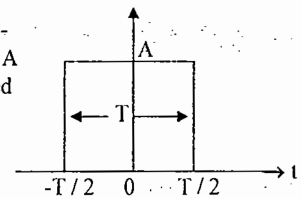
Case (ii) :- $\alpha<0$ [$\alpha = -10$]
$x(t)=e^{-10t}$



Case (iii) :- $\alpha=0$

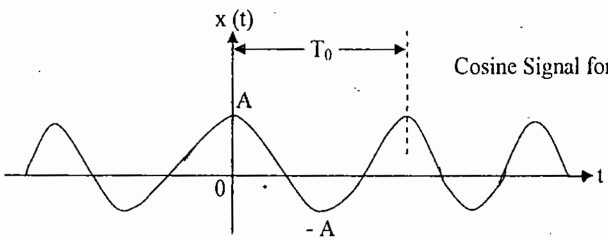


5. Rectangular Pulse:-
Pulse of amplitude A and width T centered around the origin

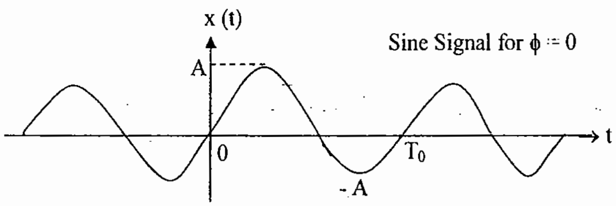


6. Sinusoidal signal:-

$x(t) = A \cos[w_0 t + \phi]$ or $A \sin[w_0 t + \phi]$ A= Max amplitude, ϕ = phase shift
w_0 = angular freq $= 2\pi f_0 = 2\pi / T_0$ (rad/sec)

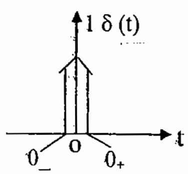


Cosine Signal for $\phi = 0$



Sine Signal for $\phi = 0$

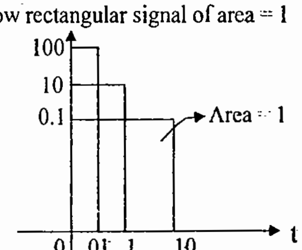
7. Unit Impulse: - $1\delta(t)$ is an ideal signal existing only at t=0 with infinite amplitude, but with unit area under it.



$\delta(t)=0 \quad t\neq0$
$\neq0 \quad t=0$

$$\int_{-\infty}^{\infty} \delta(t)\, d(t) = 1$$

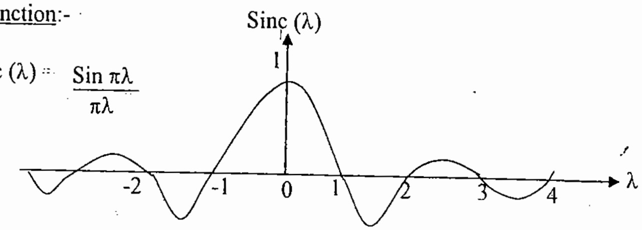
δ (t) can be considered as very very narrow rectangular signal of area = 1



When the pulse width approaches zero, amplitude goes to ∞.

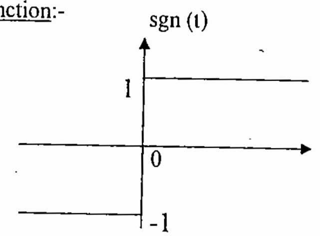
8. Sinc Function:-

$$\text{Sinc}(\lambda) = \frac{\sin \pi\lambda}{\pi\lambda}$$



The max value of sinc function equal to 1 occurs at $\lambda = 0$ and equal to zero (nulls) at $\lambda = \pm 1, \pm 2, \ldots\ldots$

9. Signum function:-



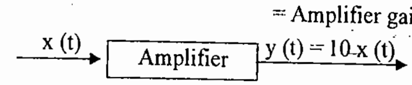
For +ve values of t the voltage =1
For –ve values of t the voltage=-1

sgn (t) = 1 t > 0
= -1 t < 0

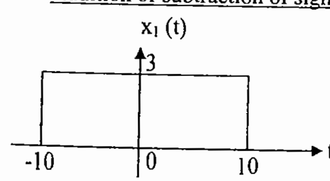
Basic Operations on signals:-

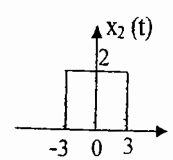
1. Amplitude scaling:-
Consider a signal x (t). Using this generate

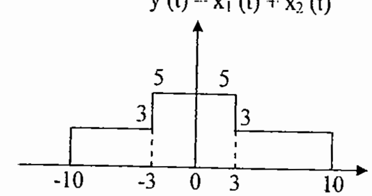
$y(t) = C\,x(t)$; C= amplitude scaling factor = Amplifier gain

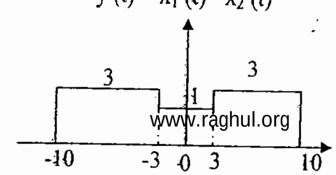


2. Addition or subtraction of signals:- $x_1(t)$ and $x_2(t)$



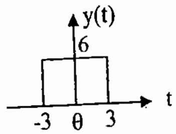




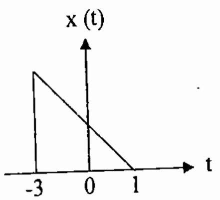


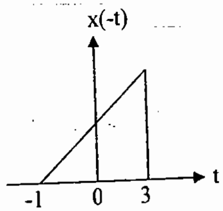
3. <u>Multiplication of signals</u>: - $x_1(t)$ and $x_2(t)$

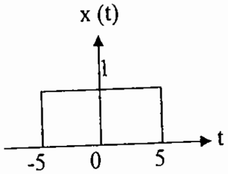
$y(t) = x_1(t)\, x_2(t)$



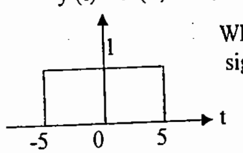
4. <u>Folding or Reflection of Signals</u>: - $y(t) = x(-t)$



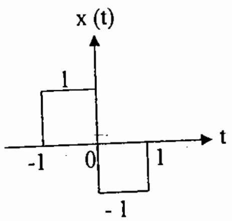




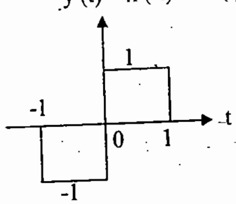
$y(t) = x(-t) = x(t)$



Whenever $x(t) = x(-t)$, $x(t)$ is an Even signal



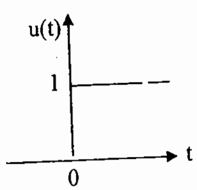
$y(t) = x(-t) = -x(t)$

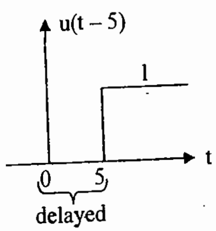


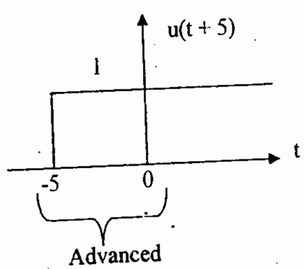
Where ever $x(t) = -x(-t)$, $x(t)$ is an odd signal

5. <u>Time shifting of signals</u>: - $y(t) = x(t - t_0) \longrightarrow$ delayed by t_o sec

$= x(t + t_0) \longrightarrow$ advanced by t_o sec

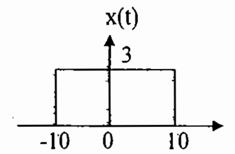






$u(t).u(t-5) = u(t-5)$, $u(t)\, u(t+5) = u(t)$

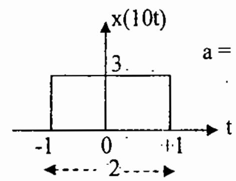
6. <u>Time Scaling of signals</u>:-

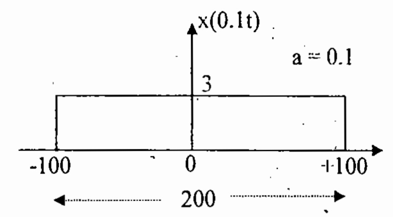


$y(t) = x(a\, t)$

When $a > 1$ Signal is compressed ←- -20- -→

$0 < a < 1$ Signal is expanded





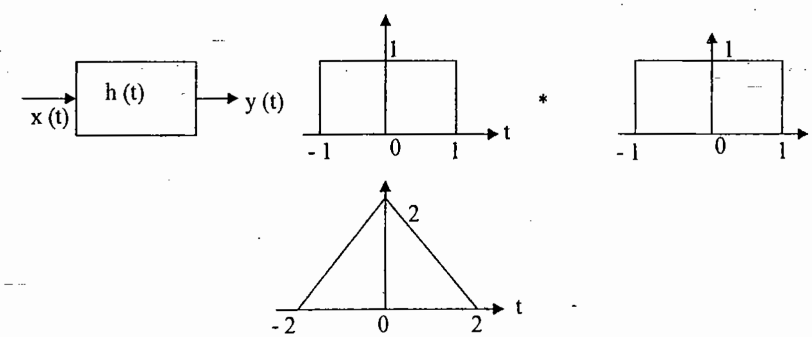
7. <u>Convolution of the $g_1(t)$ and $g_2(t) = g(t) = g_1(t) * g_2(t)$</u>

Def :- $g(t) = \int_{\tau=-\infty}^{\infty} g_1(\tau)\, g_2(t-\tau)\, d\tau = \int_{\tau=-\infty}^{\infty} g_2(\tau)\, g_1(t-\tau)\, d\tau$,

This operation is useful in system analysis.

If $x(t)$ is the input and $y(t)$ is the output of an LTI system, impulse response is $h(t)$ then,

$y(t) = x(t) * h(t)$,

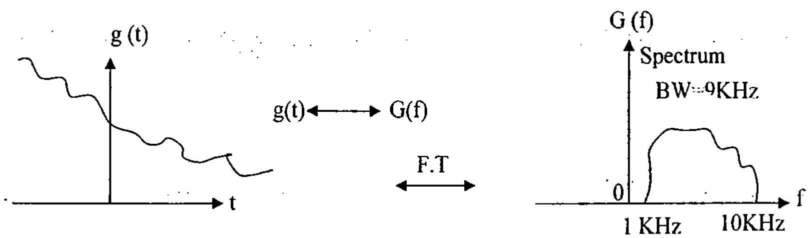


$g(t) * \delta(t) = g(t)$, $g(t) * \delta(t-t_o) = g(t-t_o)$

FOURIER TRANSFORM (SPECTRUM)

Using Fourier transform it is possible to convert a time domain signal into a frequency domain signal.

→ Spectrum gives the frequencies present in the time varying signal.
→ Using spectrum we calculate the band width.



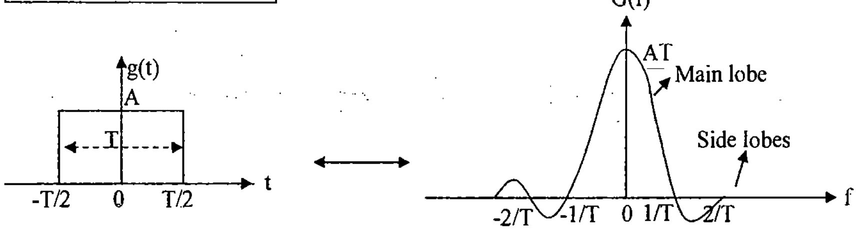
Band width:- The range of frequencies occupied by the signal is called Bandwidth.

$$\boxed{BW = f_H - f_L}$$

Fourier Transform:- **Def:** $g(t) \rightarrow G(f)$

$$\boxed{G(f) = \int_{-\infty}^{\infty} g(t)e^{-j2\pi ft}\,dt}$$

$$\boxed{g(t) = \int_{f=-\infty}^{\infty} G(f)e^{+j2\pi ft}\,df}$$



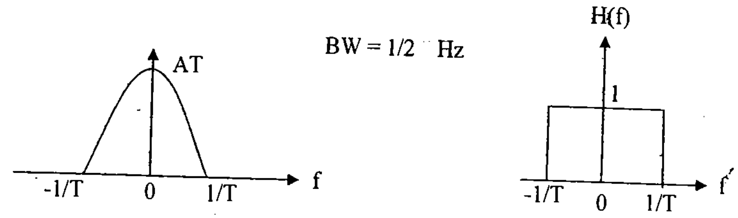
$$G(f) = \int_{-T/2}^{T/2} Ae^{-j2\pi ft}\,dt = \frac{AT\sin \pi fT}{\pi f T} = AT\,[\text{sinc}(fT)]$$

→ While calculating the BW don't consider the –ve side because –ve frequencies do not exist practically

→ In order to reduce the BW, eliminate the insignificant frequencies.

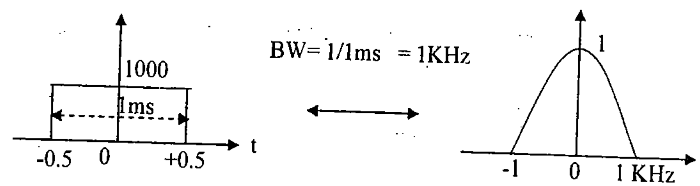
→ In order to reduce the insignificant high frequencies, pass the signal through LPF (H(f)).

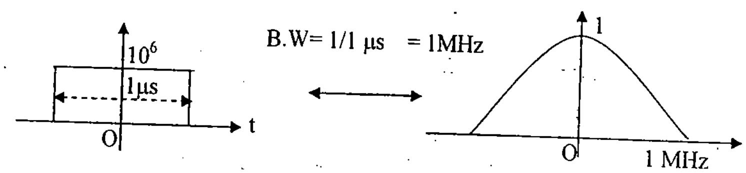
→ At the o/p we will get only main lobe



The BW of a rectangular pulse is inversely proportional to the pulse width.

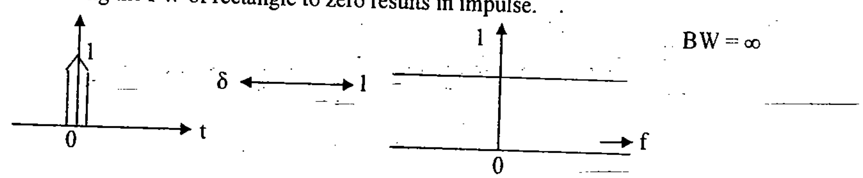
→ Take a rectangular pulse whose pulse width T = 1ms, A = 1000

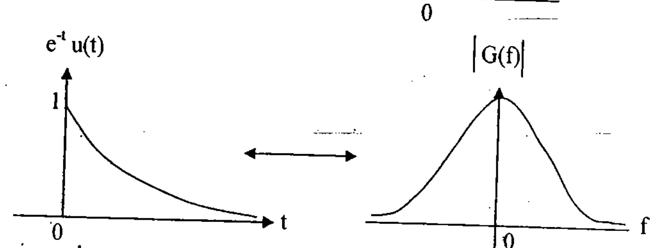




If PW decreases BW increases.

→ Reducing the PW of rectangle to zero results in impulse.





$$G(f) = \int_0^{\infty} e^{-t} e^{-j2\pi ft}\,dt \quad \boxed{= \frac{1}{1+j2\pi f}} \qquad \boxed{|G(f)| = \frac{1}{\sqrt{1+(2\pi f)^2}}} \qquad \boxed{BW = \infty}$$

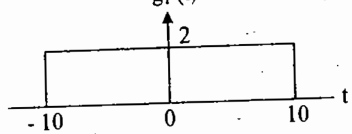
Properties of Fourier Transform:-

01. Linearity property: - $g_1(t) \longrightarrow G_1(f)$, $g_2(t) \longrightarrow G_2(f)$

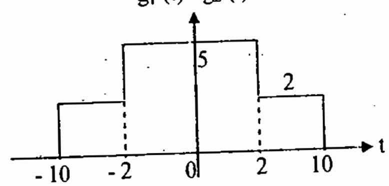
$$\boxed{g_1(t) + g_2(t) \rightarrow G_1(f) + G_2(f)}$$

Example: → Consider rectangular signals $g_1(t)$ and $g_2(t)$

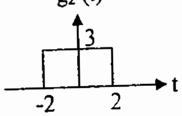
$g_1(t) \rightarrow 40\ \text{sinc}(20f)$



$g_1(t) + g_2(t)$



$g_2(t) \rightarrow 12\ \text{sinc}(4f)$

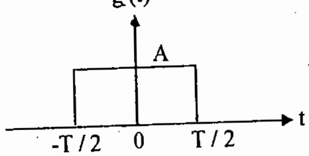


$\leftrightarrow 40\ \text{sinc}[20f] + 12\ \text{sinc}[4f]$

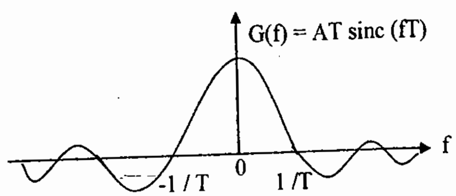
02. Duality property:

$g(t) \longleftrightarrow G(f)$

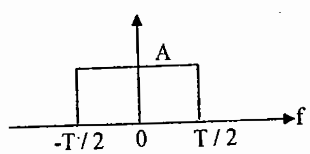
$G(t) \longleftrightarrow g(-f)$



$\longleftrightarrow$



AT Sinc (Tt) $\longrightarrow$



$\delta(t) \rightarrow 1$, $\delta(t) = \delta(-t)$ $\qquad 1 \rightarrow \delta(f)$, $\delta(f) = \delta(-f)$

03. Time Shifting property: $g(t) \leftrightarrow G(f)$

$$g(t - t_0) \leftrightarrow G(f)e^{-j2\pi f t_0} \qquad g(t + t_0) \leftrightarrow G(f)e^{+j2\pi f t_0}$$

04. Frequency shifting property (or) Modulation property: $g(t) \leftrightarrow G(f)$

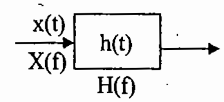
$$1 \rightarrow \delta(f), \quad e^{j2\pi f_0 t} \rightarrow \delta(f - f_0); \quad e^{-j2\pi f_0 t} \rightarrow \delta(f + f_0)$$

$$g(t)\cos(2\pi f_C t) \leftrightarrow \frac{G(f - f_C) + G(f + f_C)}{2}, \quad g(t)\sin(2\pi f_C t) \leftrightarrow \frac{G(f - f_C) - G(f + f_C)}{2j}$$

05. Convolution property: If $g_1(t) \rightarrow G_1(t)$, $g_2(t) \rightarrow G_2(t)$

$g_1(t) * g_2(t) \rightarrow G_1(t)\, G_2(t)$

$g_1(t)\ g_2(t) \rightarrow G_1(t) * G_2(t)$ apply this to an LTI system.



$y(t) = x(t) * h(t)$

$Y(f) = X(f)\ H(f)$

$H(f) = \dfrac{Y(f)}{X(f)}$ (Transfer function or frequency response of the system). Filters are classified as LPF, BPF, etc. according to the variation of H(f) with f.

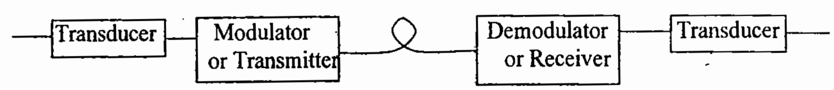
<u>**Fourier series**</u> : A periodic signal with period T_0 and fundamental frequency $f_0 = 1/T_0$ can be expressed by fourier series as a sum of terms with discrete harmonic frequencies : nf_0, n=0, 1, 2, 3, ...

$$g_p(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi n f t}; \quad C_n = \int_{t=-T_0/2}^{T_0/2} g_p(t)\, e^{-j2\pi n f t}\, dt$$

$$= \sum a_n \cos(2\pi f_0 t) + b_n \sin(2\pi n f_0 t)$$

Communication systems

Generalized Block Diagram



Irrespective of the form of communication process being considered, there are three basic elements to every communication system, namely, *transmitter*, *channel*, and *receiver*. The transmitter is located at one point in space, the receiver is located at some other point separate from the transmitter, and the channel is the physical medium that connects them. The purpose of the transmitter is to convert the message signal produced by the source of information into a form suitable for transmission over the channel.

However, as the transmitted signal propagates along the channel, it is distorted due to channel imperfections. Moreover, noise and interfering signals are added to the channel output, with the result that the received signal is a corrupted version of the transmitted signal. The receiver has the task of operating on the received signal so as to reconstruct a recognizable form of the original message signal.

Normally used communication channels are twisted pair, coaxial cable, fibre optic cable and free space.

Primary Communication Resources:

In a communication system, two primary resources are employed: *transmitted power* and *channel bandwidth*. The transmitted power is the average power of the transmitted signal. The channel bandwidth is defined as the band of frequencies allocated for the transmission of the message signal. A general system design objective is to use these two resources as efficiently as possible. In most communication channels, one source may be considered more important than the other. Therefore, communication channels are classified as power limited or band limited.

When the spectrum of a message signal extends down to zero or low frequencies, we define the bandwidth of the signal as that upper frequency above which the spectrum content of the signal is negligible and therefore unnecessary for transmitting information. The important point is unavoidable presence of noise in a communication system. Noise refers to unwanted waves that tend to disturb the transmission and processing of message signals in a communication system. The source of noise may be internal or external to the system.

A quantitative way to account for the effect of noise is to introduce *signal-to-noise ratio* (SNR) as a system parameter. We may define the SNR at the receiver input as *the ratio of the average signal power to the average noise power*, both being measured at the same point.

Modulation:
Modulation is defined as " the process in which some characteristic parameter of a high frequency carrier is varied linearly with the amplitude of the message signal ".

Generally, the carrier is represented by $c(t) = A_c \cos(2\pi f_c t + \phi)$.

The three characteristic parameters of the carrier are A_c (peak amplitude), f_c (frequency) and ϕ (phase). Accordingly

types of modulations are (1) Amplitude modulation (A.M)
...dulation (F.M)

In frequency domain, modulation is defined as " the process of translating the spectrum of a signal from low frequency region to high frequency region ".

Modulator converts i) low frequency signal to a high frequency signal.
ii) a wideband signal into narrowband signal.
iii) a baseband signal into bandpass signal.

Need for Modulation:

(1) To reduce the antenna height.

The antenna height required to transmit a signal depends on operating wavelength. For efficient radiation, the minimum antenna height should be $\lambda/10$. To transmit a low frequency signal antenna height required is very high. To reduce the antenna height, the low frequency signal is converted into a high frequency signal by modulation.

(2) For multiplexing of signals.
Multiplexing allows transmission of more than one signal through the same communication channel. By modulation it is possible to allot different frequencies to various signals so that there is no interference.

(3) To reduce noise and interference.
Some times the effect of noise will be more at some frequencies and the effect will be less at some other frequencies. If the effect of noise is more at some particular frequency, by modulation the spectrum is shifted to higher frequencies where the effect of noise is less.

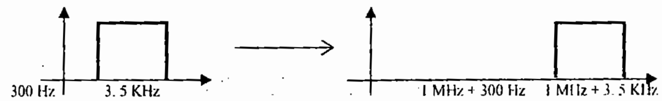
(4) For narrow banding of signals.
Not only the antenna height, the antenna dimensions also depends on operating wavelength. To transmit a wideband signal single antenna will not be sufficient because the ratio between the highest frequency to lowest frequency is very much greater than one.

Modulation converts a wideband signal into a narrowband signal whose ratio between highest frequency to lowest frequency is approximately one and single antenna will be sufficient to transmit the signal

$$\frac{f_H}{f_L} >> 1 \longrightarrow \text{Wideband signal}$$

$$\cong 1 \longrightarrow \text{Narrow band signal}$$

Eg :



(Spectrum is shifted by 1 MHz, using *modulator*)

$$\therefore \frac{f_H}{f_L} \cong 1$$

(5) To overcome equipment limitation.

The design of a communication system may be constrained by the cost and availability of hardware, hardware whose performance often depends upon the frequencies involved. Modulation permits the designer to place a signal in some frequency range that avoids hardware limitations. A particular concern along this line is the question of fractional bandwidth, defined as absolute bandwidth divided by the center frequency. Hardware costs and complications are minimized if the fractional bandwidth is kept 1 to 10 percent. Fractional-bandwidth considerations account for the fact that modulation units are found in receivers as well as in transmitters.

To win the *RACE* join the *ACE*

CHAPTER – 2 Random Signals and Noise

RANDOM VARIABLES:

$X \rightarrow x$, Probability Distribution Function $F(x) = P[X \le x]$, $-\infty \le x \le \infty$

$F(\infty) = 1$ $\quad F(x_2) \ge F(x_1)$

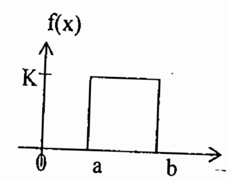
$F(-\infty) = 0$ $\quad x_2 \ge x_1$ $\quad$ Slope, $(d/dx)\,F(x) \ge 0$

Probability Density Function (PDF)

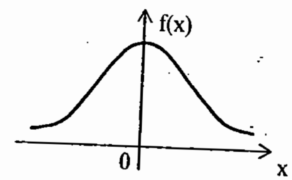
$$f(x) = \frac{d}{dx} F(x) , \qquad F(x) = \int_{x=-\infty}^{x} f(x)\, dx$$

$$f(x) \ge 0 , \qquad F(\infty) = \int_{x=-\infty}^{\infty} f(x)\, dx = 1 = \text{Area}$$

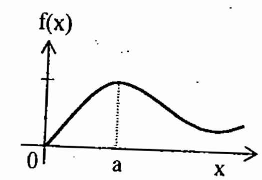
$$P[x_1 \le x \le x_2] = \int_{x=x_1}^{x_2} f(x)\, dx$$



UNIFORM



GAUSSIAN (NORMAL)

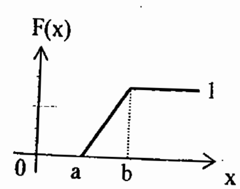


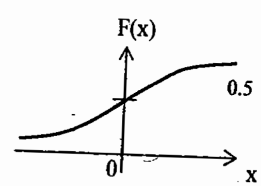
RAYLEIGH

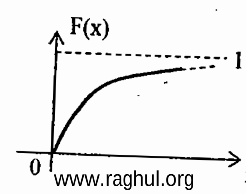
$$f(x) = \frac{1}{(b-a)} [u(x-a) - u(x-b)]$$

$$f(x) = \frac{x}{a^2} \exp [-x^2 / 2a^2]\, u(x)$$

$$f(x) = \frac{1}{\sqrt{2\pi}} \exp [-x^2 / 2\sigma^2]$$







$$\int_{t=-\infty}^{\infty} \exp[-\pi t^2]\, dt = 1, \quad z = \sqrt{\pi}\, t, \quad (1/\sqrt{\pi}) \int_{z=-\infty}^{\infty} \exp[-z^2]\, dz = 1, \quad (2/\sqrt{\pi}) \int_{z=0}^{\infty} \exp[-z^2]\, dz = 1$$

$$(2/\sqrt{\pi}) \int_{0}^{u} \exp[-z^2]\, dz + (2/\sqrt{\pi}) \int_{u}^{\infty} \exp[-z^2]\, dz = 1$$

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

$$Q(u) = (1/2)\, \cancel{\text{erfc}}\,(u/\sqrt{2}) = (1/\sqrt{\pi}) \int_{z=u/\sqrt{2}}^{\infty} \exp[-z^2]\, dz$$

$$= (1/\sqrt{2\pi}) \int_{u}^{\infty} \exp[-x^2/2]\, dx$$

Gaussian

$$F(x) = \tfrac{1}{2}[1 + \text{erf}(x/\sqrt{2}\,\sigma)] = 1 - \tfrac{1}{2}\,\text{erfc}(x/\sqrt{2}\,\sigma) = 1 - Q(x/\sigma)$$

$$P(X > K\sigma) = P(X < -K\sigma) = Q(K)$$

Rayleigh

$$F(x) = 1 - \exp[-x^2/2a^2],\ x \ge 0$$
$$= 0,\quad x < 0$$

STATISTICAL AVERAGES

$$E[x] = m_x = \int_{-\infty}^{\infty} x\, f(x)\, dx = \text{Average value of 'X'}$$

$$E[g(x)] = \int_{x=-\infty}^{\infty} g(x)\, f(x)\, dx = \text{Expected value of 'X'}$$

$$E[Y^2] = \text{MSV of } X = 2^{nd} \text{ Moment of 'X'}$$

$$= \int_{-\infty}^{\infty} x^2 f(x)\, dx$$

$$\text{VAR}(X) = E[(X - m_x)^2] = \text{MSV}(X) - (\text{M.V})^2$$

Standard Deviation (σ), VAR $= \sigma^2$

Rayleigh distribution : $\sqrt{\pi/2}\ a$, $2a^2$

Exp. Distribution:

$$f(x) = (1/a)\exp[-x/a],\ x > 0$$
$$= 0,\ x < 0$$
$$F(x) = 0,\ x < 0$$
$$= 1 - \exp[-x/a],\ x > 0$$
$$\text{M.V} = a,\ \sigma = a$$

Laplace Dist.

$$f(x) = (b/2)\exp[-b|x|]$$
$$\text{M.V} = 0,\ \text{VAR} = 2/b^2$$

Two Random Variables

$$X \to x \qquad Y \to y$$

Joint Probability Distribution function $F(x, y)$

Joint Probability Density function $f(x, y)$

$$F(x, y) = P[X \le x,\ Y \le y], \qquad f(x, y) = \frac{\partial^2}{\partial x\, \partial y} F(x, y)$$

$$F(x, y) = \int_{-\infty}^{x} \int_{-\infty}^{y} f(x, y)\, dx\, dy$$

$$\text{Volume} = \int_{-\infty}^{\infty} \int_{-\infty}^{\infty} f(x, y)\, dx\, dy = 1$$

$$f(x) = \int_{y=-\infty}^{\infty} f(x, y)\, dy, \qquad f(y) = \int_{x=-\infty}^{\infty} f(x, y)\, dx$$

$$f(x, y) = f(x/y)\, f(y) = f(y/x)\, f(x)$$

Statistical Independence: $f(x, y) = f(x)\, f(y)$

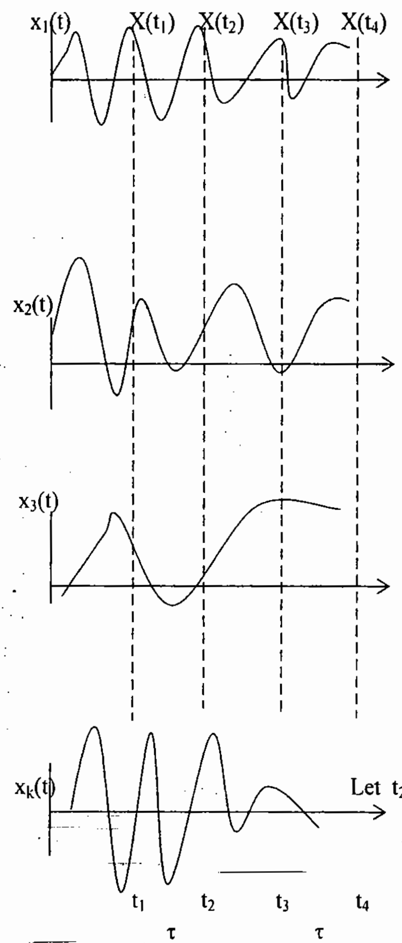
RANDOM PROCESS:

RV : X : described by $f(x)$, $F(x)$, MV, MSV, VAR, SD

2RV's : X, Y : described by $f(x, y)$, $F(x, y)$, COR (X, Y), COV (X, Y), ρ

RANDOM PROCESS X(t) : { $x_1(t)$, $x_2(t)$, ….$x_n(t)$,….}Ensemble of sample functions shown below:

EX : VOICE / TV signals, Electrical Noise



The RP observed at t_k, $k = 1, 2, 3, ----$

$X(t_1)$, $X(t_2)$, $X(t_3)$ can be considered as continuous

RV's that describe the process. Each RV above is

described by its cdf, pdf, MV, MSV, VAR, SD.

Two RV's, for example, $X(t_1)$ & $X(t_2)$ are

described by joint cdf, joint pdf, COR (auto)

COV(auto).

Random process X(t) is said to be STATIONARY

to order '2' (wide sense or weak stationarity).

If pdf's of RV's are invariant under a time

shift: T.

Let $t_2 - t_1 = \tau$, $t_3 = t_1 + T$, $t_4 = t_2 + T$, $t_4 - t_3 = \tau$

then pdf of $X(t_1)$ = Pdf of $X(t_2)$ ....= const,

independent of t.

$\therefore$ $E[X(t_k)] = E[X(t)] = \text{const} = m_X$, $k = 1,2 ......$

Also the joint PDF of $X(t_1)$ and $X(t_2)$ is same as joint pdf of $X(t_3)$ & $X(t_4)$

$\therefore$ ACF of $X(t_1)$ & $X(t_2)$ = ACF of $X(t_3)$ & $X(t_4)$ = $R_X(\tau)$

Where $\tau = t_2 - t_1 = t_4 - t_3$ = Time difference between the observation instants.

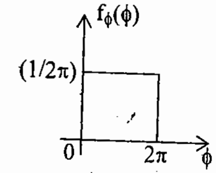
$\therefore$ ACF : $R_X(\tau) = E[X(t)\ X(t+\tau)] = E[X(t-\tau)\ X(t)]$, ACV : $K_X(\tau) = R_X(\tau) - m_x^2$.

Properties:

1) $R_X(0) = E[X^2(t)]$ = MSV = power   2) $R_X(\tau) = R_X(-\tau)$, EVEN.

3) $|R_X(\tau)| \leq R_X(0)$ 4) $R_x(\tau) \to S_x(f)$; ACF $\to$ PSD

Ex : RP $X(t) = A\cos[2\pi f_0 t + \phi]$, ϕ = UNIF DIST RV, $f_\phi(\phi) = (1/2\pi)$, $0 < \phi < 2\pi$



$$E[X(t)] = \int_{\phi=0}^{2\pi} A\cos[2\pi f_0 t + \phi]\,(1/2\pi)\,d\phi = 0$$

$R_X(\tau) = E[X(t)\,X(t+\tau)] = E[A^2 \cos(\quad)\cos(\quad)]$

$= (1/2)\,E[A^2 \cos(2\pi f_0 \tau)] + E[(1/2)A^2\cos\{4\pi f_0 t + 2\pi f_0 \tau + 2\phi\}]$

$R_X(\tau) = (A^2/2)\cos(2\pi f_0 \tau)$; $R_X(0) = E[X^2] = (A^2/2)$.

$R_X(\tau)$ is also a periodic function of 'τ' with same frequency 'f_0'.

EX: Let $Y(t) = X(t) - \rho X(t+\tau)$, to find '$\rho$' for minimizing MSV of Y(t) for given 'τ'

X(t), Y(t) zero mean station – R.P's, ρ & τ are constants.

$E[Y_t^2] = E[(X_t - \rho X_{t+\tau})^2]$

$= E[X_t^2] - 2\rho E[X_t X_{t+\tau}] + \rho 2\ E[X_{t+\tau}^2]$

$= R_X(0) - 2\rho R_X(\tau) + \rho^2 R_X(0)$

$$\frac{d}{d\rho} E[Y_t^2] = -2R_X(\tau) + 2\rho R_X(0),\quad \rho_{opt} = \frac{R_X(\tau)}{R_X(0)}$$

$$E[Y_t^2]_{min} = R_X(0) - \frac{R_X^2(\tau)}{R_X(0)}$$

EX : Let X(t), Y(t) Wide – sense stat. R P's with $R_X(\tau)$, $R_Y(\tau)$, $R_{XY}(\tau)$, $R_{YX}(\tau)$ as COR. Funs

Show that $|R_{XY}(\tau)| \leq (1/2)[R_X(0) + R_Y(0)]$

$E[(X(t \pm \tau) + Y(t)^2] \geq 0$, $E[X^2(t+\tau) + E[Y^2(t)] \pm 2E[X(t+\tau)\,Y(t)] \geq 0$

$R_X(0) + R_Y(0) \pm 2R_{XY}(\tau) \geq 0$, $\pm R_{XY}(\tau) \leq (1/2)[R_X(0) + R_Y(0)]$

$|R_{XY}(\tau)| \leq (1/2)[R_X(0) + R_Y(0)]$

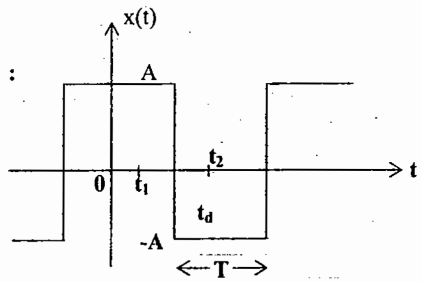
EX : Random Binary wave : $X(t)$

A typical sample function $x_1(t)$ is shown :

Duration of the pulse = T

In any pulse duration T, the value of $x(t)$

is $\pm s$ with equal probability . The value

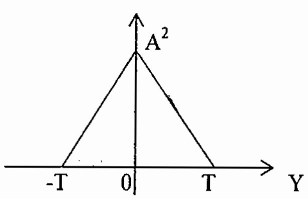


of $x(t)$ in a given pulse duration T is statistically independent of the value of $x(t)$ in other durations of 'T'. The time of occurrence of the first pulse from the origin is shown as t_d. It is a random variable , $T_d \rightarrow t_d$, $0 \le t_d \le T$

$E[X(t)] = 0$, $E[X(t_1)X(t_2)] = R_X(\tau) = A[1-(|\tau|/T)]$, $|\tau| \le T$

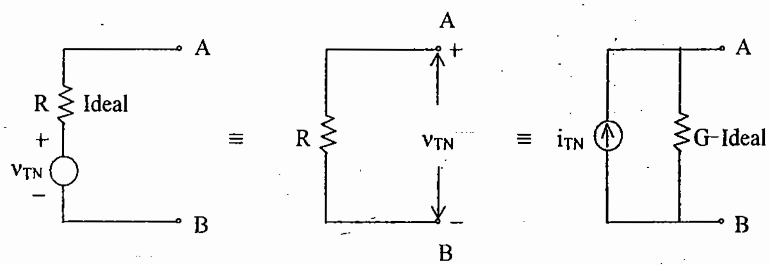
$= 0$, $|\tau| \ge T$

Where $\tau = (t_2 - t_1)$



$S_X(f) = A^2 T \, \text{sinc}^2(Tf)$

THERMAL NOISE



Phy. (Noisy) Res.   G = 1 / R

$E\left[V_{TN}^2\right] = 4\left[KTR(\Delta f)\right] V^2$; $E\left[I_{TN}^2\right] = 4\left[KTG(\Delta f)\right] A^2$

Where K = Boltzmann Constant = 1.38×10^{-23} J / ^{0}K and T = Abs. Temp in (^{0}K)

Thermal Noise is Gaussian with MV = 0.

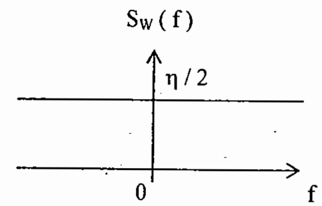
Ava. Noise Pow (max) = $\left[KT(\Delta f)\right]$ W

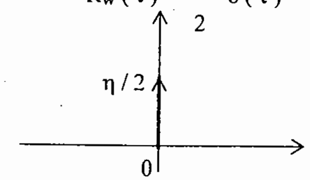
Ava. Noise PSD = $\dfrac{KT}{2} = \dfrac{\eta}{2}$ W / Hz

Where $\eta = KT$ = Noise Power per unit BW.

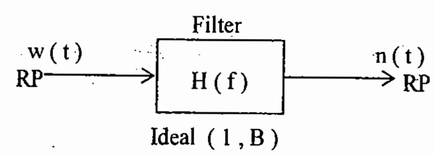
Thermal Noise is modelled as White Noise .

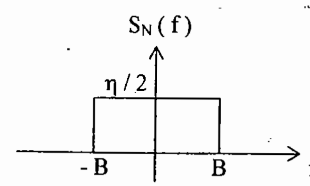
PSD of white noise : $S_W(f)$ and ACF $R_W(\tau)$ are shown below :





Ex- : White noise through Ideal LPF.

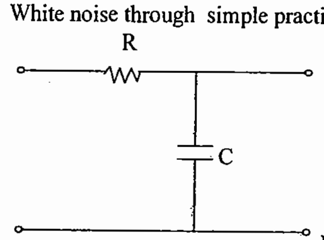




$S_N(f) = S_W(f)\,|H(f)|^2$, $R_N(\tau) = \eta B \,\text{Sinc}\,(2B\tau)$

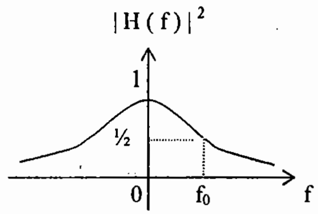
Sketch $R_N(\tau)$.

Ex -: White noise through simple practical RC LPF .

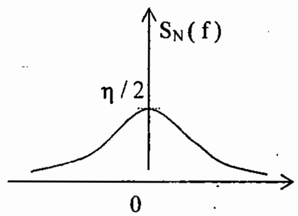


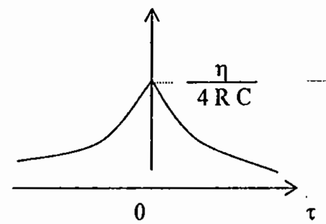
$H(f) = \dfrac{1}{1 + j(f/f_0)}$

Where f_0 = 3 dB BW = 1 / (2πRC)



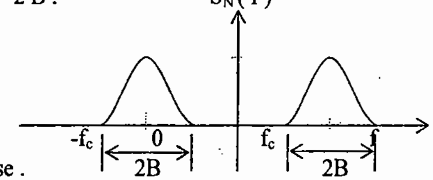
$$R_N(\tau) = \frac{\eta}{4RC} \exp\left[\frac{-|\tau|}{RC}\right]$$





BP Noise Process: $n(t)$, c . f = f_c, BW = 2 B .

A typical P.D.S: $S_N(f)$ is shown below



Normally $f_c >> 2B$.

∴ $n(t)$ is referred to as Narrow Band noise .
Similar to a deterministic BP signal g (t):

$$n(t) = n_c(t)\cos(2\pi f_c t) - n_s(t)\sin(2\pi f_c t) = r(t)\cos[2\pi f_c t + \psi(t)]$$

$$n_c(t) = n(t)\cos(2\pi f_c t) + \hat{n}(t)\sin(2\pi f_c t)$$

$$n_s(t) = \hat{n}(t)\cos(2\pi f_c t) - n(t)\sin(2\pi f_c t)$$

$$R_{Nc}(\tau) = R_{Ns}(\tau)\ ; \quad S_{Nc}(f) = S_{Ns}(f) = S_N(f - f_c) + S_N(f + f_c)\ ,\ |f| < B$$
$$= 0 \qquad , |f| > B$$

→ The power spectral density (PSD) of a noise process is given by

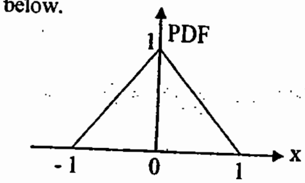
$$S_N(f) = \begin{cases} 10^{-8}\left[1 + \dfrac{|f| - 10^8}{10^8}\right] & |f| < 10^8 \\ 0 & |f| > 10^8 \end{cases}$$

The noise is passed through a unity-gain ideal bandpass filter , centered at 50 MHz and having a bandwidth of 2 MHz .

a) Sketch neatly the PSD of the output noise process .
b) Determine the output noise power .
c) Using the bandpass representation for the output noise process, sketch the PSD of the inphase and quadrature noise components, and determine their respective powers.

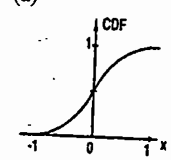
Chapter – 2 **Objective Questions**

01. The probability density function (PDF) of a random variable X is as shown below.

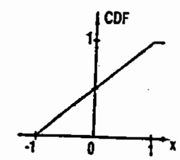


The corresponding cumulative distribution function (CDF) has the form

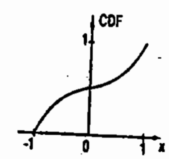
(a)



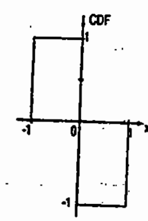
(b)



(c)



(d)



02. Noise with double-sided power spectral density of K over all frequencies is passed through a RC low pass filter with 3 dB cut-off frequency of f_c. The noise power at the filter output is

(a) K (b) Kf_c (c) $K\pi f_c$ (d) ∞

03. If $R(\tau)$ is the autocorrelation function of a real, wide-sense stationary random process, then which of the following is NOT true?

(a) $R(\tau) = R(-\tau)$
(b) $|R(\tau)| \le R(0)$
(c) $R(\tau) = -R(-\tau)$
(d) The mean square value of the process is $R(0)$

04. If $S(f)$ is the power spectral density of a real, wide-sense stationary random process, then which of the following is ALWAYS true?

(a) $S(0) \ge S(f)$ (b) $S(f) \ge 0$

(c) $S(-f) = -S(f)$ (d) $\int_{-\infty}^{\infty} S(f)\, df = 0$

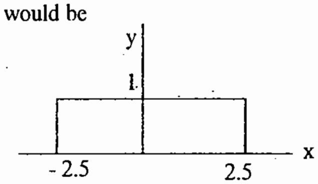
05. During transmission over a certain binary communication channel, bit errors occur independently with probability p. The probability of AT MOST one bit error in a block of n bits is given by

(a) p^n (b) $1-p^n$
(c) $np(1-p)^{n-1}+(1-p)^n$ (d) $1-(1-p)^n$

06. A uniformly distributed random variable X with probability density function

$f_X(x) = 1/10\,(u(x+5) - u(x-5))$

where u (.) is the unit step function is passed through a transformation given in the figure below. The probability density function of the transformed variable Y would be



(a) $f_Y(y) = 1/5\,(u(y+2.5)-u(y-2.5))$

(b) $f_Y(y) = 0.5\delta(y) + 0.5\delta(y-1)$

(c) $f_Y(y) = 0.25\delta(y+2.5) + 0.25\delta(y-2.5) + 0.5\delta(y)$

(d) $f_Y(y) = 0.25\delta(y+2.5) + 0.25\delta(y-2.5) + 1/10\,(u(y+2.5) - u(y-2.5))$

07. A zero – mean white Gaussian noise is passed through an ideal lowpass filter of bandwidth 10 kHZ. The output is then uniformly sampled with sampling period $t_s = 0.03$ msec. The samples so obtained would be

(a) correlated

(b) statistically independent

(c) uncorrelated

(d) orthogonal

Statement for Linked Answer Questions 08 & 09.

The following two questions refer to wide sense stationary stochastic processes

08. It is desired to generate a stochastic process (as voltage process) with power spectral density

$$S(\omega) = \frac{16}{16+\omega^2}$$

by driving a Linear – Time - Invariant systems by zero mean white noise (as voltage process) with power spectral density being constant equal to 1. The system which can perform the desired task could be

(a) first order lowpass R – L filter

(b) first order highpass R – C filter

(c) tuned L – C filter

(d) series R – L – C- lowpass filter

09. The parameters of the system obtained in Q78 would be

(a) first order R – L lowpass filter would have $R = 4\,\Omega$ $L = 4$ H

(b) first order R – C highpass filter would have $R = 4\,\Omega$ $C = 0.25$F

(c) tuned L – C filter would have $L = 4$ H $C = 4$ F

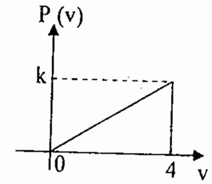
(d) series R – L – C lowpass filter would have $R = 1\,\Omega$, $L = 4$H, $C = 4$ F

10. Noise with uniform power spectral density of N_0 W /Hz is passed through a filter $H(\omega) = 2\exp(-j\omega t_d)$ followed by an ideal low pass filter of bandwidth B Hz. The output noise power in Watts is

(a) $2N_0B$ (b) $4N_0B$

(c) $8N_0B$ (d) $16N_0B$

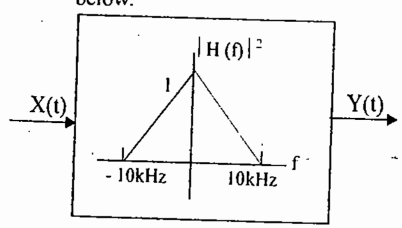
11. An output of a communication channel is a random variable v with the probability density function as shown in fig. The mean square value of v is



(a) 4 (b) 6

(c) 8 (d) 9

12. A white noise process X(t) with two – sided power spectral density 1×10^{-10} W/Hz is input to a filter whose magnitude squared response is shown below.



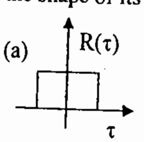
The power of the output process Y(t) is given by

(a) 5×10^{-7} W (b) 1×10^{-6} W

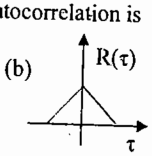
(c) 2×10^{-6} W (d) 1×10^{-5} W

13. If the power spectral density of stationary random process is a sinc – squared function of frequency, the shape of its autocorrelation is

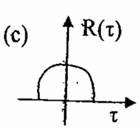
(a)



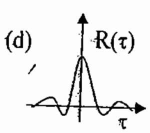
(b)



(c)



(d)



14. If the variance σ_d^2 of $d(n) = x(n) - x(n-1)$ is one-tenth the variance σ_x^2 of a stationary zero-mean discrete-time signal x(n), then the normalized autocorrelation function $R_{xx}(k)/\sigma_x^2$ at $k = 1$ is

a) 0.95 b) 0.90

c) 0.10 d) 0.05

15. The PDF of a Gaussian random variable X is given by $p_X(x) = (1/3\sqrt{2\pi})\exp[-(x-4)^2/18]$. The probability of the event $\{X = 4\}$ is

a) 1/2 b) $1/(3\sqrt{2\pi})$

c) 0 d) 1 / 4

16. During transmission over a communication channel, bit errors occur independently with probability 'p'. If a block of n bits is transmitted, the probability of at most one bit error is equal to

a) $1-(1-p)^n$

b) $p+(n-1)(1-p)$

c) $np(1-p)^{n-1}$

d) $(1-p)^n + np(1-p)^{n-1}$

17. The PSD and the power of a signal g(t) are, respectively, $S_g(\omega)$ and P_g. The PSD and the power of the signal ag(t) are, respectively

a) $a^2S_g(\omega)$ and a^2P_g

b) $a^2S_g(\omega)$ and aP_g

c) $aS_g(\omega)$ and a^2P_g

d) $aS_g(\omega)$ and aP_g

18. The amplitude spectrum of a Gaussian pulse is

a) uniform b) a sinc function

c) Gaussian d) an impulse function

19. The ACF of a rectangular pulse of duration T is
a) a rectangular pulse of duration T
b) a rectangular pulse of duration 2T
c) a triangular pulse of duration T
d) a triangular pulse of duration 2T

20. The probability density function of the envelope of narrow band Gaussian noise is
a) Poisson b) Gaussian
c) Rayleigh d) Rician

21. A probability density function is given by $p(x) = K\exp(-x^2/2), -\infty < x < \infty$. The value of 'K' should be
a) $1/\sqrt{2\pi}$ b) $\sqrt{2/\pi}$
c) $1/2\sqrt{\pi}$ d) $1/\pi\sqrt{2}$

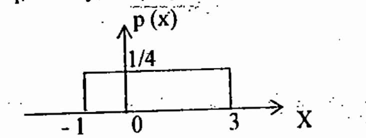
22. The power spectral density of a deterministic signal is given by $[\sin(f)/f]^2$ where 'f' is frequency. The autocorrelation function of this signal in the time domain is
a) a rectangular pulse
b) a delta function
c) a sine pulse
d) a triangular pulse

23. The autocorrelation function of an energy signal has
a) no symmetry
b) conjugate symmetry
c) odd symmetry
d) even symmetry

24. For a narrow band noise with Gaussian quadrate components, the probability density function of its envelope will be
a) uniform b) Gaussian
c) exponential d) Rayleigh

25. Two resistors R_1 and R_2 (in ohms) at temperature T_1° K and T_2° K respectively, are connected in series. Their equivalent noise temperature is ________°K.

26. For a random variable 'x' following the probability density function, p (x), shown in figure, the mean and the variance are, respectively,



a) 1 / 2 and 2/ 3 b) 1 and 4/ 3
c) 1 and 2/ 3 d) 2 and 4/ 3

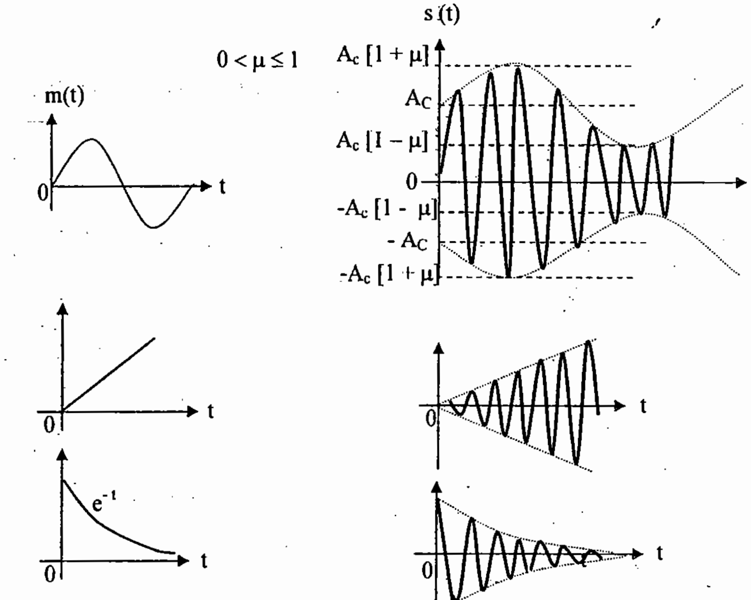
27. Zero mean Gaussian noise of variance N is applied to a half wave rectifier. The mean squared value of the rectifier output will be :
(a) Zero, (b) N / 2
(c) $N/\sqrt{2}$ (d) N

28. In a digital communication system, transmissions of successive bits through a noisy channel are assumed to be independent events with error probability p. The probability of at most one error in the transmission of an 8 - bit sequence is
(a) $7(1-p)/8 + p/8$
(b) $(1-p)^8 + 8p(1-p)^7$.
(c) $(1-p)^8 + (1-p)^7$.
(d) $(1-p)^8 + (1-p)^7$

29. The variance of a random variable X is σ_x^2. Then the variance of $-kx$ (where k is a positive constant) is.
(a) σ_x^2 (b) $-k\sigma_x^2$
(c) $k\sigma_x^2$ (d) $k^2\sigma_x^2$

(A) Amplitude Modulation (AM)

In A.M , the peak amplitude $A_c\cos(2\pi f_c t + \phi)$ of the carrier is varied linearly with the amplitude of the message signal.



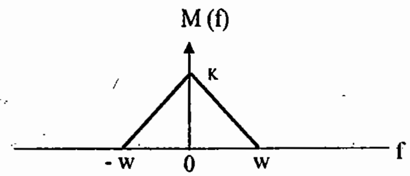
Message AM signal

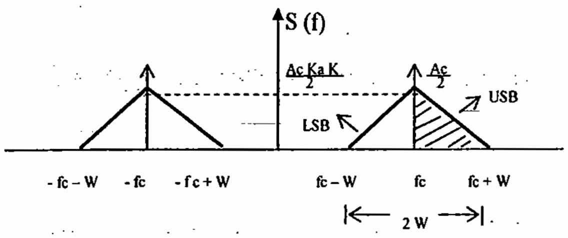
Time domain equation of A.M :

$$s(t) = A_c \cos 2\pi f_c t + A_c K_a m(t)\cos 2\pi f_c t = \boxed{A_c[1 + K_a m(t)]\cos 2\pi f_c t}$$

The amplitude of the carrier before modulation is A_c and the amplitude of the carrier after modulation is $A_c[1 + K_a m(t)]$. (After modulation, the carrier amplitude depends on the message signal), K_a = amplitude sensitivity of the modulator

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





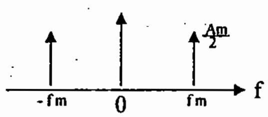
Bandwidth of the A.M signal = 2 W
= 2 × message bandwidth, Bandwidth of USB = W
Bandwidth of LSB = W

Single tone Modulation of A.M :

When the message contains single frequency or single tone, then the modulation is called single tone modulation.

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

$$M(f) = \frac{A_m}{2}\left[\delta(f - f_m) + \delta(f + f_m)\right]$$



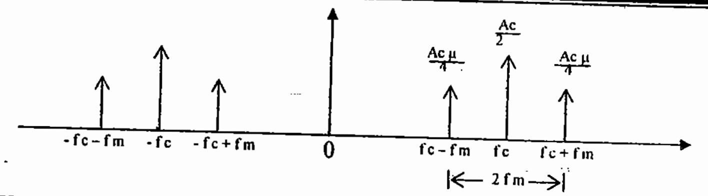
The time domain equation of A.M for single tone modulation is

$$s(t) = A_c \cos(2\pi f_c t) + A_c K_a A_m \cos(2\pi f_m t) \cdot \cos(2\pi f_c t)$$

$s(t) = A_c [1 + \mu \cos(2\pi f_m t)] \cdot \cos(2\pi f_c t)$, where $\boxed{K_a A_m = \mu = \text{modulation index}}$

and the amplitude of the carrier after modulation is $A_c [1 + \mu \cos(2\pi f_m t)]$,
the maximum value of the positive envelope is $A_c [1 + \mu]$,
the minimum value of the positive envelope is $A_c [1 - \mu]$.

$$\boxed{s(t) = \underbrace{A_c \cos 2\pi f_c t}_{\text{carrier}} + \underbrace{\frac{A_c \mu}{2} \cos 2\pi (f_c + f_m) t}_{\text{USB}} + \underbrace{\frac{A_c \mu}{2} \cos 2\pi (f_c - f_m) t}_{\text{LSB}}}$$



B.W. = 2 f_m = 2 × Highest frequency of message signal

Power calculations of A.M :

$$P = I_{rms}^2 R \text{ or } V_{rms}^2 / R$$

Assuming AM signal as voltage signal , $V_{rms} = \frac{A_c}{\sqrt{2}}$

$$P_{fc} = \frac{1}{R}\left(\frac{A_c}{\sqrt{2}}\right)^2 = \frac{A_c^2}{2R}$$

$$P_{fc+fm} = \frac{1}{R}\left(\frac{A_c \mu}{2\sqrt{2}}\right)^2 = \frac{A_c^2 \mu^2}{8R}$$

$$P_{fc-fm} = \frac{A_c^2 \mu^2}{8R}$$

$$\text{Total Power} = \frac{A_c^2}{2R}\left(1 + \frac{\mu^2}{2}\right)$$

$$= \frac{A_c^2}{2R} + \frac{A_c^2 \mu^2}{4R}$$

[note: If R is not given in the problem always normalized power is calculated by considering R as 1 Ω]

$$P_t = \underbrace{P_c}_{\text{Carrier Power}} + \underbrace{\frac{P_c \mu^2}{2}}_{\text{Sidebands (USB \& LSB) Power}}$$

$$\text{Modulation Efficiency} = \frac{\text{Power in the sidebands}}{\text{Total Power}}$$

$$\eta = \frac{P_c \mu^2 / 2}{P_c (1 + \mu^2 / 2)} = \frac{\mu^2}{2 + \mu^2}$$

Multi-tone Modulation:

When the message contains more than one frequency, it is called multi-tone modulation.

For 2 - tone modulation

$$m(t) = A_{m1} \cos 2\pi f_{m1} t + A_{m2} \cos 2\pi f_{m2} t \qquad (f_{m2} > f_{m1})$$

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

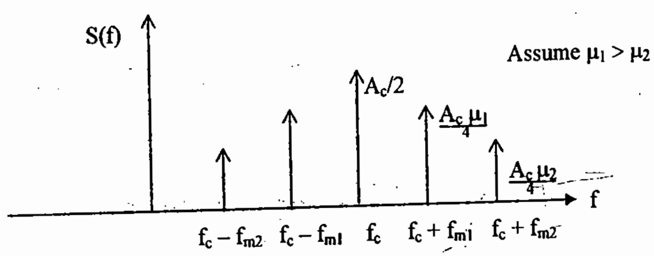
$$= A_c \cos 2\pi f_c t + A_c K_a [A_{m1} \cos 2\pi f_{m1} t + A_{m2} \cos 2\pi f_{m2} t] \cos 2\pi f_c t$$

$$\therefore s(t) = A_c [1 + \mu_1 \cos 2\pi f_{m1} t + \mu_2 \cos 2\pi f_{m2} t] \cos 2\pi f_c t$$

$$= A_c \cos 2\pi f_c t + \frac{A_c \mu_1}{2} [\cos 2\pi (f_c + f_{m1}) t + \cos 2\pi (f_c - f_{m1}) t]$$

$$+ \frac{A_c \mu_2}{2} [\cos 2\pi (f_c + f_{m2}) t + \cos 2\pi (f_c - f_{m2}) t]$$

$$K_a A_{m1} = \mu_1, K_a A_{m2} = \mu_2$$



$$B.W = 2 f_{m2} = 2 \text{ (Highest frequency of message signal)}$$

Normalized power

$$P_{carrier} = A_c^2/2$$

$$P_{fc \pm fm1} = \left(\frac{A_c \mu_1}{2\sqrt{2}}\right)^2 = \frac{A_c^2 \mu_1^2}{8}$$

$$P_{fc \pm fm2} = \left(\frac{A_c \mu_2}{2\sqrt{2}}\right)^2 = \frac{A_c^2 \mu_2^2}{8}$$

$$P_{\text{Upper sideband frequencies}} = \frac{A_c^2 \mu_1^2}{8} + \frac{A_c^2 \mu_2^2}{8}$$

$$= \frac{A_c^2}{8} [\mu_1^2 + \mu_2^2] = P_{\text{Lower sideband frequencies}}$$

$$\therefore P_t = \frac{A_c^2}{2} + \frac{A_c^2}{4} (\mu_1^2 + \mu_2^2) = \frac{A_c^2}{2} \left(1 + \frac{\mu_1^2}{2} + \frac{\mu_2^2}{2}\right)$$

$$= P_c \left(1 + \frac{\mu_t^2}{2}\right) \qquad \text{where } \mu_t^2 = \mu_1^2 + \mu_2^2$$

If the message contains 'n' frequencies, $\mu_t^2 = \mu_1^2 + \mu_2^2 + \ldots\ldots\ldots + \mu_n^2$

$$\eta = \frac{\text{Power in sidebands}}{\text{Total Power}} = \frac{P_c \mu_t^2 / 2}{P_c (1 + \mu_t^2/2)} = \frac{\mu_t^2}{2 + \mu_t^2}$$

Example : Determine η and the percentage of the total power carried by the sidebands of the AM wave for single tone modulation when (a) $\mu = 0.5$ and (b) $\mu = 0.3$.

Sol : For $\mu = 0.5$,

$$\eta = \frac{\mu^2}{2 + \mu^2} \times 100\% = \frac{(0.5)^2}{2 + (0.5)^2} \times 100\% = 11.11\%$$

Hence, only about 11% of the total power is in the sidebands.

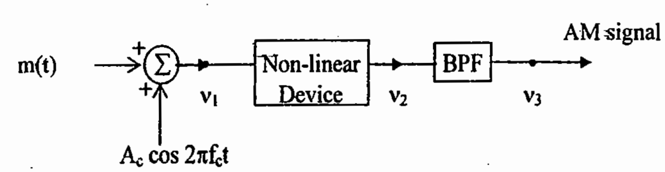
For $\mu = 0.3$,

$$\eta = \frac{(0.3)^2}{2 + (0.3)^2} \times 100\% = 4.3\%$$

Hence, only 4.3% of the total power is the useful power (power in sidebands).

Generation on AM signals :

(1) Square Law Modulator :



$v_1(t) = m(t) + \cos 2\pi f_c t$

$v_2(t) = K_1 v_1(t) + K_2 v_1^2(t)$

$V_0 = K_1 V_i + K_2 V_i^2$

If $K_1 = 0$, it becomes a square law device.

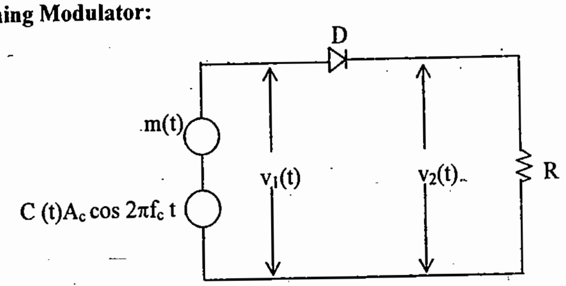
$v_2(t) = K_1 m(t) + K_1 A_c \cos 2\pi f_c t + K_2 m^2(t) + K_2 A_c^2 \cos^2 2\pi f_c t + 2K_2 m(t) A_c \cos 2\pi f_c t$

$v_3(t) = K_1 A_c \cos 2\pi f_c t + 2 K_2 m(t) A_c \cos 2\pi f_c t,$

$= K_1 A_c \left(1 + \frac{2K_2}{K_1} m(t)\right) \cos 2\pi f_c t,$ the other terms are eliminated by the BPF with centre frequency 'f_c'and BW=2W

Amplitude Sensitivity $K_a = \frac{2K_2}{K_1}$

Switching Modulator:



$v_1(t) = m(t) + A_c \cos 2\pi f_c t$; $A_c >> |m(t)|$

$v_2(t) = v_1(t)$; $c(t) > 0$

$= 0$; $c(t) < 0$

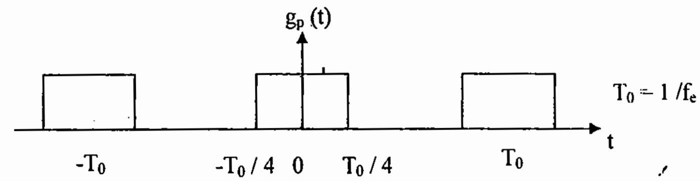
(or)

$v_2(t) = m(t) + A_c \cos 2\pi f_c t$; $c(t) > 0$

$v_2(t) = 0$; $c(t) < 0$

Analytically

$v_2(t) = v_1(t) \times g_p(t)$



Fourier serious fo $g_p(t)$: the periodic functions

$g_p(t) = (1/2) + (2/\pi) \cos(2\pi f_c t) - (2/3\pi) \cos(2\pi\, 3f_c t) + \ldots\ldots$

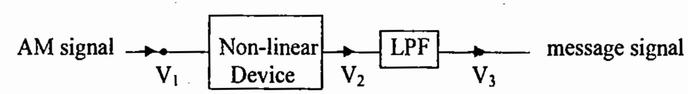
$\therefore v_2(t) = [\, m(t) + A_c \cos 2\pi f_c t \,] \left[(1/2) + (2/\pi) \cos 2\pi f_c t - (2/3\pi) \cos 2\pi\, 3f_c t + \ldots\ldots \right]$

If the above signal is passed through a BPF,

$v_2(t) = (A_c/2) \cos(2\pi f_c t) + (2/\pi)\, m(t) \cos(2\pi f_c t)$

Demodulation of AM signals :

(1) Square Law Demodulator (Detector) :



$V_1 = A_c [1 + K_a m(t)] \cos(2\pi f_c t)$

$V_2 = K_1 A_c \cos 2\pi f_c t + K_1 A_c K_a m(t) \cos 2\pi f_c t + K_2 A_c^2 \cos^2 2\pi f_c t$

$+ K_2 A_c^2 K_a^2 m^2(t) \cos^2 2\pi f_c t + 2 K_2 A_c^2 K_a m(t) \cos^2 2\pi f_c t$

$$V_3 = \frac{K_2 A_c^2}{2} + \frac{2 K_2 A_c^2 K_a m(t)}{2} + \frac{K_2 A_c^2 K_a^2 m^2(t)}{2}$$

$$= \frac{K_2 A_c^2}{2} [\, 1 + 2 K_a m(t)] + \frac{K_2 A_c^2 K_a^2 m^2(t)}{2}$$

The ratio between wanted component to unwanted component is

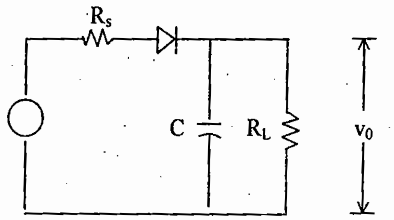
$$\frac{\frac{2 K_2 A_c^2 K_a m(t)}{2}}{\frac{K_2 A_c^2 K_a^2 m^2(t)}{2}} = \frac{2}{K_a m(t)}$$

The ratio should be as high as possible..

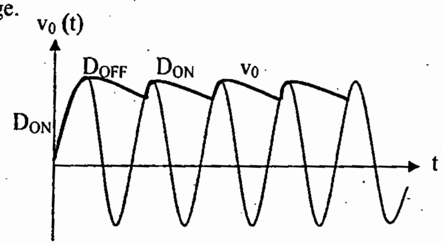
$$\text{Ratio} = \frac{2}{K_a A_m \cos 2\pi f_m t}, \text{ for single tone modulation}$$

$$\text{Ratio (Min.)} = \frac{2}{K_a A_m} = \frac{2}{\mu} \quad \text{(Ratio is high only when } \mu << 1\text{)}$$

(2) Envelope Detector :



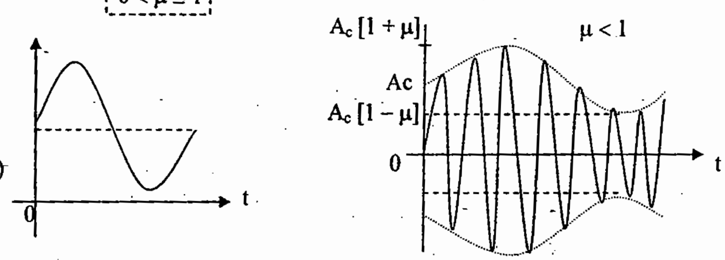
Charging time constant R_sC should be very very low. Discharging time constant R_LC should be very very large.



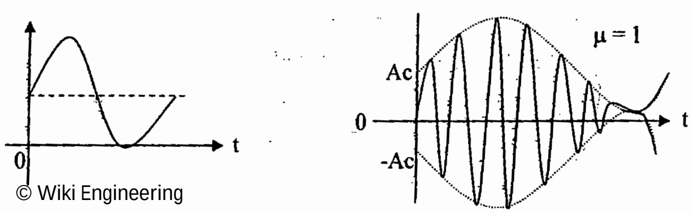
For an input of $A_c [1 + K_a m(t)] \cos 2\pi f_c t$, the output of envelope detector is $A_c + A_cK_a m(t)$. The envelope of the input must be positive to get the exact message signal.

$0 < \mu \leq 1$

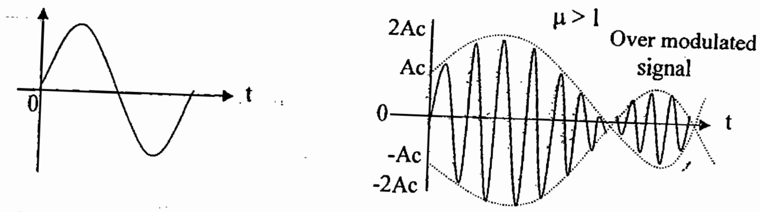
(a)



(b)

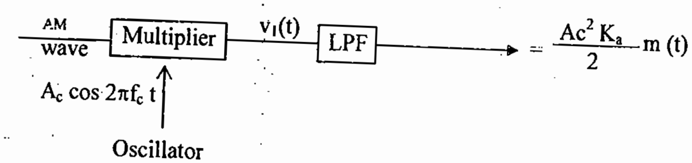


(c)



An Over modulated signal can't be demodulated by a Square law demodulator and Envelope detector.

3. Synchronous Detector:



$$v_1(t) = [A_c \cos 2\pi f_c t + A_c K_a m(t) \cos 2\pi f_c t]\, A_c \cos 2\pi f_c t$$

DSBSC Modulation

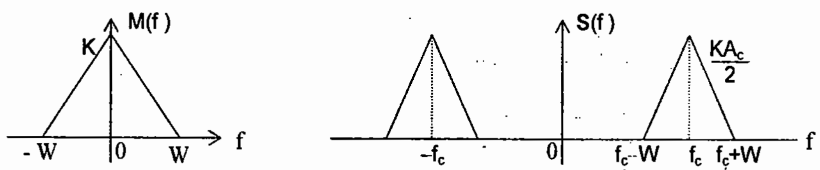
In AM the power required to transmit the carrier is very high when compared to the sidebands. So the modulation efficiency is very less. Always the power in the sidebands should be as high as possible. To increase the modulation efficiency the carrier is suppressed and only the sidebands are transmitted.

$s(t) = A_c\, m(t) \cos 2\pi f_c t + A_c \cos 2\pi f_c t$ [Time domain Equation of AM]

$s(t) = A_c\, m(t) \cos 2\pi f_c t$ [Time domain Equation of DSB, carrier is suppressed]

$s(t) = m(t)\, c(t)$

$S(f) = \dfrac{A_c}{2}\left[M(f - f_c) + M(f + f_c) \right]$ [Frequency domain Equation]



B.W. = 2W = 2 (Highest frequency component of the message)

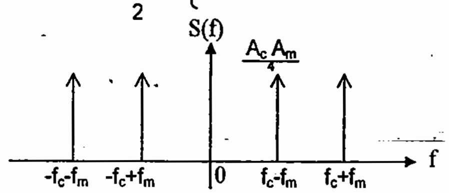
Power required to transmit a DSB wave is very less compared to AM, but the bandwidth is same as AM.

Single-tone modulation of DSB-SC :

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

$s(t) = A_m A_c \cos 2\pi f_c t \cos 2\pi f_m t$

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



B.W. = $2 f_m$

$P_{LSB} = P_{USB} = \dfrac{A_c^2 A_m^2}{8}$; $P_t = \dfrac{A_c^2 A_m^2}{4}$; $\eta = 1$

$\%\ \text{Power Saving} = \dfrac{\text{Power saved}}{\text{Total power}} \times 100$

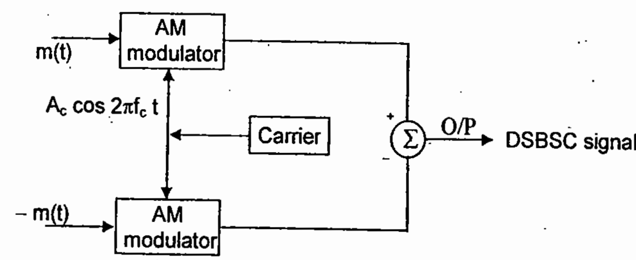
$$= \frac{P_c}{P_c(1 + \mu^2/2)} = \frac{2}{2 + \mu^2} = 1 - \eta_{AM}$$

Generation of DSB-SC signals :

(1) Balanced modulator

(2) Ring modulator

Balanced modulator :

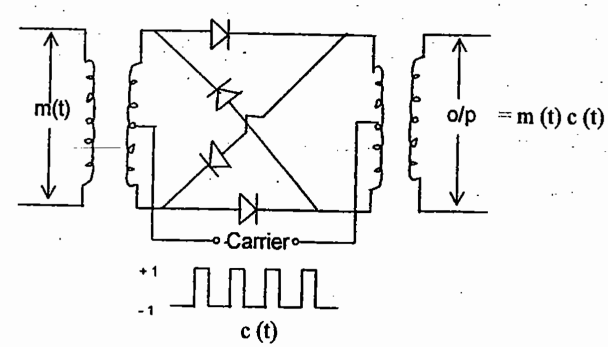


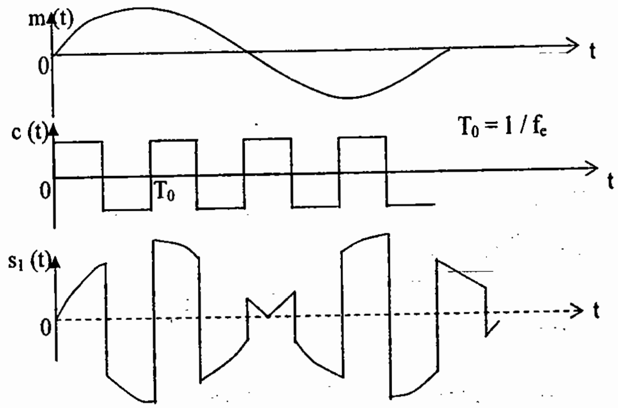
$O/P = A_c[1 + K_a m(t)] \cos 2\pi f_c t - A_c[1 - K_a m(t)] \cos 2\pi f_c t$

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

$= 2 K_a c(t)\, m(t)$

Ring modulator:



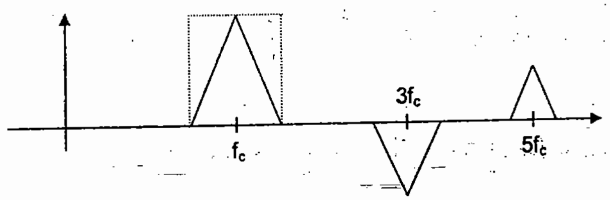


$$c(t) = (4/\pi) \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c t (2n-1)]$$

$$s_1(t) = m(t)\, c(t)$$

$$= (4/\pi)\, m(t) \sum_{n=1}^{\infty} \frac{(-1)^{n-1}}{2n-1} \cos[2\pi f_c t (2n-1)]$$

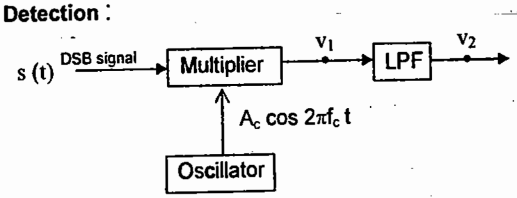
$$= (4/\pi)\, m(t) \left\{\cos 2\pi f_c t - (1/3) \cos 6\pi f_c t + (1/5) \cos 2\pi (5f_c)\, t - \ldots\ldots \right\}$$



By passing the output of Ring modulator $s_1(t)$ through a BPF with centre frequency f_c and bandwidth 2W, we can get DSB signal, $s(t) = 4/\pi\, m(t) \cos(2\pi f_c t)$

Demodulation of DSB signals:

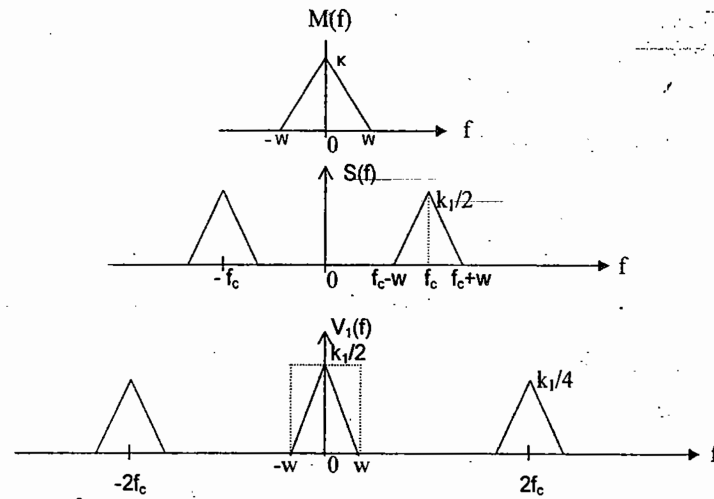
Coherent Detection :



$$s(t) = A_c \cos(2\pi f_c t)\, m(t)$$

$$v_1(t) = A_c^2 \cos^2 (2\pi f_c t)\, m(t)$$

$$v_1(t) = \frac{A_c^2}{2}\left\{1 + \cos 4\pi f_c t\right\} m(t), \quad v_2(t) = \frac{A_c^2}{2} m(t)$$



$$v_1(t) = A_c^2\, m(t) \cos 2\pi f_c t \times \cos(2\pi f_c t + \phi)$$

$$= \frac{A_c^2\, m(t)}{2}\left\{\cos\phi + \cos(4\pi f_c t + \phi)\right\}$$

$$v_2(t) = \frac{A_c^2\, m(t)}{2} \cos\phi$$

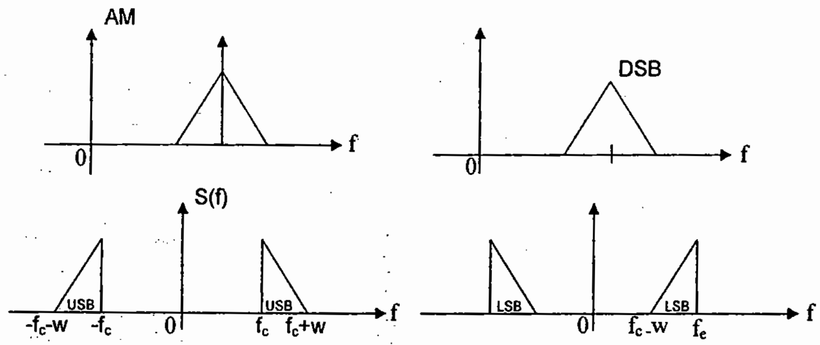
$$\phi = 0 \Rightarrow v_2(t) = \frac{A_c^2\, m(t)}{2}$$

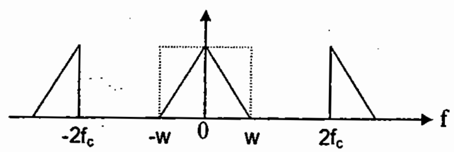
$\phi = 90^0 \Rightarrow v_2(t) = 0$ (Quadrature Null effect)

Synchronization circuits are necessary are to overcome the Quadrature Null Effect. So the complexity of the receiver is increased.

Single Sideband (SSB) Modulation

In order to reduce the bandwidth required to transmit the signal SSB modulation is used. In this technique only one sideband is transmitted (either USB or LSB). So the bandwidth and power required to transmit the signal, is reduced.





$$s(t) = m(t)\, A_c \cos 2\pi f_c t$$

$$= A_m A_c \cos 2\pi f_m t \; \cos 2\pi f_c t, \text{ For single tone modulation}$$

$$s(t) = \frac{A_c A_m}{2} \cos 2\pi (f_c + f_m)\, t + \frac{A_c A_m}{2} \cos 2\pi (f_c - f_m)\, t$$

$$s(t) = \frac{A_c A_m}{2} \cos 2\pi (f_c + f_m)\, t \qquad \text{(If USB is transmitted)}$$

$$s(t) = \frac{A_c A_m}{2} \cos 2\pi (f_c - f_m)\, t \qquad \text{(If LSB is transmitted)}$$

$$s(t) = \frac{A_c A_m}{2} \cos 2\pi (f_c \pm f_m)\, t$$

$$= \frac{A_c A_m}{2} \left\{ \cos 2\pi f_c t \; \cos 2\pi f_m t \mp \sin 2\pi f_c t \; \sin 2\pi f_m t \right\}$$

The generalized equation is

$$s(t) = \frac{A_c}{2} \left[m(t) \cos 2\pi f_c t \mp \hat{m}(t) \sin 2\pi f_c t \right]$$

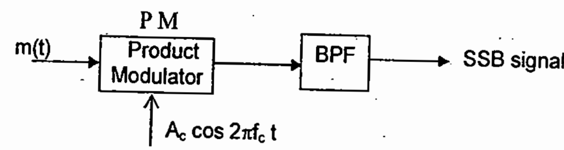
$$P_t = \frac{A_c^2 A_m^2}{8}$$

$$\text{Power saving} = \frac{P_c + P_{USB \text{ or } LSB}}{P_t} = \frac{P_c + \tfrac{1}{2}\, P_c\, \mu^2/2}{P_c (1 + \mu^2/2)} = \frac{(4 + \mu^2)}{2\,(2 + \mu^2)}$$

Generation of SSB signals :

(1) Frequency Discrimination Method
(2) Phase Discrimination Method

Frequency discrimination Method :

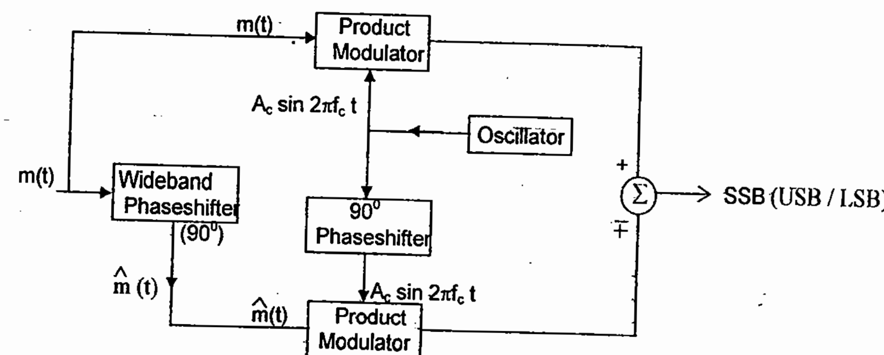


The output of PM is a DSB signal. If the DSB signal is passed through a bandpass filter, the upper sideband or lower sideband is suppressed.

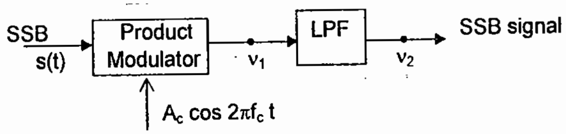
If the passband is from f_c to f_c+ W, we will get the USB.

Phase Discrimination method :

$$s(t) = \frac{A_c}{2} m(t) \cos 2\pi f_c t \mp \frac{A_c}{2} \hat{m}(t) \sin 2\pi f_c t$$



Demodulation of SSB signals :



$$s(t) = \frac{A_c}{2} m(t) \cos 2\pi f_c t \mp \frac{A_c}{2} \hat{m}(t) \sin 2\pi f_c t$$

$$v_1(t) = \frac{A_c^2}{4} [1 + \cos 4\pi f_c t]\, m(t) \mp \frac{A_c^2}{4} \hat{m}(t) \sin 4\pi f_c t$$

$$v_2(t) = \frac{A_c^2}{4} m(t)$$

Consider the locally generated signal as $A_c \cos(2\pi f_c t + \phi)$,

$$v_2(t) = \frac{A_c^2}{4} \left(\cos\phi\, m(t) \mp \hat{m}(t) \sin\phi\right)$$

$$\phi = 0^0, \quad v_2(t) = \frac{A_c^2}{4} m(t)$$

$$\phi = 90^0, \quad v_2(t) = (A_c^2/4)\, \hat{m}(t)$$

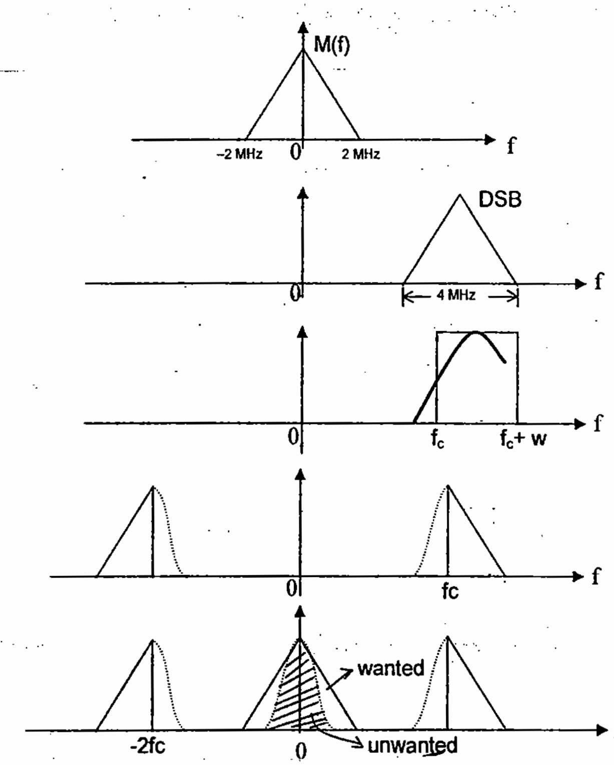
$\neq 0$ [So no Quadrature Null effect in the case of SSB, which is a major advantage over DSB]

Vestigial sideband (VSB) modulation

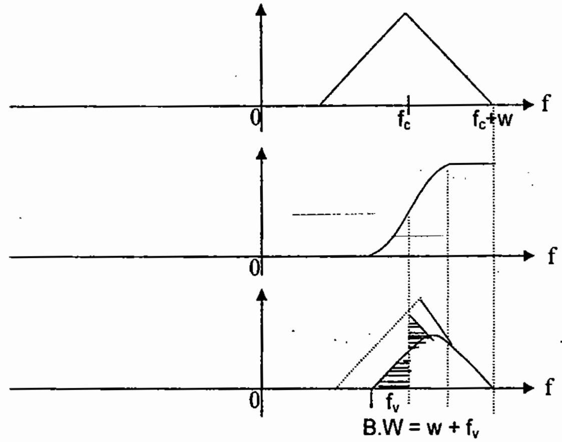
This is mainly used for the transmission of video signals.

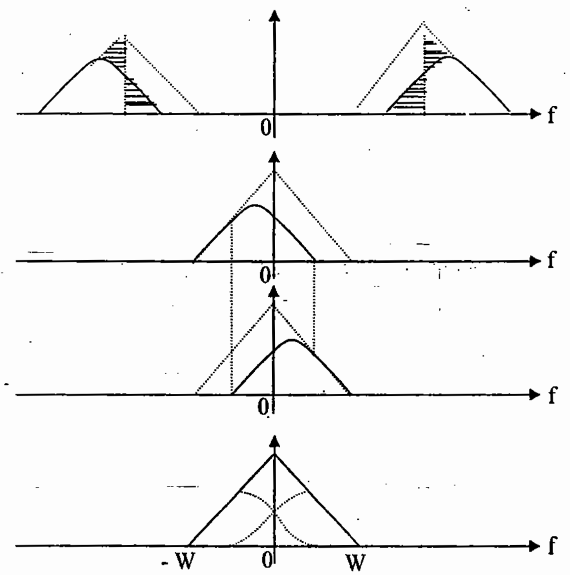
(1) Video signal has significant low frequency components.

(2) Highest frequency varies from 2 MHz to 4 MHz.



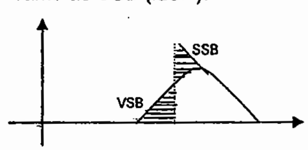
Noise affects the video signal especially at low frequencies.



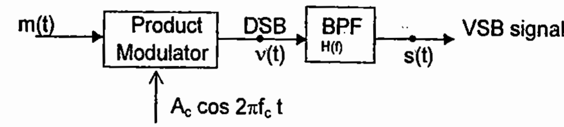


Power required to transmit a VSB signal is same as SSB (ideal).

B.W. = W + f_v



Generation of VSB signal :



$$v(t) = A_c\, m(t) \cos 2\pi f_c t$$

$$s(t) = v(t) \otimes h(t)$$

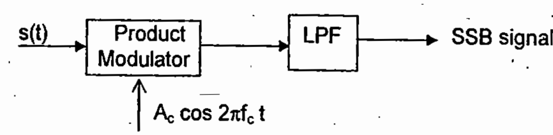
$$S(f) = V(f) \,.\, H(f)$$

$$= \frac{A_c}{2} \left[M(f - f_c) + M(f + f_c) \right] H(f)$$

$$S(f - f_c) = \frac{A_c}{2} \left[M(f - 2f_c) + M(f) \right] H(f - f_c)$$

$$S(f + f_c) = \frac{A_c}{2} \left[M(f + 2f_c) + M(f) \right] . H(f + f_c)$$

Demodulation of VSB signal :



The output of PM is $A_c\, s(t) \cos 2\pi f_c t$.

$$\xrightarrow{F.T} \frac{A_c}{2} \left[S(f - f_c) + S(f + f_c) \right]$$

$$= \frac{A_c^2}{4} \left\{ [M(f - 2f_c) + M(f)]\; H(f - f_c) + [M(f + 2f_c) + M(f)]\; H(f + f_c) \right\}$$

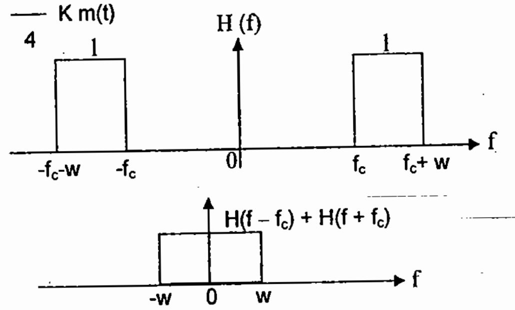
The output of LPF is

$$= \frac{A_c^2}{4} \left[M(f)\; H(f - f_c) \;+\; M(f)\; H(f + f_c) \right]$$

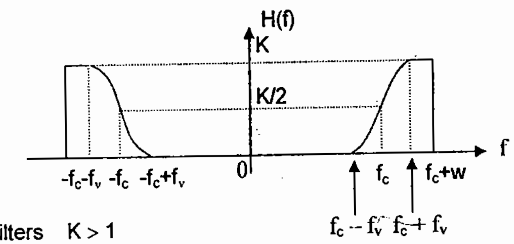
$$= \frac{A_c^2}{4}\; M(f) \left[H(f - f_c) \;+\; H(f + f_c) \right]$$

To get the exact message signal $H(f-f_c) + H(f+f_c)$ should be a constant (K).

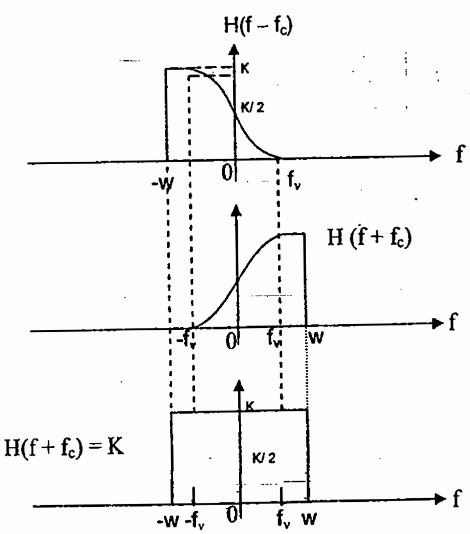
Output of LPF $= \dfrac{A_c^2}{4} K\, m(t)$



For an ideal BPF, $H(f-f_c) + H(f+f_c) = 1$.



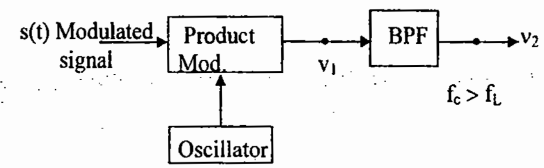
For active filters K > 1
For passive filters K < 1



$H(f-f_c) + H(f+f_c) = K$

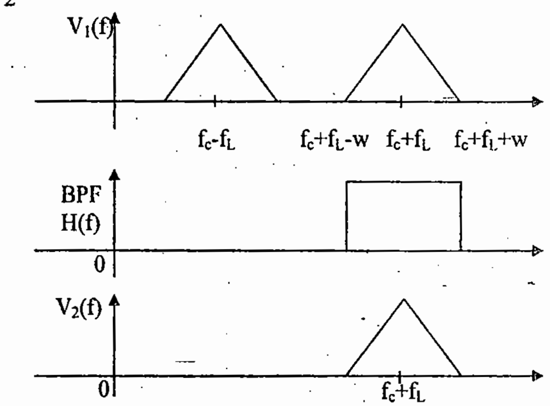
Mixer :

Mixer is a device which is used to change the carrier frequency of a modulated signal.

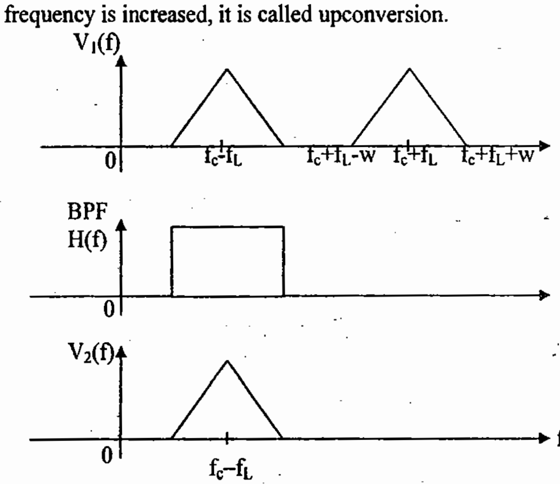


$s(t) = m(t) \cos 2\pi f_c t$

$$v_1(t) = \frac{m(t)}{2}\left\{\cos 2\pi (f_c + f_L)\, t + \cos 2\pi (f_c - f_L)\, t\right\}$$

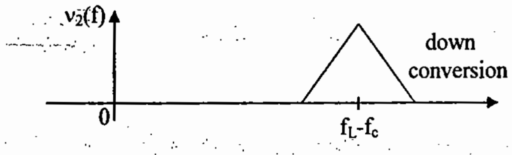


When the carrier frequency is increased, it is called upconversion.

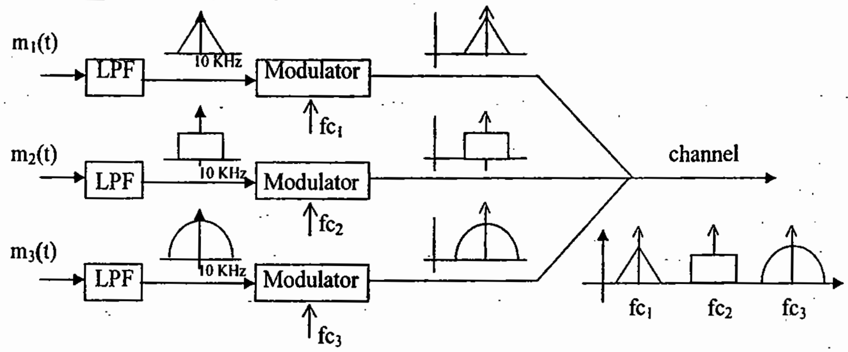


If the carrier frequency is decreased, it is said to be downconverted.

If $f_L > f_c$



Frequency Division Multiplexing (FDM) :

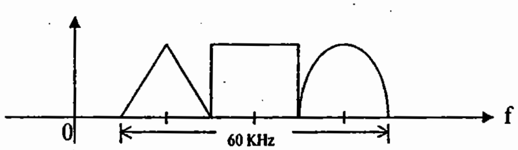


The carrier frequencies must be selected in such a way that there should not be any interference.

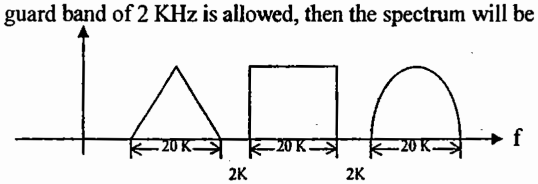
To avoid interference,

$fc_2 \geq fc_1 + 20$ KHz

$fc_3 \geq fc_2 + 20$ KHz

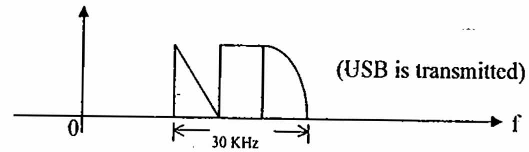


If a guard band of 2 KHz is allowed, then the spectrum will be

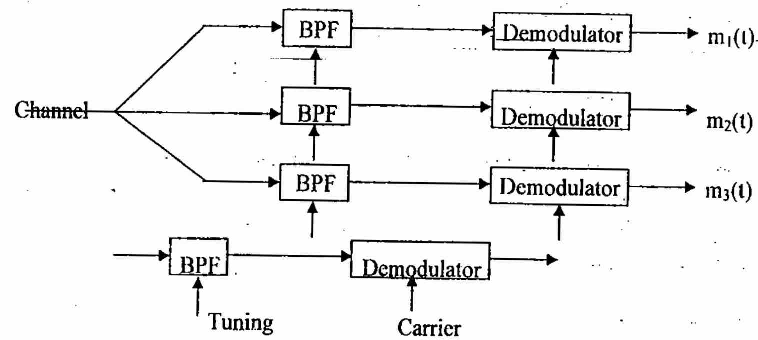


B.W. = 64 KHz

If only SSB is used, then the spectrum of the combined signal is



FDM Receiver :



By tuning we can change the resonant frequency to carrier frequency of the required channel.

(B) Angle Modulation:

Angle modulation is defined as the process in which the angle of the carrier (either frequency or phase) is varied linearly according to the message signal. So there are two types of angle modulation.

(1) Frequency Modulation
(2) Phase Modulation

Phase Modulation :

Changing the phase according to the message signal is called Phase Modulation.

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

$$\phi(t) = K_p m(t)$$

K_p = Phase sensitivity of the modulator (radians / volt)

For single tone modulation

$\phi(t) = K_p A_m \cos 2\pi f_m t$ $\qquad \phi_{max} = K_p A_m$ = Phase Deviation

Frequency Modulation :

Changing the frequency of the carrier according to the message signal is called Frequency Modulation.

$$f_i(t) = f_c + K_f m(t)$$

K_f = Frequency sensitivity (Hertz / Volt)

For single tone modulation

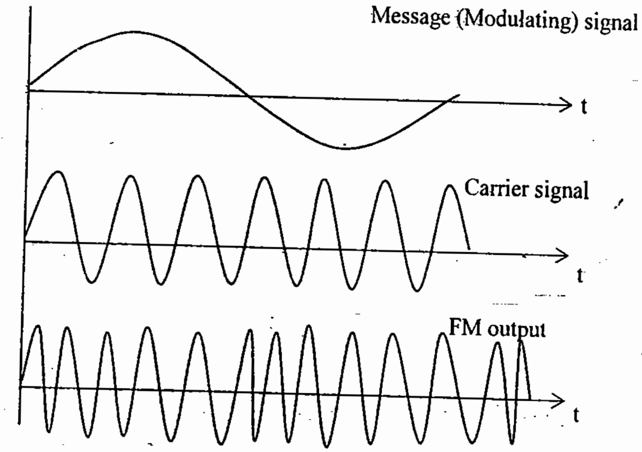
$$f_i(t) = f_c + K_f A_m \cos 2\pi f_m t$$

$$f_{i,\,max} = f_c + K_f A_m$$

$$f_{i,\,min} = f_c - K_f A_m$$

$\Delta f = K_f A_m$ = Frequency deviation

Carrier Swing or Total variation of carrier frequency $= 2\,\Delta f$



Let $s(t) = A_c \cos \theta(t)$ be the FM wave

$$\theta(t) = 2\pi f_i t$$

$$d\theta(t) / dt = 2\pi f_i$$

$$\frac{1}{2\pi} \frac{d\theta(t)}{dt} = f_i$$

$$\theta(t) = 2\pi \int f_i(t)\, dt$$

$$\theta(t) = 2\pi \int [\, f_c + K_f m(t) \,]\, dt$$

$$\theta(t) = 2\pi f_c t + 2\pi K_f \int_0^t m(t)\, dt$$

$$s(t) = A_c \cos [\, 2\pi f_c t + 2\pi K_f \int m(t)\, dt \,] \longrightarrow \text{FM}$$

$$s(t) = A_c \cos [\, 2\pi f_c t + K_p m(t) \,] \longrightarrow \text{PM}$$

For a single tone modulation

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

$$s(t) = A_c \left(\cos \left(2\pi f_c t + \frac{2\pi K_f A_m}{2\pi f_m} \sin 2\pi f_m t \right) \right)$$

$$= A_c \left(\cos \left(2\pi f_c t + \frac{K_f A_m}{f_m} \sin 2\pi f_m t \right) \right)$$

$$\text{Modulation index of FM} = \beta = \frac{K_f A_m}{f_m} = \frac{\Delta f}{f_m}$$

$\beta << 1$, Narrow band FM

$\beta >> 1$, Wide band FM

Narrow Band FM :

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

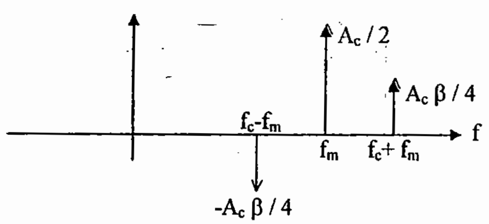
$$= A_c \left[\cos 2\pi f_c t \cdot \cos(\beta \sin 2\pi f_m t) - \sin 2\pi f_c t \cdot \sin(\beta \sin 2\pi f_m t)\right]$$

$$\cong A_c \cos 2\pi f_c t - A_c \beta \sin 2\pi f_c t \sin(2\pi f_m t), \text{ for } \beta << 1$$

$$\cong A_c \cos 2\pi f_c t - \frac{A_c \beta}{2}\left[\cos 2\pi (f_c - f_m) t - \cos 2\pi (f_c + f_m) t\right]$$

Spectrum of NBFM :

$$s(t) = A_c \cos 2\pi f_c t + \frac{A_c \mu}{2} \cos 2\pi (f_c + f_m) t + \frac{A_c \mu}{2} \cos 2\pi (f_c - f_m) t$$

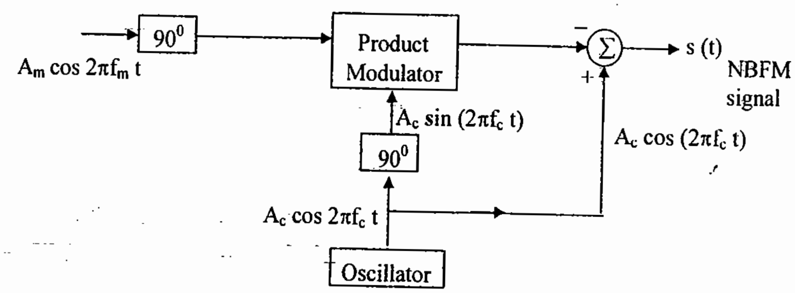


B.W of NBFM $= 2 f_m$

The spectrum of AM and FM are identical except that the spectral component at $f_c - f_m$ is 180^0 out of phase.

$$P_t = P_c (1 + \beta^2 / 2)$$

Generation of NBFM signal :



Wideband FM :

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

Bessel function of order 'n' is given by

$$J_n(x) = (1/2\pi)\int e^{j(x \sin\theta - n\theta)}\, d\theta$$

Properties :

$$J_n(x) = (-1)^n J_{-n}(x)$$

$$\sum_{n=-\infty}^{\infty} J_n^2(x) = 1$$

$$g_p(t) = \sum_{n=-\infty}^{\infty} C_n e^{j2\pi nt/To}$$

$$C_n = (1/T_0)\int_{-To/2}^{To/2} g_p(t)\, e^{-j2\pi nt/To}\, dt$$

$$\cos\theta = \text{Re}[e^{j\theta}]$$

$$s(t) = A_c \text{Re}\, e^{j(2\pi fc\, t + \beta \sin 2\pi fm\, t)}$$

$$= A_c \text{Re}\, e^{j2\pi fc\, t} \cdot e^{j\beta \sin 2\pi fm\, t}$$

$$= A_c \text{Re}\, e^{j2\pi fc\, t} \cdot \sum_{n=-\infty}^{\infty} C_n e^{+j2\pi nfm\, t}$$

$$C_n = (1/T_0)\int_{-1/2fm}^{1/2fm} e^{j\beta \sin 2\pi fm\, t} \cdot e^{-j2\pi nfm\, t}\, dt$$

$$C_n = f_m \int_{-1/2fm}^{1/2fm} e^{j(\beta \sin 2\pi fm\, t - 2\pi nfm\, t)}\, dt$$

Compare this with standard form of Bessel function,

$$\theta = 2\pi f_m t$$

$$C_n = \frac{f_m}{2\pi f_m} \int_{-\pi}^{\pi} e^{j(\beta \sin\theta - n\theta)}\, d\theta$$

$$C_n = J_n(\beta)$$

$$s(t) = A_c \,\mathrm{Re}\, e^{j2\pi f_c t} \cdot \sum_{n=-\infty}^{\infty} J_n(\beta)\, e^{j2\pi n f_m t}$$

$$= A_c \,\mathrm{Re} \sum_{n=-\infty}^{\infty} J_n(\beta)\, e^{j2\pi f_c t} \cdot e^{j2\pi n f_m t}$$

$$= A_c \sum_{n=-\infty}^{\infty} J_n(\beta)\, e^{j2\pi (f_c + n f_m) t}$$

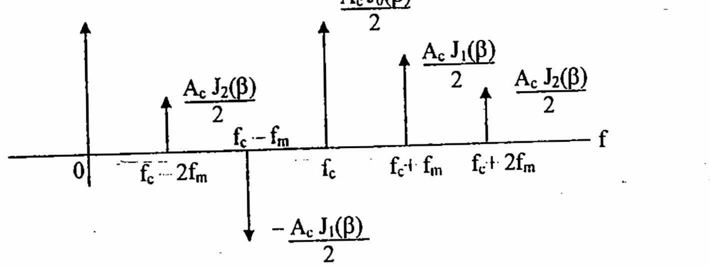
$$= A_c \,\mathrm{Re} \sum_{n=-\infty}^{\infty} J_n(\beta) \cos[2\pi (f_c + n f_m) t] \qquad n = 0, \pm1, \pm2, \ldots\ldots \pm\infty$$

$$s(t) = A_c J_0(\beta) \cos 2\pi f_c t + A_c J_1(\beta) \cos 2\pi (f_c + f_m) t + A_c J_{-1}(\beta) \cos 2\pi (f_c - f_m) t +$$

$$A_c J_2(\beta) \cos 2\pi (f_c + 2f_m) t + A_c J_{-2}(\beta) \cos 2\pi (f_c - 2f_m) t + \ldots\ldots\ldots$$

$$s(t) = A_c J_0(\beta) \cos 2\pi f_c t + A_c J_1(\beta) \{\cos 2\pi (f_c + f_m) t - \cos 2\pi (f_c - f_m) t\}$$

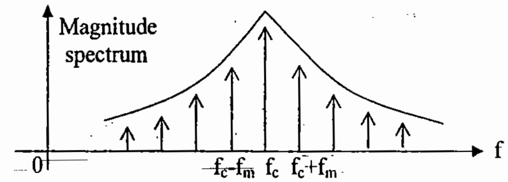
$$+ A_c J_2(\beta) \{\cos 2\pi (f_c + 2f_m) t + \cos 2\pi (f_c - 2f_m) t\}$$



Theoretical bandwidth of a WBFM is '∞'.

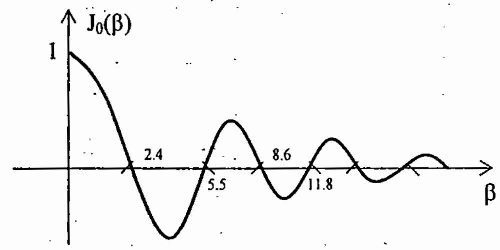
Characteristics of a WBFM signal :

(1) WBFM spectrum consists of carrier and infinite number of sidebands, each separated by f_m.



(2) The amplitudes of the spectral components depends on the Bessel function coefficients $J_n(\beta)$ which decease as 'n' increases. So the amplitudes of the spectral components also decreases on both sides of the carrier.

(3) In WBFM spectrum amplitude of carrier component depends on $J_0(\beta)$ and hence on modulation index β.



The Bessel function coefficient $J_0(\beta) = 0$, when $\beta = 2.4, 5.5, 8.6, 11.8, \ldots..$ For these values of β, the amplitude of the carrier component in the spectrum is zero and the modulation efficiency is 1.

(4)

$$P_{fc} = P_{carrier} = \frac{A_c^2 J_0^2(\beta)}{2}$$

$$P_{fc+fm} = \frac{A_c^2 J_1^2(\beta)}{2}$$

$$P_{fc-fm} = \frac{A_c^2 J_1^2(\beta)}{2}$$

$$P_{1^{st}\ \text{order sideband}} = A_c^2 J_1^2(\beta)$$

$$P_{2^{nd}\ \text{order sideband}} = A_c^2 J_2^2(\beta)$$

$$P_{n^{th}\ \text{order sideband}} = A_c^2 J_n^2(\beta)$$

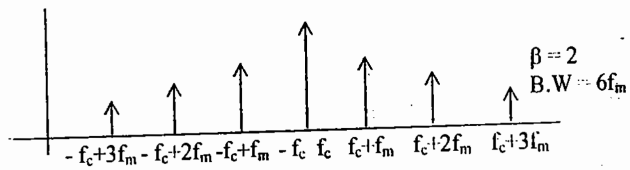
$$\text{Total Power} = J_0^2(\beta) + \frac{A_c^2}{2} J_1^2(\beta) + \frac{A_c^2}{2} J_2^2(\beta) + \frac{A_c^2}{2} J_{-2}^2(\beta) + \frac{A_c^2}{2} \ldots\ldots$$

$$= \frac{A_c^2}{2} \sum_{n=-\infty}^{\infty} J_n^2(\beta) = \frac{A_c^2}{2} = \text{Unmodulated Carrier Power}$$

The total power is independent of modulation index. AM takes more power compared to FM for the same message and carrier.

Calculation of practical B.W of WBFM using Carson's rule :

Theoretical bandwidth of FM is '∞' but practically the signal is bandlimited using Carson's rule by passing the FM signal through BPF. Power spectral density gives how the power is distributed among different frequency components.
Carson stated that only certain frequencies will have significant amplitudes.



By using BPF, the higher components of the FM signal are suppressed so that the power in the remaining sidebands is 99% of the total power.

Carson has proved that the number of sidebands having significant amplitudes containing 99% of the total power is $\beta+1$.

$$B.W = 2(\beta + 1) f_m$$

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

Example : An angle-modulated signal with carrier frequency $\omega_c = 2\pi \times 10^5$ is described by the equation

$$\varphi_{EM} = 10\cos(\omega_c t + 5\sin 3000t + 10\sin 2000\pi t)$$

(a) Find the power of the modulated signal.
(b) Find the frequency deviation.
(c) Find the deviation ratio β.
(d) Find the phase deviation $\Delta\phi$.
(e) Estimate the bandwidth of $\varphi_{EM}(t)$.

Sol : The signal bandwidth is the highest frequency in m(t) (or its derivative). In this case $B = 2000\pi / 2\pi = 1000$ Hz.

(a) The carrier amplitude is 10, and the power is

$$P = 10^2/2 = 50$$

(b) To find the frequency deviation Δf, find the instantaneous frequency ω_i, given by

$$\omega_i = \frac{d}{dt}\theta(t) = \omega_c + 15{,}000\cos 3{,}000\,t + 20{,}000\pi\cos 2000\pi\,t$$

The carrier deviation is $15{,}000\cos 3{,}000\,t + 20{,}000\pi\cos 2000\pi\,t$. The sinusoids will add in phase at some point, and the maximum value of this expression is $15{,}000 + 20{,}000\pi$. This is the maximum carrier deviation $\Delta\omega$. Hence,

$$\Delta f = \frac{\Delta\omega}{2\pi} = 12{,}387.32 \text{ Hz}$$

(c)
$$\beta = \frac{\Delta f}{B} = \frac{12{,}387.32}{1000} = 12.387$$

(d) The angle $\theta(t) = \omega t + (5\sin 3000t + 10\sin 2000\pi t)$. The phase deviation is the maximum value of the angle inside the parenthesis, and is given by $\Delta\phi = 15$ rad.

(e) $B_{EM} = 2(\Delta f + B) = 26{,}774.65$ Hz

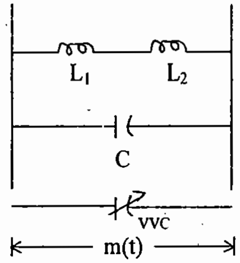
Generation of WBFM signal :

(1) Direct Method
(2) Indirect Method (or) Armstrong Method

Direct Method :

A voltage controlled oscillator is used to generate FM signal.

For a Hartley oscillator,



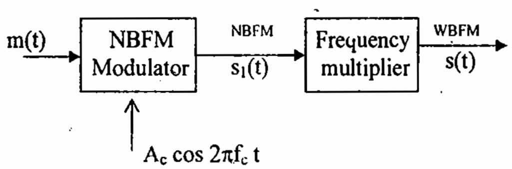
$$f = \frac{1}{2\pi\sqrt{L_1 + L_2}\ C}$$

$$f = \frac{1}{2\pi\sqrt{(L_1 + L_2)(C + C_1)}}$$

C_1 is (WC) voltage variable capacitor and depends upon m (t)

Indirect Method :

In this method, NBFM signal is converted into WBFM signal.



$$s(t) = A_c \cos\left[2\pi f c_1 t + \beta_1 \sin 2\pi f_m t\right]$$

$$s(t) = A_c \cos\left[n 2\pi f c_1 t + n\beta_1 \sin 2\pi f_m t\right]$$

$$= A_c \cos\left[2\pi f c_2 t + \beta_2 \sin 2\pi f_m t\right]$$

where $f_{c2} = n f c_1$; $\beta_2 = n\beta_1$

Demodulation of FM signals :

- **Slope Detector :**

Figure shows the schematic diagram for a *single-ended- slope detector*, which is the simplest form of tuned-circuit frequency discriminator. The single-ended- slope detector has the most nonlinear voltage-versus frequency characteristics. However, its circuit operation is basic to all tuned-circuit frequency discriminators.

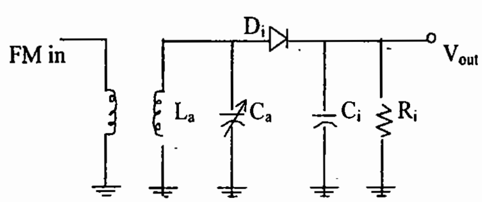


Fig.(a)

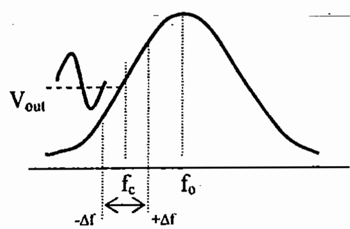


Fig.(b)

In fig.(a), the tuned circuit (L_a and C_a) produces an output voltage that is proportional to the input frequency. The maximum output voltage occurs at the resonant frequency of the tank circuit (f_o), and its operation decreases proportionately as the input frequency deviates above or below f_o. The circuit is designed so that the IF center frequency (f_c) falls in the center of the most linear portion of the voltage-versus-frequency curve, as shown in fig.(b). When the intermediate frequency deviates above f_c, the output voltage increases; when the intermediate frequency deviates below f_c, the output voltage decreases. Therefore, *the tuned circuit converts frequency variations to amplitude variations* (FM-to-AM conversion).

D_i, C_i, and R_i make up a simple peak detector that converts the amplitude variations to an output voltage that varies at a rate equal to that of the input frequency changes and whose amplitude is proportional to the magnitude of the frequency changes.

Balanced Slope Detector :

The figure shown below is the schematic diagram for a *balanced slope detector*.
A single-ended slope detector is a tuned-circuit frequency discriminator, and a balanced slope detector is simply two single-ended slope detectors connected in parallel and fed 180° out of phase. The phase inversion is accomplished by center tapping the tuned secondary windings of transformer T_1.

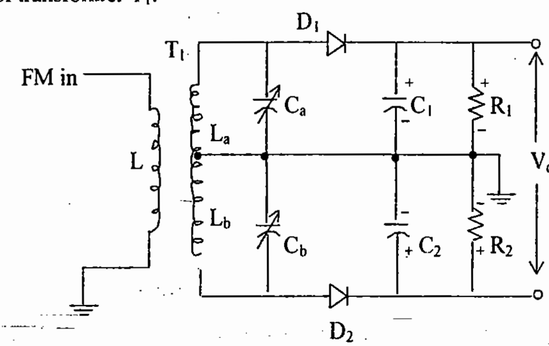


Fig.(a)

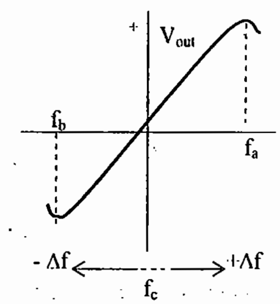


Fig.(b)

In Fig.(a), the tuned circuits (L_a, C_a, and L_b, C_b) perform the FM-to-AM conversion, and the balanced peak detectors (D_1, C_1, R_1 and D_2, C_2, R_2) remove the information from AM envelope. The top tuned circuit (L_a and C_a) is tuned to a frequency (f_a) that is above the IF center frequency (f_c) by approximately $1.33 \times \Delta f$ (for the FM broadcast and this is approximately 1.33×7.5 kHz = 100 kHz). The lower tuned circuit (L_b and C_b) is tuned to a frequency (f_b) that is below the IF center frequency by an equal amount.

The output voltage from each tuned circuit is proportional to the input frequency, and each output is rectified by its respective peak detector. Therefore, the closer the input frequency is to the tank-circuit resonant frequency, the greater the tank-circuit output voltage. The IF center frequency falls exactly halfway between the resonant frequencies of the two tuned circuits. Therefore at the IF center frequency, the output voltages from the two tuned circuits are equal in amplitude but oppose in polarity. Consequently, the rectified output voltage across R_1 and R_2, when added, produce a differential output voltage $V_{out} = 0$ V. When the IF deviates above resonance, the top tuned circuit produces a higher output voltage than the

lower tank circuit and V_{out} goes positive. When the IF deviates below resonance, the output voltage from the lower tank circuit is larger than the output voltage from the upper tank circuit and V_{out} goes negative. The output-versus-frequency response curve is shown in fig.(b).
The slope detector is simplest FM detector. It has several disadvantages, which include poor linearity, difficulty in tuning, and lack of provisions for limiting.

Foster-Seeley Discriminator :

A *Foster-Seeley discriminator* (phase shift discriminator) is a tuned-circuit frequency discriminator whose operation is very similar to that of the balanced slope detector. The schematic diagram for a Foster-Seeley discriminator is shown in fig.(a).

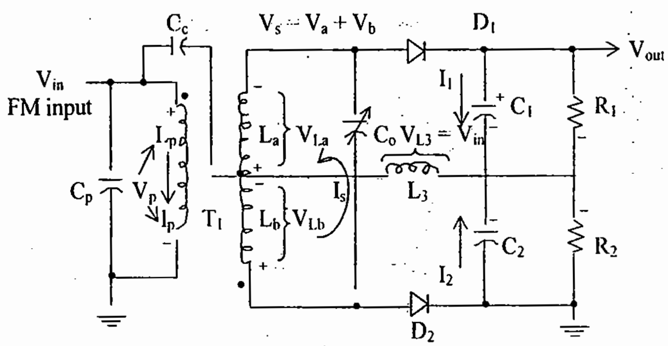


Fig.1.(a)

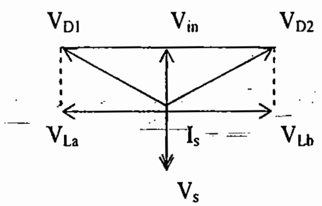


Fig.(b)

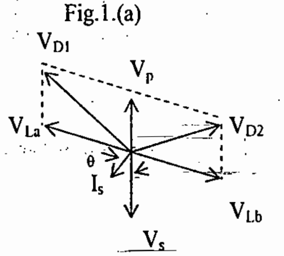


Fig.(c)

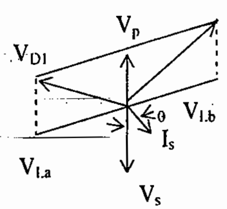


Fig.(d)

The capacitance value for C_c, C, and C_2 are chosen such that they are short circuits for the IF center frequency. Therefore, the right side of L_3 is at ac ground potential, and the IF signal (V_{in}) is fed directly (in phase) across L_3 (V_{L3}). The upcoming IF is inverted 180^0 by transformer T_1 and divided equally between L_a and L_b. At the resonant frequency of the second tank circuit (the IF center frequency), the secondary current (I_s) is in phase with the total secondary voltage (V_s) and 180° out of phase with V_{L3}. Also, due to loose coupling, the primary of T_1 acts as an inductor, and the primary current I_p is 90° out of phase with V_{in}, and, because magnetic induction depends on primary current, the voltage induced in the secondary is 90° out of phase with V_{in} (V_{L3}).The voltage across the top diode (V_{D1}) is the vector sum of V_{L3} and V_{La} and the voltage across the bottom diode V_{D2} is the vector sum of V_{L3} and V_{Lb}. The corresponding vector diagrams are shown in Fig.(b).

The figure shows that the voltage across the diodes D_1 and D_2 are equal. Therefore, at resonance, I_1 and I_2 are equal and C_1 and C_2 charge to equal magnitude voltages except with opposite polarities. Consequently, $V_{out} = V_{C1} - V_{C2} = 0$ V. When the IF goes above resonance ($X_L > X_C$), the secondary tank circuit impedance becomes inductive, and the secondary current lags the secondary voltage by some angle θ, which is proportional to the magnitude of the frequency deviation. The corresponding phasor diagram is shown in Fig.(c). The figure shows that the vector sum of the voltage across D_1 is greater than the vector sum of the voltages across D_2. Consequently, C_1 charges while C_2 discharges and V_{out} goes positive. When the IF goes below resonance ($X_L < X_C$), the secondary current leads the secondary voltage by some angle θ, which is again proportional to the magnitude of the change in frequency. The corresponding phasors are shown in Fig.(d). It can be seen that the vector sum of the voltages across D_1 is now les than the vector sum of the voltage across D_2. Consequently, C_1 discharges while C_2 charges and V_{out} goes negative. A Foster-Seeley discriminator is tuned by injecting a frequency equal to the IF center frequency and tuning C_o for 0 volts. As shown in figure.1, *the output voltage from a Foster-Seeley discriminator is directly proportional to the magnitude and direction of the frequency deviation.*

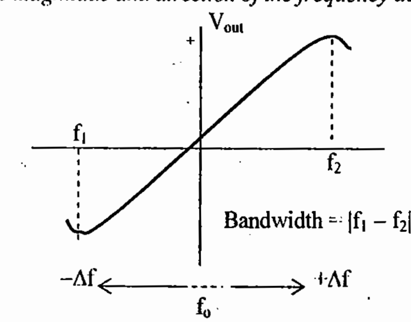


Fig.2

Fig.2 shows a typical voltage-versus-frequency response curve for a Foster-Seeley discriminator. It can be seen that the output voltage-versus-frequency deviation curve is more linear than that of a slope detector, and because there is only one tank circuit, it is easier to tune. For distorsionless demodulation, the frequency deviation should be restricted to the linear portion of the secondary tuned circuit frequency response curve. As with the slope detector, a Foster-Seeley discriminator responds to amplitude as well as frequency variations and, therefore, must be preceded by separate limiter circuit.

Ratio Detector :

The ratio detector has one major advantage over the slope detector and Foster-Seeley discriminator for FM demodulation: A ratio detector is relatively immune to amplitude variations in its input signal. The schematic diagram for a ratio detector is shown in fig.(a).

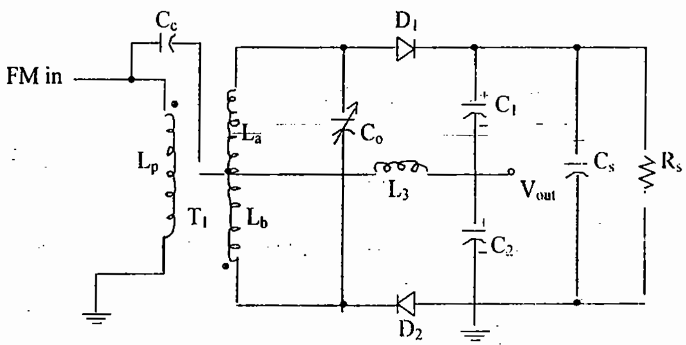


Fig.(a)

As with the Foster-Seeley discriminator, a ratio detector has a single tuned circuit in the transformer secondary. Therefore, the operation of a ratio detector is similar to that of the Foster-Seeley discriminator. In fact, the voltage vectors for D_1 and D_2 are identical. However, with the ratio detector, one diode is reversed (D_2), and current (I_d) can flow around the outermost loop of the circuit. Therefore, after several cycles of the input signal, shunt capacitor C_s charges to approximately the peak voltage across the secondary winding of T_1. The reactance of C_s is low, and R_s simply provides a dc path for diode current. Therefore, the time constant for R_s and C_s is sufficiently long so that rapid changes in the amplitude of the input signal due to thermal noise or other interfering signals are shorted to ground and have no effect on the average voltage across C_s. Consequently, C_1 and C_2 charge and discharge proportional to frequency changes in the input signal and are relatively immune to amplitude variations. Also, the output voltage from a ratio detector is taken with respect to ground, and for the diode polarities shown in fig.(a), the average output voltage is positive. At resonance, the output voltage is divided equally between C_1 and C_2 and distributed as the input frequency is deviated above and below resonance. Therefore, changes in V_{out} are due to changing ratio of the voltage across C_1 and C_2, while the total voltage is clamped by C_s.

Figure below shows the output frequency response curve for the ratio detector. It can be seen that at resonance, V_{out} is not equal to 0 V but, rather, to one-half of the voltage across the secondary windings of T_1.

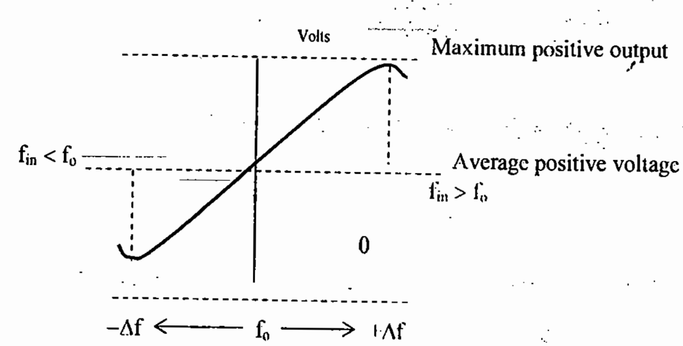


Fig.(b)

Because a ratio detector is relatively immune to amplitude variations, it is often selected over a discriminator. However, a discriminator produces a more linear output voltage-versus-frequency response curve.

Phase Modulation :

$$s(t) = A_c \cos[2\pi f_c t + K_p m(t)] \qquad \text{for PM}$$

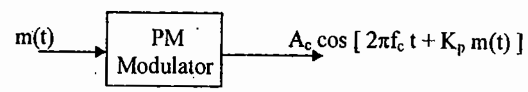
$$s(t) = A_c \cos[2\pi f_c t + 2\pi K_f \int m(t)\,dt] \qquad \text{for FM}$$

For an angle modulated wave ,

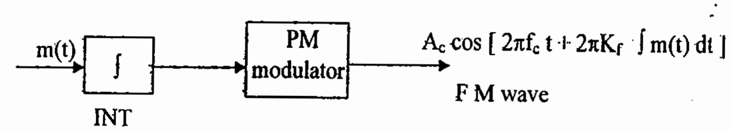
$$s(t) = A_c \cos[2\pi f_c t + \phi(t)]$$

$$\phi(t) = K_p m(t) \qquad \text{for PM}$$

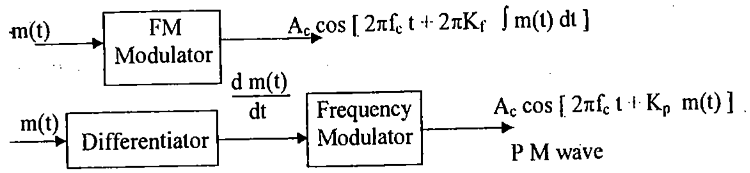
$$= 2\pi K_f \int m(t)\,dt \qquad \text{for FM}$$



FM can be generated using Phase Modulator by prior integration of m(t)



PM can be generated using FM signal.



For sinusoidal signal and exponential , there is no difference between FM ~~and PM~~ except for a phase shift of 90^0. d / dt or Integration of sinusoidal signal is again a sinusoidal signal but with a phase shift of 90^0.

As frequency is continuously changing in PM, the phase is also changing.

For single-tone modulation ,

$$\phi(t) = K_p\, m(t)$$

$$= K_p A_m \cos 2\pi f_m t$$

$$\Delta\phi = K_p A_m = \beta_{PM}$$

$$s(t) = A_c \cos [2\pi f_c t + \beta_{PM} \cos 2\pi f_m t]$$

The bandwidth and power of PM are same as that of FM.

$$BW = 2(\beta_{PM} + 1) f_m$$

$$P_{PM} = \frac{A_c^2}{2}$$

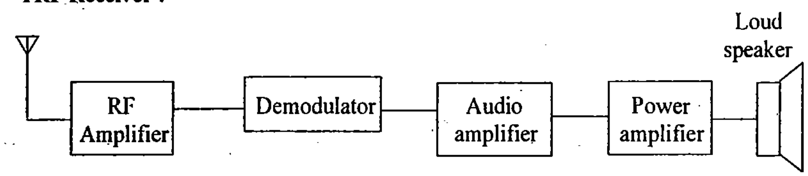
'β' of PM is independent of message frequency.

(C) Receivers:

AM Receivers :

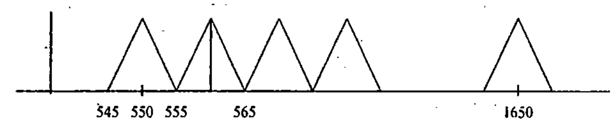
(1) Tuned radio frequency (TRF) Receiver
(2) Superheterodyne Receiver

TRF Receiver :

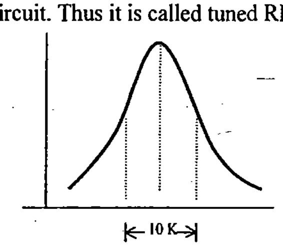


$f_c \longrightarrow$ 550 to 1650 KHz

Bandwidth allotted for each channel in MW is 10 KHz. Message frequency should be limited to 5 KHz.



Ionospheric propagation is used for medium wave. No two stations can have same carrier frequency. The received signal strength is of the order of mW or μW.
RF amplifier must be low noise amplifier. RF amplifier itself acts as a BPF. RF amplifier itself consists of a tuned circuit. Thus it is called tuned RF amplifier.



$$B.W = \frac{f_r}{Q}$$

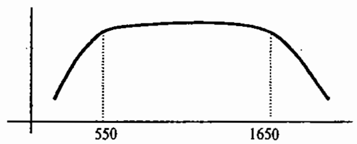
By tuning arrangement we are making the resonant frequency of the tuned circuit equal to the carrier frequency of the required channel. The bandwidth of the tuned circuit should be 10 KHz.

Characteristic parameters of Receiver :

Sensitivity is defined as the minimum signal that should be applied at the input of a receiver to get a standard output. Sensitivity depends on the gain of the amplifier.
If the gain of the RF amplifier is high, the sensitivity is also high.

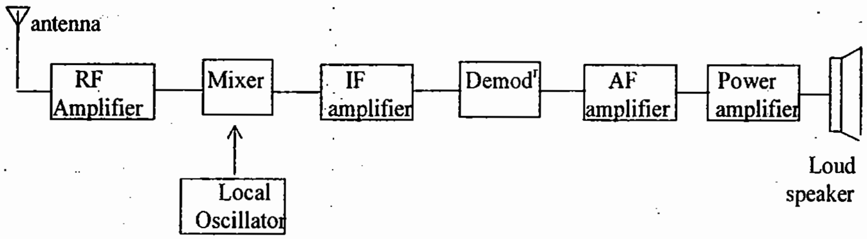
Selectivity is defined as the ability of the receiver to reject unwanted components (or) Sensitivity is defined as the ability of the receiver to select wanted component. The receiver is having very poor selectivity.

Fidelity is defined as the ability of the receiver to reproduce all frequencies equally in the entire tuning range.



The disadvantage of TRF receiver is poor selectivity.

Superheterodyne Receiver :



In the superheterodyne receiver the signal voltage is combined with the local oscillator voltage and converted into a signal of lower fixed frequency. The signal at this intermediate frequency contains the same modulation as the original carrier and is now amplified and detected to reproduce the original information. A constant frequency difference is maintained between the local oscillator and the RF circuits.

In mixer, down conversion is done with respect to the tuned circuit. Tuning means changing the local oscillator frequency. Mixer will change the carrier frequency from f_s to f_{IF}. Intermediate frequency for MW is 455 KHz.

Image frequency :

$$f_{si} = f_s + 2\,IF$$

Image section can be suppressed using a tuned circuit. Image frequency should be removed before the mixer stage.

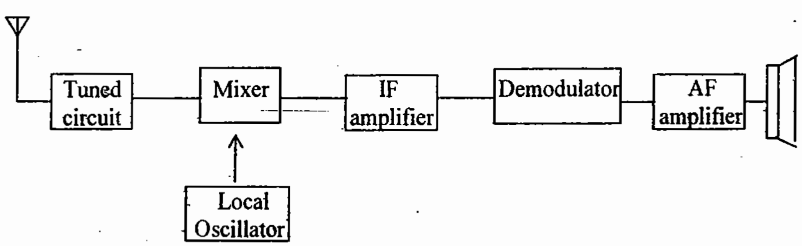
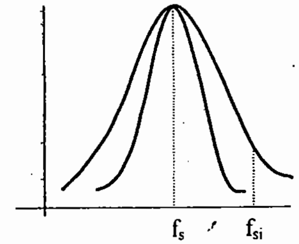


Image (Frequency) Rejection Ratio :



$$IRR = \frac{\text{Gain at } f_s}{\text{Gain at } f_{si}}$$

$$\text{Gain at } f_{si} << 1$$

By increasing the Intermediate frequency, IRR can be increased. By increasing the bandwidth, the gain at f_{si} can be decreased so that IRR increases.

$$IRR \propto \frac{1}{B.W}$$

A minimum of 10 KHz bandwidth should be maintained at the first tuned circuit that is placed before mixer.

$$IRR \propto Q$$

$$IRR = \sqrt{1 + Q^2\rho^2}$$

$$\text{where } \rho = \frac{f_{si}}{f_s} - \frac{f_s}{f_{si}}$$

IRR should be as high as possible. If two tuned circuits are cascaded then the overall IRR =

$$= \sqrt{1 + Q_1^2\rho^2}\,.\,\sqrt{1 + Q_2^2\rho^2}$$

RF amplifier :

(1) IRR increases
(2) The overall gain of the receiver also increases.
(3) The sensitivity of the receiver increases.
(4) SNR at the output of the receiver increases.

Mixer: The purpose of the mixer is to down convert the incoming signal. The output of the mixer is always equal to the difference of the input frequencies. If f_s is the incoming signal frequency and f_l is the local oscillator frequency the output of the mixer is equal to $f_l - f_s$. *Always the local oscillator frequency should be greater than the signal frequency.* If the receiver is tuned to f_s, the local oscillator frequency should be adjusted so that the output of the mixer is equal to IF.

IF amplifier :
It is a tuned amplifier. For a medium wave receiver, Q = 45.5, then we will get a bandwidth of 10 KHz. The resonant frequency of IF tuned amplifier is constant i.e., IF.

$$f_l - f_s = IF$$

Choice of IF:

01. If the IF is too high poor selectivity and poor adjacent channel rejection.
02. A high value of IF increases tracking difficulties.
03. If IF is very low image frequency rejection becomes poorer.
04. If the IF is very low, the frequency stability of the local oscillator should be very high.
05. The IF must not fall with in the tuning range of the receiver.

Tracking of a Receiver:

It is not possible to make a receiver track perfectly over an entire wide range of frequencies. The perfect situation occurs when the RF amplifier and mixer tuned circuits are exactly together and the LO is above these two by an amount exactly equal to the IF frequency. The following steps are employed for tracking.

01. A small variable capacitance in parallel with each section of the ganged capacitor, called the *trimmer*, is adjusted for proper operation at the highest frequency. The highest frequency requires the main capacitor to be at its minimum value. The trimmers are then adjusted to balance out the remaining stray capacitances to provide perfect tracking at the highest frequency.

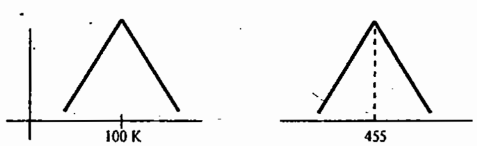
02. At the low frequency, when the ganged capacitors are fully meshed, a small variable capacitor known as the padder capacitor is put in series with the tank inductor. The padders are adjusted to provide tracking at the low frequency in the band.

03. The final adjustment is made at midfrequency by slight adjustment of the inductance in each tank.
Adjust local oscillator frequency so that the output of the mixer is always IF.

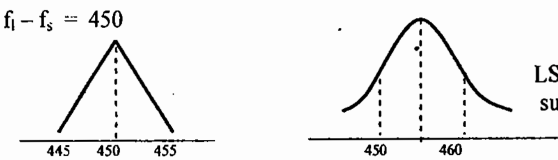
$$f_l - f_s = IF$$

When $f_l - f_s \neq IF$, then it is called tracking error. For a medium wave, it will be within ± 2 KHz.

Case i) $f_l - f_s = IF$



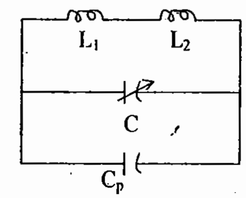
Case ii) $f_l - f_s = 450$



LSB will be suppressed

Tracking error can be minimized using Padder capacitor & Trimmer cirucuit.

$$f_l = \frac{1}{2\pi\sqrt{(L_1+L_2)\,C}}$$



Changing C is nothing but tuning of the receiver. Adjusting Padder capacitor is called fine tuning of a receiver.

$$f_l = \frac{1}{2\pi\sqrt{(L_1+L_2)(C+C_p)}}$$

Fine tuning is always done to reduce the tracking error. When the tracking error is high, trimmer capacitor is placed in series with C.

$$C_{eq} = \frac{C \cdot C_T}{C + C_T} + C_P$$

Example : A superheterodyne receiver is tuned to $f_s = 555$ kHz. Its local oscillator frequency is 1010 kHz. Calculate the IRR when the antenna of this receiver is connected to a mixer through a tuned circuit whose quality factor is 50.

Sol : IF = 455 kHz
f_{si} = 1465 kHz

$$IRR = \sqrt{1 + Q^2 \rho^2}$$

Where $$\rho = \frac{1465}{555} - \frac{555}{1465} = 2.2608$$

$$\therefore \quad IRR = 113.04$$

Automatic Gain Control (AGC) :

The purpose of AGC circuits in a receiver is to maintain a constant output irrespective of variations in the input signal strength. If the input signal strength increases the gain of the receiver is reduced or if the input signal strength decreases the gain of the receiver is increased so that the output level is constant.

The following list gives some of the problems that would be encountered in a receiver without this provision.

01. Tuning the receiver would be nightmare. So as to not miss the weak stations, we should have the volume control (in the non-AGC set).

02. The received signal from any given station is constantly changing as a result of changing weather and ionospheric conditions. The AGC allows to listen to a station without constantly monitoring the volume control.

03. Many radio receivers are utilized under mobile conditions. For instance, a standard broadcast AM car radio would be virtually unusable without a good AGC to compensate for the signal variation in different locations.

The signal from the antenna is not always constant due to fading.

Fading : Variation in signal strength at the input of the receiver due to atmospheric conditions.

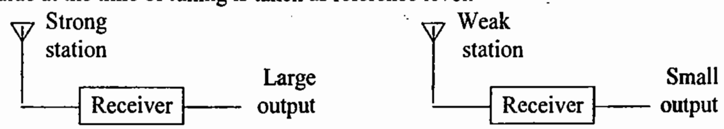
If the receiver gain is constant, then we will hear a loud voice whenever the signal strength is large and we will hear a feasible sound whenever the signal strength is small.

In AGC, we are changing the gain of the receiver according to the input signal strength, so that the output of the receiver is always constant. In AGC, the output of the receiver is constant irrespective of signal strength.

Types of AGC: (1) Simple AGC
(2) Delayed AGC

Simple AGC :

The initial value at the time of tuning is taken as reference level.



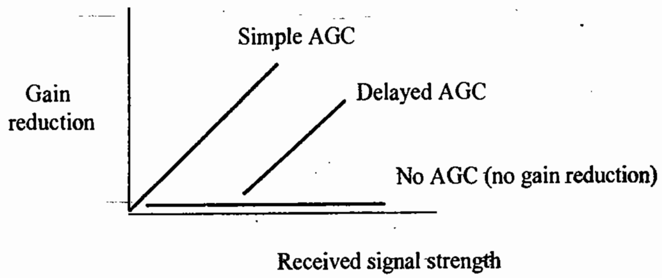
Disadvantage: It provides some gain reduction even to very weak signals.

Delayed AGC :

Delayed AGC does not provide any gain reduction until some arbitrary signal level is attained and therefore has no gain reduction for weak signals.

AGC section is delayed until the input reaches the reference level. After the reference level AGC circuits work and the gain decreases so as to get a constant output.

The characteristics of AGC are shown below.



Squelch circuit :

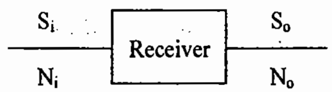
The purpose is to switch on the receiver when the signal is present and to switch off the receiver when the signal is absent.

Double Spotting :

1600 KHz station is selected when the local oscillator frequency is 1100 KHz as well as 2100 KHz for an IF of 500 KHz. So each station is selected twice on the frequency scale. This is called double spotting.

(D) Noise in Analog Modulation

Noise Analysis of AM:



The output signal to noise ratio must be as high as possible. Assume that the channel as AWGN (Additive White Gaussian Noise) channel.

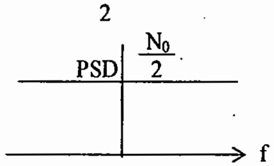
Thermal noise is white noise. It effects all frequencies equally.

$P = K T_e B$; KT = PSD of thermal noise

T_e = Noise equivalent temperature of the receiver

$T_e = (F-1) T_0$; F = Noise figure

$$KT_e = \frac{N_0}{2} = \text{PSD of a white noise} = \frac{\eta}{2}$$



Total power is uniformly distributed to all frequencies.

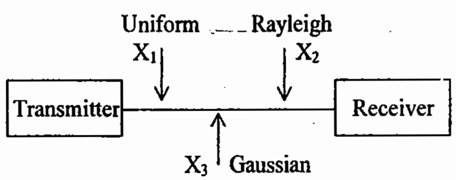
We assume that the noise at the input of the receiver follows Gaussian pdf, this can be justified by central limit theorem.

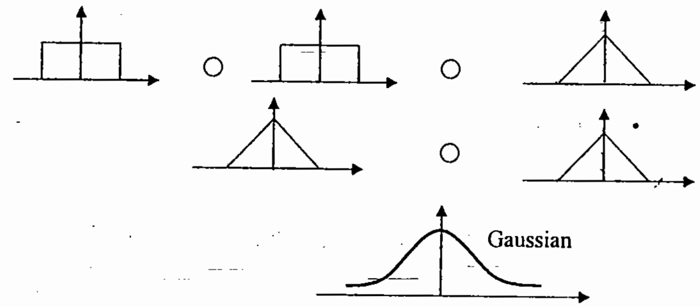
Central Limit Theorem :

Under certain conditions, the sum of a large number of independent random variables follows a Gaussian function irrespective of individual distributions.

$Z = X + Y$

$F_z(z) = f_x(x) * f_y(y)$





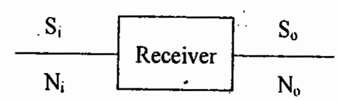
Noise Power = Mean Square Value

ACF at origin = $R_X(0) = \int_{-\infty}^{\infty} S_X(f)\,df$ = Area under PSD graph

Noise Power = Mean square Value = $\int x^2 f_X(x)\,dx$

$= R_X(0) = \int_{-\infty}^{\infty} S_X(f)\,df$ = Area under PSD graph

Figure of Merit :



$$(S/N)_i = (S_i/N_i) = \frac{\text{Power of the modulated signal}}{\text{Power of noise in message bandwidth}}$$

$$(S/N)_o = (S_o/N_o) = \frac{\text{Power of the demodulated signal}}{\text{Power of noise in message bandwidth}}$$

$$\text{Figure of Merit} = \frac{(S/N)_o}{(S/N)_i} = \frac{1}{\text{Noise Figure}}$$

$$\text{Noise Figure} = \frac{(S/N)_i}{(S/N)_o}$$

$(S/N)_o$ depends mainly on modulation scheme and receiver characteristics.

Figure of Merit of a DSB system :

$$S_i = \left(\frac{A_c\, m(t)}{2}\right)^2 = \frac{A_c^2\, m^2(t)}{4} = \frac{A_c^2}{2} P$$ where $m^2(t)$ = power in the message signal = P

For a single tone modulation

$$P = \frac{A_m^2}{2} \quad ; \quad S_i = \frac{A_c^2 A_m^2}{4}$$

$$N_i = W N_0$$

$$(S/N)_i = \frac{A_c^2 P}{2WN_0}$$

Input to the PM is DSB signal + Narrow band noise

$$= A_c\, m(t) \cos 2\pi f_c t + n_i(t) \cos 2\pi f_c t - n_q(t) \sin 2\pi f_c t$$

Output of the PM

$$= A_c\, m(t) \cos^2 2\pi f_c t + n_i(t) \cos^2 2\pi f_c t - n_q(t) \cos 2\pi f_c t \sin 2\pi f_c t$$

Output of the LPF

$$\frac{A_c\, m(t)}{2} + \frac{n_i(t)}{2}$$

$$S_0 = \left(\frac{A_c\, m(t)}{2}\right)^2 = \frac{A_c^2\, m^2(t)}{4} = \frac{A_c^2 P}{4}$$

N_0 = (1/4) [In phase component noise power]

$= (1/4)\ [\, 2WN_0 \,] = (N_0 W)/2$

$$(S/N)_0 = \frac{A_c^2 P}{2WN_0}$$

$$\therefore \text{ Figure of merit } = \frac{A_c^2 P / 2WN_0}{A_c^2 P / 2WN_0} = 1$$

Figure of Merit of a SSB system :

$$S_i = \frac{A_c^2 P}{4} \quad ; \quad N_i = \frac{2WN_0}{2} = W N_0$$

$$(S/N)_i = \frac{A_c^2 P}{4WN_0}$$

Input to the PM = SSB + Narrow band Noise

$$= [A_c\, m(t) \cos 2\pi f_c t + A_c\, \hat{m}(t) \sin 2\pi f_c t\,].\ ½ + n_i(t) \cos 2\pi f_c t - n_q(t) \sin 2\pi f_c t$$

$$\text{Output of LPF} = \frac{A_c\, m(t)}{4} + \frac{n_i(t)}{2}$$

$$S_0 = \frac{A_c^2 P}{16}$$

$$N_0 = (1/4)\,[N_0] \times W = \frac{N_0 W}{4}$$

$$(S/N)_0 = \frac{A_c^2 P / 16}{N_0 W / 4} = \frac{A_c^2 P}{4N_0 W}$$

$$\therefore \text{ Figure of merit } = \frac{A_c^2 P / 4WN_0}{A_c^2 P / 4WN_0} = 1$$

Noise in AM :

$$s(t)_{AM} = A_c \cos 2\pi f_c t + A_c K_a\, m(t) \cos 2\pi f_c t$$

$$S_i = \frac{A_c^2}{2} + \frac{A_c^2 K_a^2 P}{2}$$

$$S_i = \frac{A_c^2}{2}(1 + K_a^2 P) \quad ; \quad N_i = WN_0$$

$$(S_i / N_i) = \frac{A_c^2 (1 + K_a^2 P)}{2WN_0}$$

Input to the Envelope Detector = AM signal + Narrow band Noise

$$= A_c \cos 2\pi f_c t + A_c Ka\, m(t) \cos 2\pi f_c t + n_i(t) \cos 2\pi f_c t - n_q(t) \sin 2\pi f_c t$$

$$\text{Output of Envelope detector} = \sqrt{[\,A_c + A_c Ka\, m(t) + n_i(t)\,]^2 + [\,n_q(t)\,]^2}$$

$$\cong A_c + A_c K_a\, m(t) + n_i(t)$$

$$S_0 = A_c^2 K_a^2 m^2(t) = A_c^2 K_a^2 P \; ; \qquad N_0 = 2WN_0$$

$$(S/N)_0 = \frac{A_c^2 K_a^2 P}{2W N_0}$$

$$\text{Figure of Merit} \cong \frac{K_a P}{1+K_a^2 P} \; \frac{K_a^2 A_m^2}{2+K_a^2 A_m^2} = \frac{\mu^2}{2+\mu^2} = \eta \text{ [modulation efficiency]}$$

Noise in FM :

$$s(t) = A_c \cos [\, 2\pi f_c t + 2\pi K_f \int_0^t m(t)\, dt \,]$$

$$(S/N)_i = \frac{A_c^2}{2\,W\,N_0} \; ; \qquad (S/N)_0 = \frac{3\,A_c^2 K_a^2 P}{2\,N_0\,W^3}$$

$$\text{Figure of Merit} = \frac{3\,K_a^2 P}{W^2} = \frac{3}{2}\beta^2 \quad \text{(For single-tone modulation)}$$

Example : Calculate the transmission bandwidth B_T and the required transmitter power S_T of DSB, SSB, and AM systems for transmitting an audio signal which has a bandwidth of 10 kHz with an output SNR of 40 dB. Assume that the channel introduces a 40 dB power loss and channel noise in AWGN with power spectral density $\eta/2 = 10^{-9}$ W/Hz. Assume $\mu^2 S_X = 0.5$ for AM.

Sol : For DSB and SSB:

$$(S/N)_0 = (S_i/\eta B) = 10^4 \, (= 40 \text{ dB})$$

and
$$S_i = \eta B (10^4) = 2(10^{-9})(10^4)(10^4) = 0.2 \text{ W}$$

Since the channel power is 40 dB, the required transmitted power S_T is

$$S_T = 0.2\,(10^4) = 2000 \text{ W} = 2 \text{ KW [in the case of DSB and SSB]}$$

For AM: $(S/N)_0 = (1/3)(S_i/\eta B)$

Thus, the required transmitted power is 3 times that for the DSB or SSB systems, that is,

$$S_T = 6 \text{ KW}$$

Chapter – 3 Objective Questions Set - A

01. The message signal contain three frequencies 5kHz , 10kHz, 20kHz respectively. The bandwidth of the AM signal is
    a) 40 kHz b) 10kHz
    c) 20 kHz d) 30 kHz

02. If the carrier of a 100 percent modulated AM wave is suppressed, the percentage power saving will be
    a) 50 b) 150
    c) 100 d) 66.66

03. The modulation index of an AM wave is changed from 0 to 1. The transmitted power is
    a) unchanged
    b) halved
    c) doubled
    d) increased by 50 percent

04. A carrier is simultaneously modulated by two sine waves with modulation indices of 0.3 and 0.4; the total modulation index
    a) is 1
    b) can not be calculated unless the phase relations are known
    c) is 0.5
    d) is 0.7

05. Amplitude modulation is used for broadcasting because
    a) It is more noise immune than other modulation systems
    b) compared with other systems it requires less transmitting power
    c) its use avoids receiver complexity
    d) no other modulation system can provide the necessary bandwidth for high fidelity

06. The positive RF peaks of an AM voltage rise to a maximum value of 12 V and drop to a minimum value of 4V. The modulation index assuming single tone modulation is
    a) 3 b) 1/3 c) ¼ d) ½

07. The most suitable method for detecting a modulated signal $(2.5 + 5 \cos \omega_m t) \cos \omega_c t$ is
    a) envelope detector
    b) synchronous detector
    c) ratio detector
    d) both a and b

08. The main advantage of superheterodyne receiver is,
    a) simple circuit
    b) better tracking
    c) improvement in selectivity and sensitivity
    d) better alignment

09. The received signal frequency at any time of a superheterodyne receiver having IF = 456 KHz, is 1 MHz. The corresponding signal is
    a) within its medium band
    b) outside the medium band
    c) depends on modulation index
    d) depends on modulating frequency

10. The resonant frequency of an RF amplifier is 1 MHz and its bandwidth is 10 KHz. The Q-factor will be
    a) 10 b) 100
    c) 0.01 d) 0.1

11. A plot of modulation index versus carrier amplitude yields a
    a) horizontal line b) vertical line
    c) parabola d) hyperbola

12. A carrier is amplitude modulated to a depth of 40%. The increase in power is
a) 40% b) 20%
c) 16% d) 8%

13. Following is not the purpose of modulation
a) multiplexing
b) effective radiation
c) narrowbanding
d) increase in signal power

14. An AM wave is given by $e_{AM} = 10 (1 + 0.4 \cos 10^3 t + 0.3 \cos 10^4 t) \cos 10^6 t$. The modulation index is
a) 0.4 b) 0.5 c) 0.3 d) 0.9

15. The intermediate frequency of a supreheterodyne receiver is 450 KHz. If the image frequency of a station is 2100 KHz, its actual frequency is
a) 750 KHz b) 900 KHz
c) 1200 KHz d) 1010 KHz

16. In AM wave has modulated index of 100%. If the carrier is suppressed, the percentage power saving will be
a) 50 % b) 66.6 %
c) 75 % d) 33.3 %

17. The modulation index of over-modulated wave is
a) < 1 b) > 1
c) infinity d) 1

18. If the modulation index of an AM wave is changed from 0 to 1, the transmitted power
a) increases by 50 %
b) increases by 75 %
c) increases by 100 %
d) None

19. In an AM wave, the total power content is 600 W and that of each sideband is 75 W. The modulation index is
a) 53.5 % b) 60.7 %
c) 81.6 % d) 40.3 %

20. In the above question, the carrier power is
a) 325 W b) 450 W
c) 525 W d) 425 W

21. The AM broadcast band is given by
a) 10 KHz to 30 KHz
b) 500 KHz to 1500 KHz
c) 3 MHz to 30 MHz
d) None

22. The number of AM broadcast stations that can be accommodated in a 100 KHz bandwidth for the highest modulating frequency of 5 KHz will be
a) 5 b) 10 c) 20 d) 50

23. In AM, if the modulation index is more than 100 %, then
a) Power of the wave increases
b) Efficiency of transmission increases
c) The wave gets distorted
d) None

24. In AM transmission, the frequency which is not transmitted is
a) Upper side frequency
b) Lower side frequency
c) Audio frequency
d) None

25. An audio signal $15 \sin 2\pi (1500 t)$ amplitude modulates $60 \sin 2\pi (10^5 t)$. The modulation index will be
a) 20% b) 25 %
c) 50 % d) 75 %

26. In the above question, the total bandwidth required to transmit the AM wave is
a) 1.5 KHz b) 3 K
c) 100 KHz d) 10 KHz

27. An AM wave is given by $e_{AM} = 10 (1 + 0.4 \cos 10^3 t + 0.3 \cos 10^4 t) \cos 10^6 t$. The message bandwidth is
a) 1 KHz b) 2 KHz
c) 5 KHz d) 10 KHz

28. An AM wave is given by $e_{AM} = 10 (1 + 0.4 \cos 10^3 t + 0.3 \cos 10^4 t) \cos 10^6 t$. The bandwidth of the AM signal is
a) 5 KHz b) 10 KHz
c) 20 KHz d) 30 KHz

29. An AM wave is given by $e_{AM} = 10 (1 + 0.4 \cos 10^3 t + 0.3 \cos 10^4 t) \cos 10^6 t$. The following frequency will not be present in the spectrum
a) 1 MHz b) 2 MHz
c) 1010 KHz d) 990 KHz

30. An AM wave is given by $e_{AM} = 10 (1 + 0.4 \cos 10^3 t + 0.3 \cos 10^4 t) \cos 10^6 t$. The highest frequency of the signal after modulation is
a) 1 MHz b) 2 MHz
c) 1010 KHz d) 990 KHz

Key :

01. a	02. d	03. d	04. c	05. c	06. d
07. c	08. c	09. a	10. b	11. d	12. d
13. d	14. b	15. c	16. b	17. b	18. a
19. c	20. b	21. b	22. b	23. c	24.c
25. b	26. b	27. d	28. c	29. b	30. c

Chapter – 3 **Objective Questions Set - B**

01. Indicate the *false* statement regarding the advantages of SSB over double sideband, full-carrier AM.
    a) More channel space is available.
    b) Transmitter circuits must be more stable, giving better reception.
    c) The signal is more noise-resistant.
    d) Much less power is required for the same signal strength.

02. When the modulation index of an AM wave is doubled, the antenna current is also doubled. The AM system being used is
    a) Single-sideband, full carrier (H3E)
    b) Vestigial sideband (C3F)
    c) Single-sideband, suppressed carrier (J3E)
    d) Double-sideband, full carrier (A3E)

03. Indicate which one of the following advantages of the phase cancellation method of obtaining SSB over the filter method is *false* :
    a) Switching from one sideband to the other is simpler.
    b) It is possible to generate SSB at any frequency.
    c) SSB with lower audio frequencies present can be generated.
    d) There are more balanced modulators; therefore the carrier is suppressed better.

04. The most commonly used filters in SSB generation are
    a) mechanical b) LC
    c) RC d) low-pass

05. In an SSB transmitter, one is most likely to find a
    a) class C audio amplifier
    b) tuned modulator
    c) class B RF amplifier
    d) class A RF amplifier

06. Indicate in which one of the following only sideband is transmitted.
    a) H3E b) A3E
    c) B8E d) C3F

07. One of the following *cannot* be used to remove the unwanted sideband in SSB. This is the
    a) filter system
    b) phase-shift method
    c) third method
    d) balanced modulator

08. R3E modulation is sometimes used to
    a) allow the receiver to have a frequency synthesizer
    b) simplify the frequency stability problem in reception
    c) reduce the power that must be transmitted
    d) reduce the bandwidth required for transmission

09. To provide two or more voice circuits with the same carrier, it is necessary to use
    a) ISB
    b) carrier reinsertion
    c) SSB with pilot carrier
    d) Lincompex

10. Vestigial sideband modulation (C3F) is normally used for
    a) HF point-to-point communications
    b) monaural broadcasting
    c) TV broadcasting
    d) stereo broadcasting

11. Which of the following AM techniques provide the advantages of greater signal power and reduction of bandwidth ?
    a) DSB-SC b) SSB
    c) USB d) None

12. Completely suppressed carrier transmission is
    a) cheaper in cost
    b) simpler in design
    c) never produced commercially
    d) None

13. Which of the following techniques is acceptable for voice communication ?
    a) DSB-SC b) SSB
    c) Both d) None

14. The DSB-SC signal is detected by
    a) removing the carrier
    b) removing one sideband
    c) reinjecting the carrier
    d) None

15. The VSB signal is produced from the DSB signal by employing
    a) Simpler filters
    b) Balanced modulator
    c) Ring modulator
    d) None

16. The SSB can be obtained from balanced modulator by connecting at its output a
    a) Buffer b) Clipper
    c) Filter d) None

17. In phase-shift SSB modulator, the input signals to one of the balanced modulators are phase-shifted by
    a) 45^0 b) 90^0 c) 180^0 d) 30^0

18. The SSB modulator is known as
    a) Balanced modulator
    b) Product modulator
    c) Amplitude modulator
    d) None

19. The advantage of SSB-SC system is that it provides
    a) Higher frequency of transmission
    b) Better quality of communication
    c) Simpler and inexpensive circuitry
    d) None

20. The filter required to obtain SSB from DSB signal is
    a) LPF b) HPF
    c) BPF d) None

Key :

01. b 02. c 03. d 04. a 05. c 06. a
07. d 08. b 09. a 10. c 11. b 12. c
13. b 14. c 15. c 16. c 17. b 18. b
19. a 20. c

Chapter – 3 **Objective Questions Set - C**

01. In the stabilized reactance modulator AFC system,
 a) the discriminator must have a fast time constant to prevent demodulation
 b) the higher the discriminator frequency, the better the oscillator frequency stability
 c) the discriminator frequency must not be too low, or the system will fail
 d) phase modulation is converted into FM by the equalizer circuit

02. In the spectrum of a frequency-modulated wave
 a) the carrier frequency dissipates when the modulation index is large
 b) the amplitude of any sideband depends on the modulation index
 c) the total number of sidebands depends on the modulation index
 d) the carrier frequency cannot disappear

03. The difference between the phase and frequency modulation
 a) is purely theoretical because they are the same in practice
 b) is too great to make the two systems compatible
 c) lies in the poorer audio response of phase modulation
 d) lies in the different definitions of the modulation index

04. Indicate the *false* statement regarding the Armstrong modulation system
 a) The system is basically phase, not frequency, modulation.
 b) AFC is not needed, as a crystal oscillator is used.
 c) Frequency multiplication must be used.

05. An FM signal with a modulation index m_f is passed through a frequency tripler. The wave in the output of the tripler will have a modulation index of
 a) $m_f / 3$ b) m_f
 c) $3\ m_f$ d) $9\ m_f$

06. An FM signal with a deviation δ is passed through a mixer, and has its frequency reduced fivefold. The deviation in the output of the mixer is
 a) $5\ \delta$ b) indeterminate
 c) $\delta/5$ d) δ

07. A pre-emphasis circuit provides extra noise immunity by
 a) boosting the base frequencies
 b) amplifying the higher audio frequencies
 c) preamplifying the whole audio band
 d) converting the phase modulation to FM

08. Since noise phase-modulates the FM wave, as the noise sideband frequency approaches the carrier frequency, the noise amplitude
 a) remains constant b) is decreased
 c) is increased d) is equalized

09. When the modulating frequency is doubled, the modulation index is halved, and the modulating voltage remains constant. The modulation system is
 a) amplitude modulation
 b) phase modulation
 c) frequency modulation
 d) any of the three

10 Indicate which one of the following is not an advantage of FM over AM
 a) Better noise immunity is provided
 b) Lower bandwidth is required

11. One of the following is an indirect way of generating FM. This is the
 a) reactance FET modulator
 b) varactor diode modulator
 c) Armstrong modulator
 d) reactance bipolar transistor modulator

12. In an FM stereo multiplex transmission, the
 a) sum signal modulates the 19 kHz subcarrier
 b) difference signal modulates the 19 kHz subcarrier
 c) difference signal modulates the 38 kHz subcarrier
 d) difference signal modulates the 67 kHz subcarrier

13. Armstrong F.M transmitter performs frequency multiplication in stages
 a) to increase the overall S/N ratio
 b) to reduce bandwidth
 c) to find the desired value of carrier frequency as well as frequency deviation
 d) for convenience

14. Limiter is not essential in the following detector
 a) Foster - Seeley
 b) balanced slope
 c) ratio
 d) None

15. Figure of merit is always unity in
 a) SSB - SC b) AM
 c) FM d) All the three

16. The output V_R of the ratio detector is related with the output V_F of similar Foster - Seeley discriminator as follows :
 a) $V_F = V_R$
 b) $V_F > V_R$
 c) $V_F = 0.51\ V_R$
 d) $V_F = 2\ V_R$

17. Which one is an advantages of AM over FM
 a) FM is more immune to noise
 b) FM has better fidelity
 c) Probability of noise spike generation is less in AM
 d) FM has wide bandwidth

18. The message carrying efficiency is best in
 a) FM b) AM
 c) AM - SC d) Phase modulation

19. Following is not advantage of FM over AM.
 a) noise immunity b) fidelity
 c) capture effect d) sputtering effect

20. The modulating frequency in frequency modulation is increased from 10 KHz to 20 KHz. The bandwidth is
 a) doubled b) halved
 c) increases by 20 KHz
 d) increases tremendously

21. A narrowband FM does not have the following feature:
 a) it has two sidebands
 b) both sidebands are equal in amplitude
 c) both sidebands have same phase difference with respect to carrier
 d) it does not show amplitude variations

22. In time division multiplexing, the FM detector has
 a) more b) less
 c) equal
 d) unknown
 noise contribution

23. In a single-tone FM discriminator (S_o / N_o) is
 a) proportional to deviation
 b) proportional to cube of deviation

24. Assuming other parameters unchanged, if the modulating frequency is halved in a modulation system, the modulation index is doubled. The modulation system is

a) AM b) FM

c) Phase modulation

d) Angle modulation

25. Which of the following processes is a linear modulation process ?

a) NBFM b) WBFM

c) NBPM d) None

26. An increase in the modulation index leads to increase in bandwidth in case of

a) AM b) FM

c) PM d) None

27. FM broadcast band lies in

a) VHF band b) UHF band

c) SHF band d) None

28. The threshold in FM depends on

a) Frequency deviation

b) Power level of frequencies radiated at the receiver

c) Both (a) & (b)

d) None

29. In NBFM, the maximum modulation frequency is 3 KHz and maximum deviation is 5 KHz. The modulation index is

a) less than 1 b) equal to 1

c) Greater than 1 d) None

30. The disadvantage of FM over AM is that

a) Noise is very high for high frequency signals

b) Larger bandwidth is required

c) High modulating power is required

d) None

Key :

01. c	02. b	03. d	04. d	05. c	06. d
07. b	08. b	09. c	10. b	11. c	12. c
13. c	14. c	15. a	16. c	17. c	18. c
19. d	20. c	21. c	22. b	23. d	24. b
25. c	26. b	27. a	28. c	29. a	30. b

Chapter – 3 Additional Objective Questions Set - D

01. A sinusoidal voltage of amplitude 1 kV is amplitude modulated by another sinusoidal voltage to produce 30 % modulation. The amplitude of each sideband term is
a) 300 volts b) 150 volts c) 500 volts d) 250 volts

02. A sinusoidal voltage, amplitude modulates another sinusoidal voltage of amplitude 1 kV to result in two sideband terms of amplitude 200 volts each. The modulation index is
a) 0.1 b) 0.2 c) 0.4 d) 0.5

03. A carrier voltage of unmodulated carrier power of 1 kW, on being amplitude modulated by an audio sinusoidal voltage to a depth of 100 %, has total modulated carrier power of
a) 1.25 kW b) 1.5 kW c) 2 kW d) 4 kW

04. In AM broadcast, the maximum modulation frequency is restricted to
a) 3 kHz b) 5 kHz c) 10 kHz d) 15 kHz

05. A sinusoidal carrier voltage of amplitude 100 volts is amplitude modulated by a sinusoidal voltage of frequency 1 kHz resulting in maximum modulated carrier amplitude of 130 volts. The modulation index is
a) 0.03 b) 0.3 c) 1.3 d) 0.13

06. An amplitude modulated voltage in volts is given by $v = 20(1 + 0.5 \sin 6280t) \sin(6.28 \times 10^6)t$. The rms value of the unmodulated carrier voltage in volts is
a) 20 b) $20/\sqrt{2}$ c) 10 d) $10/\sqrt{2}$

07. An amplitude modulated voltage in volts is given by $v = 20(1 + 0.5 \sin 6280t) \sin(6.28 \times 10^6)t$. The rms value of sideband voltage in volts is
a) $5/\sqrt{2}$ b) 5 c) 0.5 d) $10/\sqrt{2}$

08. An amplitude modulated voltage in volts is given by $v = 20(1 + 0.5 \sin 6280t) \sin(6.28 \times 10^6)t$. The percentage modulation index of the modulated voltage is
a) 25 % b) 50 % c) $50/\sqrt{2}$ % d) 100 %

09. An amplitude modulated voltage in volts is given by $v = 20(1 + 0.5 \sin 6280t) \sin(6.28 \times 10^6)t$. The modulated frequency is
a) 500 Hz b) 1000 Hz c) 6280 Hz d) 2000 Hz

10. An amplitude modulated voltage in volts is given by $v = 20(1 + 0.5 \sin 6280t) \sin(6.28 \times 10^6)t$. The carrier frequency is
a) 6.28×10^6 Hz b) 10^6 Hz c) 2×10^6 Hz d) 0.5×10^6 Hz

11. In amplitude modulation system, if modulation index is raised from 1 to 1.2, then
a) Power of the wave increases b) Efficiency of transmission increases
c) Bandwidth increases d) The signal gets distorted

12. An amplitude modulated voltage has modulation index of 100 %. If the carrier is suppressed, the percentage power saving is
a) 50 % b) 66.6 % c) 75 % d) 25 %

13. In amplitude modulation , the modulation envelope has a peak value double the unmodulated carrier value. The modulation index is
a) 25 % b) 50 % c) 75 % d) 100 %

14. A sinusoidal carrier voltage of frequency 1200 kHz is amplitude modulated by a sinusoidal voltage of frequency 20 kHz resulting in maximum and minimum modulated carrier amplitudes of 110 volts and 90 volts respectively. The unmodualted carrier amplitude is
a) 110 volts b) 90 volts c) 100 volts d) 50 volts

15. A sinusoidal carrier voltage of frequency 1200 kHz is amplitude modulated by a sinusoidal voltage of frequency 20 kHz resulting in maximum and minimum modulated carrier amplitudes of 110 volts and 90 volts respectively. The modulation index is
a) 0.1 b) 0.2 c) 0.4 d) 0.05

16. A sinusoidal carrier voltage of frequency 1200 kHz is amplitude modulated by a sinusoidal voltage frequency 20 kHz resulting in maximum and minimum modulated carrier amplitudes of 110 volts and 90 volts respectively. The amplitude of each sideband is
a) 10 volt b) 5 volts c) 20 volts d) 40 volts

17. A carrier voltage of rms value 100 volts is amplitude modulated by a sinusoidal audio voltage to cause modulation index of 0.2. The rms value of carrier on modulation is
a) 101 volts b) 102 volts c) 104 volts d) 120 volts

18. One of the advantages of base modulation over collector modulation of a class C amplifier is
a) better frequency b) lower modulation power requirement
c) better linearity of modulation d) higher power output per transistor

19. In AM transmission, the frequency which is not transmitted is
a) upper sideband b) lower side band
c) carrier frequency d) audio frequency

20. The antenna current of an AM transmitter under unmdoulated condition is 10 amp. The current increases to 10.4 amp on amplitude modulation of the carrier . The modulation index is
a) 0.2 b) 0.4 c) 0.04 d) 0.8

21. A carrier is simultaneously amplitude modulated by two sine waves causing individual modulation of 30 % and 40 %. The overall modulation index is
a) 50 % b) 35 % c) 70 % d) 40 %

22. In an AM transmitter, the unmodulated output current is I_C. The modulated current I_t equals
a) $I_C\sqrt{1+m^2}$ b) $I_C\sqrt{1+(m^2/2)}$ c) $I_C\sqrt{1+m}$ d) $I_C\ (m/2)$

23. In an amplitude modulated carrier, the total power is 1160 watts while that of each sideband is 80 watts. The unmodulated carrier power is
a) 1000 watts b) 1040 watts c) 1080 watts d) 500 watts

24. In an AM signal, the peak antenna current is 13 Amp and the minimum current is 7 Amp. The percent modulation is
a) 20 % b) 30 % c) 50 % d) 100 %

25. Which system is free from noise ?
a) FM b) AM c) Both FM & AM d) None of the above

26. The draw back of FM relative to AM is that
a) noise is very high for high modulation frequencies
b) larger bandwidth is required
c) higher modulating power is required
d) higher output power is required

27. An AM signal is detected using an envelope detector. The carrier frequency and modulating signal frequency are 1 MHz and 2 KHz respectively. An appropriate value for the time constant of the envelope detector is
a) 500 μ sec b) 20 μ sec c) 0.2 μ sec d) 1 μ sec

28. In FM broadcast, the maximum modulation frequency is restricted to
a) 5 kHz b) 10 kHz c) 15 kHz d) 20 kHz

29. In frequency modulation, if the amplitude of the modulating voltage is doubled, the maximum frequency deviation
a) doubles b) becomes four times
c) becomes half d) remains unaltered

30. In frequency modulation , if the frequency of the modulating voltage is doubled, the rate of deviation of carrier frequency
a) doubles b) becomes four times
c) becomes half d) remains unaltered

31. In frequency modulation , if the frequency of the modulating voltage is doubled, the maximum frequency deviation
a) doubles b) becomes four times
c) becomes half d) remains unaltered

32. A frequency modulated voltage with modulation index 'm_f' is passed through a frequency doubler. The FM signal in the output of the doubler will have modulation index of
a) $2m_f$ b) m_f c) m_f d) $4\ m_f$

33. An FM signal with frequency deviation δ is passed through a mixer and has its frequency reduced three fold. The frequency deviation in the output of the mixer is
a) $\delta/3$ b) δ c) 3δ d) 9δ

34. The bandwidth requirement of a telephone channel is
a) 3 Hz b) 5 kHz c) 10 kHz d) 15 kHz

35. A 4 volt audio modulating signal changes the carrier frequency from 200 kHz to 210 kHz, the frequency deviation is
a) 5 kHz b) 10 kHz c) 15 kHz d) 20 kHz

36. In FM, the carrier frequency deviation is determined by
a) Modulating voltage b) Modulating frequency
c) Both modulating voltage & frequency d) None of the above

37. A 2.5 volt 500 Hz voltage frequency modulates the carrier to cause frequency deviation of 5 kHz. The modulation index is
a) b) 25 c) 5 d) 50

38. A 2.5 volt 500 Hz voltage frequency modulates the carrier to cause frequency deviation of 5 kHz. On increasing the modulating voltage to 10 volts, the frequency deviation becomes
a) 8 kHz b) 16 kHz c) 4 kHz d) 1 kHz

39. In FM, the carrier frequency deviation, on having increased the modulating voltage to 10 volts, the modulating index becomes
a) 16 b) 4 c) 8 d) 32

40. A 2 volt, 1 kHz signal, frequency modulates a carrier voltage to cause frequency deviation of 5 kHz. If the modulating voltage is changed to 20 volts, 200 Hz, the new frequency deviation is
a) 5 kHz b) 50 kHz c) 0.5 d) 100 kHz

41. A 2 volt, 1 kHz signal, frequency modulates a carrier voltage to cause frequency deviation of 5 kHz. If the modulating voltage is changed to 20 volts, 200 Hz, with modulating voltage 20 volts , 200 Hz, with modulating voltage of 20 volts, 20 Hz
a) 5 kHz b) 50 kHz c) 250 d) 100

42. A 1 kW carrier is modulated to a depth of 60 %. The total power in the modulated carrier is
a) 1 kW b) 1.06 kW c) 1.18 kW d) 1.6 kW

43. A frequency modulated carrier is represented by $v = 20\sin(6.28 \times 10^8 t + 4\sin 628 t)$. The carrier frequency is
a) 6.28×10^8 Hz b) 3.14×10^8 Hz c) 108 Hz d) 2×10^8 Hz

44. A frequency modulated carrier is represented by $v = 20\sin(6.28 \times 10^8 t + 4\sin 628 t)$. The modulation frequency is

a) 628 Hz b) 100 Hz c) 200 Hz d) 314 Hz

45. A frequency modulated carrier is represented by $v = 20\sin(6.28 \times 10^8 t + 4\sin 628 t)$ The frequency deviation is
a) 400 Hz b) 628 Hz c) 2512 Hz d) 314 Hz

46. In frequency modulation
a) noise decreases by increasing frequency deviation
b) noise decreases by decreasing frequency deviation
c) noise is unaffected by change of frequency deviation
d) noise decreases by increasing the bandwidth

47. In frequency modulation, for a given frequency deviation, the modulation varies
a) inversely as the modulation frequency
b) directly as the modulating frequency
c) inversely as the square of modulating frequency
d) directly as the square of modulating frequency

48. In FM system, if the depth of modulation is doubled, the output power
a) increases by factor of $\sqrt{2}$ b) increases by factor of $\sqrt{3}$
c) increases by factor of 2 d) remains at unmodulated value

49. In FM, frequency deviation is
(a) proportional to amplitude of modulated signal
(b) proportional to frequency of modulation signal
(c) directly proportional to amplitude and inversely proportional to the frequency of the modulating signal.
(d) None of the above

50. Which of the following statements is not true for FM ?
a) the carrier never becomes zero
b) the J – coefficients occasionally are negative
c) the total power remains constant in spite of change of modulation index
d) the total bandwidth increases with increase in modulation index

51. In FM, the output noise may be decreased by
a) decreasing frequency deviation b) increasing frequency deviation
c) by keeping, deviation constant d) None of the above

52. In a frequency modulated voltage, the maximum modulating frequency is 15 kHz and the maximum frequency deviation is 75 kHz. If the significant sideband pairs extend upto 8^{th}, the theoretical bandwidth required is
a) 30 kHz b) 150 kHz c) 180 kHz d) 480 kHz

53. In the above question the practical bandwidth is
a) 30 kHz b) 150 kHz c) 180 kHz d) 240 kHz

54. In a FM system, modulation index is 7 and the practical bandwidth is 160 kHz. The frequency deviation is
a) 20 kHz b) 35 kHz c) 70 kHz d) 140 kHz

55. In a FM system, the carrier frequency is 200 MHz, maximum modulating frequency is 10 kHz and maximum frequency deviation is 1 MHz. The practical bandwidth requirement is
a) 1 MHz b) 2 MHz c) 2.5 MHz d) 4 MHz

56. In the above question, if the modulating signal amplitude is doubled, the practical bandwidth required will be
a) 1 MHz b) 2 MHz c) 2.5 MHz d) 4 MHz

57. In which of the following modulation systems does the increase of modulation index result in increase in bandwidth ?
a) amplitude modulation b) frequency modulation
c) phase modulation d) both frequency and phase modulations

58. Practical bandwidth of a wideband FM signal ($1 < \delta < 100$) with modulating frequency f_m and maximum frequency deviation f_d approximately equals
a) $2 f_m$ b) f_d c) $2 f_d$ d) $2 (f_d + f_m)$

59. Practical bandwidth of a narrow band FM signal ($\delta < 1$) equals
a) f_m b) $2 f_m$ c) f_d d) $2 f_d$

60. Practical bandwidth of a very wideband FM signal approximately equals
a) f_m b) $2 f_m$ c) f_d d) $2 f_d$

where f_m is the modulating frequency and f_d is frequency deviation.

61. From bandwidth point of view, narrow band FM is equivalent to
a) AM b) Phase modulation c) SSB d) Suppressed carrier - DSB

62. In a TV systems, the modulation methods employed for video and audio signals are
a) both amplitude modulation
b) both frequency modulation
c) respectively amplitude modulation and frequency modulation
d) respectively frequency modulation and amplitude modulation

63. Two carriers 40 MHz and 80 MHz respectively are frequency modulated by a signal of frequency 4 KHz, such that the bandwidths of the FM signal in the two cases are the same. The peak deviations in the two cases are in the ratio of
a) 1 : 4 b) 1 : 2 c) 1 : 1 d) 2 : 1

64. Pre-emphasis in FM systems involves
a) compression of the modulating signal
b) expansion of the modulating signal
c) amplification of lower frequency components of the modulating signal
d) amplification of higher frequency components of the modulating signal

65. As the modulation index of an FM signal with sinusoidal modulation is increased from zero to three, the power in the carrier component will
a) increase continuously b) decrease continuously
c) first increase, attain a maximum and then decrease
d) first decrease, become zero and then increase

66. An FM signal with a deviation δ is passed through a mixer and has its frequency reduced five fold. The deviation in the output of the mixer is
a) 5 δ b) intermediate c) (δ / 5) d) δ

67. Let x(t) = 5 cos (50 t + sin 5t). Its instantaneous frequency (in rad/s) at t = 0 has the value
a) 5 b) 50 c) 55 d) 250

68. An FM wave uses a 2.5 V, 500 Hz modulating frequency and has a modulation index of 50. The deviation is
a) 500 Hz b) 1000 Hz c) 1250 Hz d) 25000 Hz

69. In the spectrum of a FM wave
a) the carrier frequency cannot disappear
b) the carrier frequency disappears when the modulation index is large
c) the amplitude of any sideband depends on the modulation index
d) the total number of sidebands depends on the modulation index

70. In FM reception, amplitude disturbances due to static change
a) reduce the signal frequencies b) increase the signal frequencies
c) disturb the signal frequencies d) do not affect the signal frequencies

71. The FM modulation index is given by

a) $\dfrac{\text{Maximum frequency}}{\text{Minimum frequency}}$ b) $\dfrac{\text{Minimum frequency}}{\text{Maximum frequency}}$

c) $\dfrac{\text{Modulating frequency}}{\text{Maximum frequency deviation}}$ d) $\dfrac{\text{Maximum frequency deviation}}{\text{Modulating frequency}}$

72. Under identical conditions FM & PM are indistinguishable for a single modulating frequency. Now if the modulating frequency is increased,
a) PM modulation index will remain constant, where as FM modulation index will increase
b) PM modulation index will remain constant, where as FM modulation index will decrease
c) PM modulation index will decrease and FM modulation index will increase
d) Both PM as well as FM modulation indices will increase

72. The image channel selectivity of superheterodyne receiver depends upon
a) IF amplifiers only b) RF and IF amplifiers only
c) Preselector, RF and IF amplifiers d) Preselector, and RF amplifiers only

73. The image channel rejection in a superheterodyne receiver comes from
a) IF stages only b) RF stages only
c) detector and RF stages only d) detector RF, and IF stage

74. The image (second) channel selectivity of a superheterodyne communication receiver is determined by
a) antenna and preselector b) the preselector and RF amplifier
c) the preselector and IF amplifier d) The RF and IF amplifier

75. A superheterodyne radio receiver with an intermediate frequency of 455 KHz is tuned to a station operating at 1200 KHz. The associated image frequency is

76. A superheterodyne receiver is to operate in the frequency range 550 kHz 1650 kHz, with the intermediate frequency of 450 kHz.

Let $R = \dfrac{C_{max}}{C_{min}}$ denote the required capacitance ratio of the local oscillator and I denote the image frequency (in kHz) of the incoming signal. If the receiver is tuned to 700 kHz, then

a) R = 4.41, I = 1600 b) R = 2.10, I = 1150
c) R = 3.0, I = 1600 d) R = 9.0, I = 1150

77. An AM superheterodyne receiver with IF of 455 kHz is tuned to the carrier frequency of 1000 kHz. The image frequency is
a) 545 kHz b) 1 MHz c) 1455 kHz d) 1910 kHz

78. A broad cast radio receiver with IF = 455 kHz is tuned to 1500 kHz. The image frequency will be
a) 1045 kHz b) 1500 kHz c) 1955 kHz d) 2410 kHz

79. Match List – I with List – II and select the correct answer by using the codes given below the lists:

List – I	List – II
A. RF amplifier	1. Amplifies received carrier and sidebands
B. Loudspeaker	2. Gives acoustic output
C. Demodulator	3. Has IF input and AF output
D. IF amplifier	4. Fixed tuned to intermediate frequency

	A	B	C	D		A	B	C	D
a)	1	2	3	4	b)	2	1	3	4
c)	3	4	1	2	d)	1	4	3	2

80. An AM superheterodyne receiver with IF of 455 kHz is tuned to the carrier frequency of 1000 kHz. The image frequency is
a) 545 kHz b) 1 MHz c) 1455 kHz d) 1910 kHz

81. The correct sequence of subsystems in an FM receiver is
a) mixer, RF amplifier, limiter, IF amplifier, discriminator, audio amplifier
b) RF amplifier, mixer, IF amplifier, limiter, discriminator, audio amplifier
c) RF amplifier, mixer, limiter, discriminator, IF amplifier, audio amplifier
d) mixer, IF amplifier, limiter, audio amplifier, discriminator

82. A superheterodyne receiver has an IF of 465 kHz. If it is tuned to a station broadcasting at 500 kHz and its oscillator is operating at 965 kHz, then the 1430 kHz frequency would be the

a) adjacent channel frequency b) image frequency c) gyro frequency

Chapter – 3 **Additional Objective Questions** Set - E

01. Consider the amplitude modulated (AM) signal $A_c \cos \omega_c t + 2 \cos \omega_m t \cos \omega_c t$. For demodulating the signal using envelope detector, the minimum value of A_c should be
(a) 2 (b) 1
(c) 0.5 (d) 0

02. Consider the frequency modulated signal $10\cos[2\pi \times 10^5 t + 5\sin(2\pi \times 1500t) + 7.5\sin(2\pi \times 1000t)]$ with carrier frequency of 10^5 Hz. The modulation index is

(a) 12.5 (b) 10 (c) 7.5 (d) 5

03. The signal $\cos\omega_c t + 0.5\cos\omega_m t \sin\omega_c t$ is

(a) FM only
(b) AM only
(c) both AM and FM
(d) neither AM nor FM

04. The diagonal clipping in Amplitude Demodulation (using envelope detector) can be avoided if RC time – constant of the envelope detector satisfies the following condition, (here W is message bandwidth and ω_c is carrier frequency both in rad / sec)

(a) $RC < \dfrac{1}{W}$

(b) $RC > \dfrac{1}{W}$

(c) $RC < \dfrac{1}{\omega_c}$

(c) $RC > \dfrac{1}{\omega_c}$

5. A message signal with bandwidth 10kHz is Lower – Side Band SSB modulated with carrier frequency $f_{c1} = 10^6$ Hz. The resulting signal is then passed through a Narrow – Band Frequency Modulator with carrier frequency $f_{c2} = 10^9$ Hz. The bandwidth of the output would be

(a) 4×10^4 Hz (b) 2×10^6 Hz

(c) 2×10^9 Hz (d) 2×10^{10} Hz

06. An Amplitude Modulated signal is given as $x_{AM}(t) = 100(p(t) + 0.5\, g(t)) \cos \omega_c t$ in the interval $0 \le t \le 1$, where $p(t) = u(t) - u(t-1)$ and $g(t) = p(t) * p(t)$ '*' denotes convolution. One set of possible values of the modulating signal and modulation index would be

(a) t, 0.5 (b) t, 1.0

(c) t, 2.0 (d) t^2, 0.5

Statement for Linked Answer Questions 07 & 08:

Consider the following Amplitude Modulated (AM) signal, where $f_m < B$:

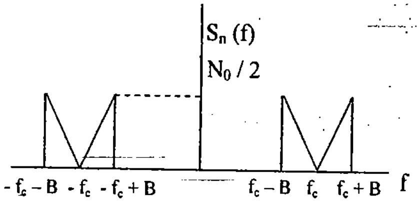
$x_{AM}(t) = 10(1 + 0.5 \sin 2\pi f_m t) \cos 2\pi f_c t$

07. The average side – band power for the AM signal given above is

(a) 25 (b) 12.5

(c) 6.25 (d) 3.125

08. The AM signal gets added to a noise with Power Spectral Density $S_n(f)$ given in the figure below. The ratio of average sideband power to mean noise power would be:



(a) $\frac{25}{8 N_0 B}$ (b) $\frac{25}{4 N_0 B}$
(c) $\frac{25}{2 N_0 B}$ (c) $\frac{25}{N_0 B}$

09. Which of the following analog modulation scheme requires the minimum transmitted power and minimum channel bandwidth?

(a) VSB (b) DSB – SC
(c) SSB (d) AM

10. A device with input x (t) and output y (t) is characterized by: $y(t) = x^2(t)$. An FM signal with frequency deviation of 90 kHz and modulating signal bandwidth of 5 kHz is applied to this device. The bandwidth of the output signal is

(a) 370 kHz (b) 190 kHz
(c) 380 kHz (d) 95 kHz

11. A carrier is phase modulated (PM) with frequency deviation of 10 kHz by a single tone frequency of 1 kHz. If the single tone frequency is increased to 2 kHz, assuming that phase deviation remains unchanged, the bandwidth of the PM signal is

(a) 21 kHz (b) 22 kHz
(c) 42 kHz (d) 44 kHz

12. For a message signal $m(t) = \cos(2\pi f_m t)$ and carrier of frequency f_c, which of the following represents a single side – band (SSB) signal?

(a) $\cos(2\pi f_m t)\cos(2\pi f_c t)$

(b) $\cos(2\pi f_c t)$

(c) $\cos[2\pi(f_c + f_m)t]$

(d) $[1 + \cos(2\pi f_m t)]\cos(2\pi f_c t)$

13. A message signal given by $m(t) = (1/2)\cos\omega_1 t - (1/2)\sin\omega_2 t$ is amplitude – modulated with a carrier of frequency ω_c to generate $s(t) = [1 + m(t)]\cos\omega_c t$
What is the power efficiency achieved by this modulation scheme?

(a) 8.33 % (b) 11.11%
(c) 20 % (d) 25 %

14. A communication channel with AWGN operating at a signal to noise ratio SNR >> 1 and bandwidth B has capacity C_1. If the SNR is doubled keeping B constant, the resulting capacity C_2 is given by

(a) $C_2 \approx 2C_1$ (b) $C_2 \approx C_1 + B$
(c) $C_2 \approx C_1 + 2B$ (d) $C_2 \approx C_1 + 0.3B$

15. An AM signal is detected using an envelope detector. The carrier frequency and modulating signal frequency are 1 MHz and 2 KHz respectively. An appropriate value for the time constant of the envelope detector is

a) 500 μ sec b) 20 μ sec

c) 0.2 μ sec d) 1 μ sec

16. An AM signal and a narrow-band FM signal with identical carriers, modulating signals and modulation indices of 0.1 are added together. The resultant signal can be closely approximated by

a) broadband FM

b) SSB with carrier

c) DSB-SC

d) SSB without carrier

17. A 1 m W video signal having a bandwidth of 100 MHz is transmitted to a receiver through a cable that has 40 dB loss. If the effective one-sided noise spectral density at the receiver is 10^{-20} W/Hz, then the signal-to-noise ratio at the receiver is

a) 50 dB b) 30 dB

c) 40 dB d) 60 dB

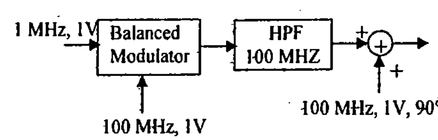
18. A 100 MHz carrier of 1 V amplitude and a 1 MHz modulating signal of 1 V amplitude are fed to a balanced modulator. The output of the modulator is passed through an ideal high-pass filter with cut-of frequency of 100 MHz. The output of the filter is added with 100 MHz signal of 1 V amplitude and 90 ⁰ phase-shift. The envelope of the resultant signal is

a) constant

b) $\sqrt{1 + \sin(2\pi \times 10^6 t)}$

c) $\sqrt{5/4 - \sin(2\pi \times 10^6 t)}$

d) $\sqrt{5/4 + \cos(2\pi \times 10^6 t)}$



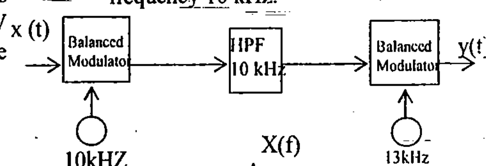
19. Two sinusoidal signals of same amplitude and frequencies 10 KHz and 10.1 KHz are added together. The combined signal is given to an ideal frequency detector. The output of the detector is
a) 0.1 KHz sinusoid
b) 20.1 KHz sinusoid
c) a linear function of time
d) a constant

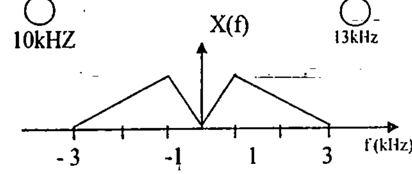
20. Choose the correct one from among the alternatives A, B, C, and D after matching an item from Group 1 with the most appropriate item in Group 2.

Group 1	Group 2
1. FM	P : Slope overload
2. DM	Q : μ - law
3. PSK	R : Envelope detector
4. PCM	S : Capture effect
	T : Hilbert transform
	U : Matched filter

(a) 1 – T, 2 – P, 3 – U, 4 – S
(b) 1 – S, 2 – U, 3 – P, 4 – T
(c) 1 – S, 2 – P, 3 – U, 4 – Q
(d) 1 – U, 2 – R, 3 – S, 4 – Q

21. Consider a system shown in fig. Let X(f) and Y(f) denote the Fourier transforms of x(t) and y(t) respectively. The ideal HPF has the cut-off frequency 10 kHz.





The positive frequencies where Y(f) has spectral peaks are
a) 1 kHz and 24 kHz
b) 2 kHz and 24 kHz
c) 1 kHz and 14 kHz
d) 2 kHz and 14 kHz

22. A DSB-SC signal is to be generated with a carrier frequency $f_c = 1$ MHz using a non-linear device with the input-output characteristic
$v_o = a_0 v_i + a_1 v_i^3$
where a_0 and a_1 are constants. The output of the non-linear device can be filtered by an appropriate band-pass filter. Let $v_i = A_c \cos(2\pi f_c' t) + m(t)$ where m(t) is the message signal. Then the value of 'f_c'' (in MHz) is

a) 1.0 b) 0.333

c) 0.5 d) 3.0

The data for Q.23 – 24 is given below :

Let $m(t) = \cos[(4\pi \times 10^3)t]$ be the message signal and $c(t) = 5\cos[(2\pi \times 10^6)t]$ be the carrier.

23. c(t) and m(t) are used to generate an AM signal. The modulation index of the generated AM signal is 0.5.

Then the quantity $\dfrac{\text{Total sideband power}}{\text{Carrier power}}$ is

a) 1 / 2 b) 1 / 4

d) 1/3 d) 1/8

24. c(t) and m(t) are used to generate an FM signal. If the peak frequency deviation of the generated FM signal is three times the transmission bandwidth of the AM signal, then the coefficient of the term $\cos[2\pi(1008 \times 10^3 t)]$ in the FM signal (in terms of the Bessel coefficients) is

a) $5J_4(3)$ b) $(5/2)\,J_8(3)$

c) $(5/2)\,J_8(4)$ d) $5\,J_4(6)$

25. Choose the correct one from the alternatives A, B, C, D after matching an item in group 1 with the most appropriate item in group 2.

Group 1	Group 2
P: Ring modulator	1 Clock recovery
Q: VCO	2 Demodulation of FM
R Foster-Seely discriminator	3 Frequency conversion
S Mixer	4 Summing the two inputs
	5 Generation of FM
	6 Generation of DSB-SC

(a)	(b)	(c)	(d)
P – 1,	P – 6,	P – 6,	P - 5
Q – 3,	Q – 5	Q – 1,	Q - 6
R – 2,	R – 2,	R – 3,	R - 1
S – 4,	S – 3,	S – 2,	S - 3

26. A superheterodyne receiver is to operate in the frequency range 550 kHz – 1650 kHz, with the intermediate frequency of 450 kHz.

Let $R = \dfrac{C_{max}}{C_{min}}$ denote the required capacitance ratio of the local oscillator and I denote the image frequency (in kHz) of the incoming signal. If the receiver is tuned to 700 kHz, then
a) R = 4.41, I = 1600
b) R = 2.10, I = 1150
c) R = 3.0, I = 1600
d) R = 9.0, I = 1150

27. The input to a coherent detector is DSB – SC signal plus noise. The noise at the detector output is
(a) the in – phase component
(b) the quadrature component
(c) zero
(d) the envelope

28. The noise at the input to an ideal frequency detector is white. The detector is operating above threshold. The power spectral density of the noise at the output is
(a) raised – cosine (b) flat
(c) parabolic (d) Gaussian

29. A linear phase channel with phase delay τ_p and group delay τ_g must have
a) $\tau_p = \tau_g$ = constant
b) $\tau_p \propto f$ and $\tau_g \propto f$
c) τ_p = constant and $\tau_g \propto f$
d) $\tau_p \propto f$ and τ_g = constant

30. A 1 MHz sinusoidal carrier is amplitude modulated by a symmetrical square wave of period 100 μsec. Which of the following frequencies will NOT be present in the modulated signal?
a) 990 KHz b) 1010 KHz
c) 1020 KHz d) 1030 KHz

31. An angle-modulated signal is given by $s(t) = \cos 2\pi(2 \times 10^6 t + 30\sin 150t + 40\cos 150t)$. The maximum frequency and phase deviations of s(t) are
a) 10.5 KHz, 140π rad
b) 6 KHz, 80π rad
c) 10.5 KHz, 100π rad
d) 7.5 KHz, 100π rad

32. A bandlimited signal is sampled at the Nyquist rate. The signal can be recovered by passing the samples through
a) an RC filter
b) an envelope detector
c) a PLL
d) an ideal low-pass filter with the appropriate bandwidth

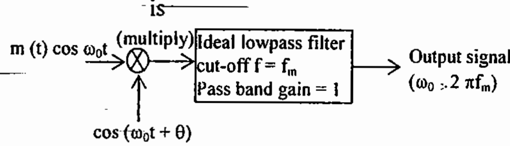
33. The amplitude modulated wave form $s(t) = A_C[1 + K_a m(t)]\cos\omega_c t$ is fed to an ideal envelope detector. The maximum magnitude of $K_a m(t)$ is greater than 1. Which of the following could be the detector output ?

a) $A_C\, m(t)$
b) $A_C^2[1 + K_a m(t)]^2$
c) $|A_C[1 + K_a m(t)]|$
d) $A_C[1 + K_a m(t)]^2$

34. In an FM system, a carrier of 100 MHz is modulated by a sinusoidal signal of 5 KHz. The bandwidth by Carson's approximation is 1 MHz. If $y(t)$ = (modulated waveform)^3, then by using Carson's approximation, the bandwidth of y(t) around 300 MHz and the spacing of spectral components are, respectively

a) 3 MHz, 5 KHz b) 1 MHz, 15 KHz
c) 3 MHz, 15 KHz d) 1 MHz, 5 KHz

35. A message m(t) bandlimited to the frequency f_m has a power of P_m. The power of the output signal in the figure is



(a) $\dfrac{P_m \cos\theta}{2}$ (b) $\dfrac{P_m}{4}$

(c) $\dfrac{P_m \sin^2\theta}{4}$ (d) $\dfrac{P_m \cos^2\theta}{4}$

36. A system has a phase response given by $\phi(\omega)$ where 'ω' is the angular frequency. The phase delay and group delay at $\omega = \omega_0$ are respectively given by

(a) $-\dfrac{\phi(\omega_0)}{\omega_0}, -\dfrac{d\phi(\omega)}{d\omega}\Big|_{\omega=\omega_0}$

(b) $\phi(\omega_0), -\dfrac{d^2\phi(\omega)}{d\omega^2}\Big|_{\omega=\omega_0}$

(c) $\dfrac{\omega_0}{\phi(\omega_0)}, -\dfrac{d\phi(\omega)}{d\omega}\Big|_{\omega=\omega_0}$

(d) $\omega_0\phi(\omega_0)$ $\int_{-\infty}^{\omega_0} \phi(\lambda)\, d\lambda$

37. The input to a channel is a band pass signal. It is obtained by linearly modulating a sinusoidal carrier with a single-tone signal. The output of the channel due to this input is given by
$y(t) = (1/100)\cos(100t - 10^{-6})$
$\cos(10^6 t - 1.56)$
The group delay (t_g) and the phase delay (t_p) in seconds, of the channel are
a) $t_g = 10^{-6}, t_p = 1.56$
b) $t_g = 1.56, t_p = 10^{-6}$
c) $t_g = 10^{-8}, t_p = 1.56 \times 10^{-6}$
d) $t_g = 10^{-8}, t_p = 1.56$

38. A modulated signal is given by,
$s(t) = m_1(t)\cos(2\pi f_c t) + m_2(t)$
$\sin(2\pi f_c t)$ where the baseband signals $m_1(t)$ and $m_2(t)$ have bandwidths of 10 kHz and 15 kHz, respectively. The bandwidth of the modulated signal, in kHz, is
a) 10 b) 15 c) 25 d) 30

39. A modulated signal is given by
$s(t) = e^{-at}\cos[(\omega_c + \Delta\omega)t]\,u(t)$,
where a, ω_c and $\Delta\omega$ are positive constants, and $\omega_c >> \Delta\omega$. The complex envelope of s(t) is given by
a) $\exp(-at)\exp[j(\omega_c + \Delta\omega)t]\,u(t)$
b) $\exp(-at)\exp(j\Delta\omega t)\,u(t)$
c) $\exp(j\Delta\omega t).\,u(t)$
d) $\exp[j(\omega_c + \Delta\omega)t]$

40. The image channel selectivity of superheterodyne receiver depends upon ---
a) IF amplifiers only
b) RF and IF amplifiers only
c) Preselector, RF and IF amplifiers
d) Preselector, and RF amplifiers only

41. A DSB-SC signal is generated using the carrier $\cos(\omega_c t + \theta)$ and modulating signal x(t).
The envelope of the DSB-SC signal is
a) x(t)
b) | x(t) |
c) only positive portion of x(t)
d) $x(t)\cos\theta$

42. The image channel rejection in a superheterodyne receiver comes from
a) IF stages only
b) RF stages only
c) detector and RF stages only
d) detector RF, and IF stages

43. An FM signal with a modulation index 9 is applied to a frequency tripler. The modulation index in the output signal will be ____
a) 0 b) 3 c) 9 d) 27

44. The image (second) channel selectivity of a superheterodyne communication receiver is determined by
a) antenna and preselector
b) the preselector and RF amplifier
c) the preselector and IF amplifier
d) The RF and IF amplifier

45. A PLL can be used to demodulate
a) PAM signals b) PCM signals
c) FM signals d) DSB-SC signals

Match the following : (Q. No. 46 – 47)

46. a) AM system 1) Coherent detection
b) DSB-SC system 2) Envelope detection
c) PAM system 3) Correlation detection
4) PLL
5) LPF

47. a) AM system 1) B (Bandwidth of the modulating signal)
b) SSB system 2) 2B
c) PCM (n bit) system 3) Between B and 2B
4) 2nB
5) nB

48. $v(t) = 5[\cos(10^6\pi t) - \sin(10^3\pi t) \times \sin(10^6\pi t)]$ represents
a) DSB suppressed carrier signal
b) AM signal
c) SSB upper sideband signal
d) Narrow band FM signal

49. A 10 MHz carrier is frequency modulated by a sinusoidal signal of 500 Hz, the maximum frequency deviation being 50 KHz. The bandwidth required, as given by the Carson's rule is________KHz

50. Match the following
a) SSB 1) Envelope detector
b) AM 2) Integrate and dump
c) BPSK 3) Hilbert transform
4) Ratio detector
5) PLL

51. Which of the following demodulator(s) can be used for demodulating the signal $x(t) = 5(1 + 2\cos 200\pi t)\cos 20000\pi t$.
a) Envelope demodulator
b) Square-law demodulator
c) Synchronous demodulator
d) None of the above

52. A superheterodyne radio receiver with an intermediate frequency of 455 KHz is tuned to a station operating at 1200 KHz. The associated image frequency is ________ KHz.

53. The maximum power efficiency of an AM modulator is ---
a) 25 % b) 50 %
c) 75 % d) 100 %

54. In commercial TV transmission in India, picture and speech signals are modulated respectively as :
(Picture) (Speech)
a) VSB and VSB
b) VSB and SSB
c) VSB and FM
d) FM and VSB

55. A 4 GHz carrier is DSB SC modulated by a low pass message signal with maximum frequency of 2 MHz. The resultant signal is to be ideally sampled. The minimum frequency of the sampling impulse train should be:
a) 4 MHz b) 8 MHz
c) 8 GHz d) 8.004 GHz

56. Which of the following schemes suffer(s) from the threshold effect?
(a) AM detection using envelope detection.
(b) AM detection using synchronous detection.
(c) FM detection using a discriminator.
(d) SSB detection with synchronous detection.

57. A signal $x(t) = 2\cos(\pi\, 10^4 t)$ Volts is applied to an FM modulator with the sensivity constant of 10 kHz / volt. Then the modulation index of the FM wave is:
(a) 4 (b) 2
(c) $4/\pi$ (d) $2/\pi$

58. A part of a communication system consists of an amplifier of effective noise temperature, $T_e = 21$ K, and a gain of 13 dB, followed by a cable with a loss of 3 dB. Assuming the ambient temperature to be 300 K, we have for this part of the communication system,

(a) effective noise temperature = 30 K.
(b) effective noise temperature = 36 K.
(c) noise figure = 0. 49 dB.
(d) noise figure = 1.61 dB

59. In a superheterodyne AM receiver, the image channel selectivity is determined by:

(a) The preselector and RF stages
(b) The preselector, RF and If stages
(c) The IF stages
(d) All the stages

60. In a radar receiver the antenna is connected to the receiver through a waveguide. Placing the preamplifier on the antenna side of the waveguide rather than on the receiver side leads to:

(a) A reduction in the overall noise figure.
(b) A reduction in interference
(c) An improvement in selectivity characteristics.
(d) An improvement in directional characteristics.

61. A carrier A_c COS W_c t is frequency modulated by a signal E_m. COS W_m t. The modulation index is m_f. The expression for the resulting FM signal is:

(a) $A_c \cos [w_c t + m_f \sin w_m t]$
(b) $A_c \cos [w_c t + m_f \cos w_m t]$
(c) $A_c \cos [w_c t + 2\pi m_f \cos w_m t]$
(d) $A_c \cos [w_c t + \frac{2\pi m_f E_m}{w_m} \cos w_m t]$

Chapter – 4 Fundamentals of information theory and channel capacity theorem:

Objective Questions

01. A memoryless source emits n symbols each with a probability p. The entropy of the source as a function of n

(a) increases as log n
(d) decreases as log (1 / n)
(c) increases as n
(d) increases as n log n

02. Consider a Binary Symmetric Channel (BSC) with probability of error being p. To transmit a bit, say 1, we transmit a sequence of three 1s. The receiver will interpret the received sequence to represent 1 if at least two bits are 1: The probability that the transmitted bit will be received in error is

(a) $p^3 + 3p^2(1 - p)$ (b) p^3
(c) $(1 - p)^3$ (d) $p3+p^2(1-p)$

03. A source generates three symbols with probabilities 0.25, 0.25, 0.50 at a rate of 3000 symbols per second. Assuming independent generation of symbols, the most efficient source encoder would have average bit rate as ______

(a) 6000 bits /sec
(b) 4500 bits /sec
(c) 3000 bits /sec
(d) 1500 bits /sec

04. A video transmission system transmits 625 picture frames per second. Each frame consists of a 400 × 400 pixel grid with 64 intensity levels per pixel. The date rate of the system is
a) 16 Mbps b) 100 Mbps
c) 600 Mbps d) 6.4 Gbps

05. Source encoding in a data communication system is done in order to
a) enhance the information transmission rate
b) reduce the transmission errors
c) conserve the transmitted power
d) facilitate clock recovery in the receiver.

06. A binary source has symbol probabilities 0.8 and 0.2. If extension coding (blocks of 4 symbols) is used, the lower and upper bounds on the average code word length are
a) lower ______
b) higher ______

07. An image uses 12 × 512 picture elements. Each of the picture elements can take any of the 8 distinguishable intensity levels. The maximum entropy in the above image will be:

a) 2097152 bits b) 786432 bits
c) 648 bits d) 144 bits

08. A source produces 4 symbols with probabilities 1 / 2, 1/ 4, 1 / 8, and 1 / 8. For this source, a practical coding scheme has an average codeword length of 2 bits /symbols. The efficiency of the code is:
(a) 1 (b) 7 / 8
(c) 1 / 2 (d) 1 / 4

09. A source deliver symbols X_1, X_2, X_3 and X_4 with probabilities 1/2, 1/4, 1/8 and 1/8 respectively. The entropy of the system is

(a) 1.75 bits per second

(b) 1.75 bits per symbol

(c) 1.75 symbols per second

(d) 1.75 symbols per bit

10. A telephone channel has bandwidth B of 3 kHz and SNR ($S/\eta B$) of 30 dB. It is connected to a teletype machine having 32 different symbols. The symbol rate required for errorless transmission is nearly

(a) 1800 symbols/s

(b) 3000 symbols/s

(c) 5000 symbols/s

(d) 6000 symbols/s

11. In a single error correcting Hamming code, the number of message bits in a block is 26. The number of check bits in the block would be

(a) 3 (b) 4 (c) 5 (d) 7

12. To permit the selection of 1 out of 16 equiprobable events, the number of bits required is

(a) 2 (b) $\log_{10} 16$

(c) 8 (d) 4

13. When the channel is noisy, producing a conditional probability of error p = 0.5; the channel capacity and entropy function would be, respectively.

(a) 1 and 1 (b) 1 and 0.5

(d) 0.5 and 1 (d) zero and 1

14. A ternary source produces alphabets A, B and C with probabilities $P_A = P_B = p$ and P_C. Which one of the following gives the correct values for the maximum value of the entropy of the source: and the corresponding value of p and the range of p?

(a) 1.58, 0.33, (0,0.5)

(b) 1.0, 0.5, (0,1)

(c) 3.0, 0.67, (0,0.5)

(d) 2.0, 4.2, (0,0.3)

15. In order to permit the selection of 1 out 16 equiprobable events, what is the number of bits required?

(a) 8 (b) 4

(c) $\log_{10} 16$ (d) 2

16. Match List – I with List – II and select the correct answer using the code given below the lists:

List – 1

A. Entropy coding
B. Channel capacity
C. Minimum length code
D. Equivocation

List – II

1. McMillan's Rule
2. Redundancy
3. Shannon Fano
4. Shannon law

Codes:

	A	B	C	D
(a)	1	2	3	4
(b)	3	4	1	2
(c)	1	4	3	2
(d)	3	2	1	4

17. A communication channel has a bandwidth of 100 MHz. The channel is extremely noisy such that the signal power is very much below the noise power. What is the capacity of this channel?

(a) 100 Mbps (b) 50 Mbps
(c) 2400 bps (d) Nearly 0 bps

18. A source generates four messages with probability 1/8, 1/8, 1/4, and 1/2. What is the entropy of the source (bits / message)?

(a) 1 (b) 1.75 (c) 2 (d) 4

19. A source produces 26 symbols with equal probability. What is the average information produced by this source?

(a) < 4 bits / symbol
(b) 6 bits / symbol
(c) 8 bits / symbol
(d) Between 4 and 6 bits / symbol

20. A good line code should have which of the following?

1. Favorable psd
2. Low intersymbol interference
3. Adequate timing content
4. Transparency

Select the correct answer using the code given below:

(a) 1, 3 and 4 (b) 1, 2 and 4
(c) 2, 3 and 4 (d) 1, 2 and 3

21. Which one of the following is correct?

(a) Coding reduces the noise in the signal
(b) Coding deliberately introduces redundancy into messages
(c) Coding increases the information rate
(d) Coding increases the channel bandwidth

22. Which one of the following is the code that is very close to 'trellis code modulation'?

(a) Combines analog and digital modulations
(b) Combines modulation and encoding
(c) Encodes following trellis diagram
(d) Combines amplitude and frequency modulation

23. Source S_1 Produces 4 discrete symbols with equal probability. Source S_2 produces 6 discrete symbols with equal probability If H_1 and H_2 are the entropies of sources S_1 and S_2 respectively, then which one of the following is correct?

(a) H_1 is always less than H_2
(b) H_1 is always greater than H_2
(c) H_1 is always equal to H_2
(d) H_2 is 1.5 times H_1 only

24. The entropy of a digital source is 2.7 bits/ symbol. It is producing 100 symbols per second. The source is likely to be which one of the following?

(a) A binary source
(b) A quatenary source
(c) An octal source
(d) A hexadecimal source

Chapter - 5 Digital Communications

(A) Introduction to digital communication, sampling, PAM, PWM, PPM, PCM, DPCM, and DM

Introduction:

Advantages of Digital communication over Analog communication:

01. Digital communication is more rugged than analog communication because it can withstand channel noise and distortion much better as long as the noise and the distortion are within limits.

02. The greatest advantage of digital communication over analog communication is the viability of regenerative repeaters in the former.

03. Digital hardware implementation is flexible and permits the use of microprocessors, miniprocessors, digital switching, and large-scale integrated circuits.

04. Digital signals can be coded to yield extremely low error rates and high fidelity as well as privacy.

05. It is easier and more efficient to multiplex several digital signals.

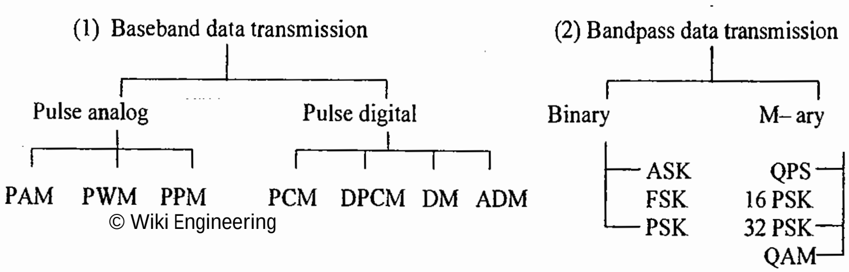
06. Digital communication is inherently more efficient than analog in realizing the exchange of SNR for bandwidth.

07. Digital signal storage is relatively easy and inexpensive. It also has the ability to search and select information from distinct electronic storehouses.

08. Reproduction with digital messages is extremely reliable without deterioration. Analog messages such as photocopies and films, for example, lose quality at each successive stage of reproduction, and have to be transported physically from one distant place to another, often at relatively high cost.

09. The cost of digital hardware continues to halve every two or three years, while performance or capacity doubles over the same time period.

Digital Communication is classified into two types :



The fundamental difference between baseband and bandpass data transmission is with respect to channel. For bandpass, the channel is free space and for baseband, the channel is coaxial cable or fibre optic cable or twisted pair.

Whenever the channel is free space, one can't transit the digital data directly because the spectrum of digital data is a low frequency spectrum. The height of the antenna required is very large. So the baseband is converted into bandpass or vice-versa by using an analog carrier.

Sampling Theorem :

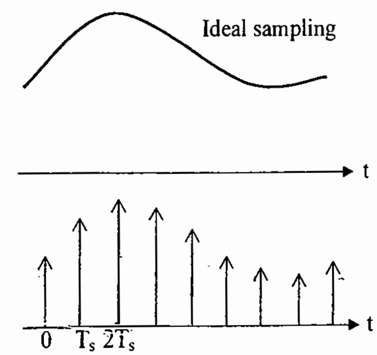
Statement : It sates that "if the highest frequency in the signal spectrum is B, the signal can be reconstructed from its samples, taken at a rate not less than 2B samples per second".

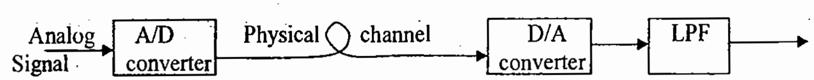
By sampling, the continuous time signal is converted into discrete time signal but the amplitude is constant.

The signal is defined only at 0, T_s, $2T_s$, ...

where T_s is the sampling period.

A/D converter performs both sampling and quantization.
Disadvantage in having more number of samples is the pulse width decreases, it increases the bandwidth required to transmit the information.





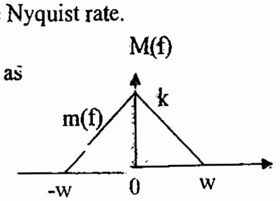
Condition on T_s to get a faithful reproduction of the input analog signal is

$$T_s \le \frac{1}{2W} \quad ; \quad f_s \ge 2W$$

If the number of samples are more, the reconstructed signal is very close to the input signal.

$f_s = 2W$ is called the Nyquist rate.

Let us consider the spectrum as

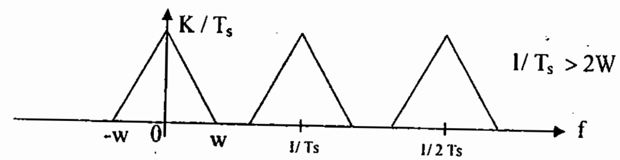


Fourier Transform of Ideally sampled signal is

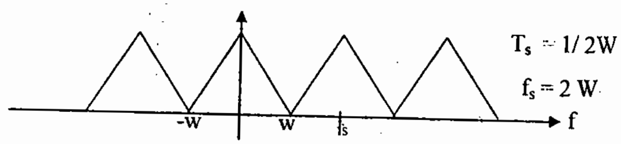
$$= (1/T_s) \sum_{n=-\infty}^{\infty} M(f - n/T_s) \qquad n = 0, \pm 1, \pm 2, \ldots\ldots$$

$$= (1/T_s)\, M(f) + (1/T_s)\, M(f - 1/T_s) + (1/T_s)\, M(f - 2/T_s) + \ldots\ldots$$

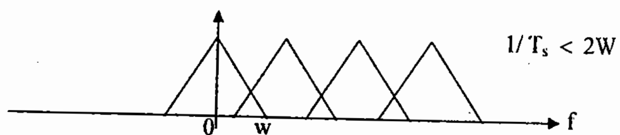
$$+ (1/T_s)\, M(f + 1/T_s) + (1/T_s)\, M(f + 2/T_s) + \ldots\ldots$$



Bandwidth of the signal after sampling is '∞'.

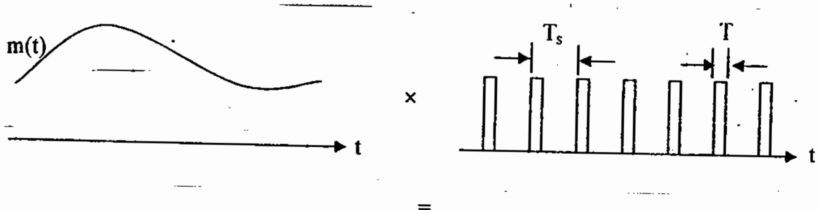


In this case, ideal LPF is used to get back the message signal.

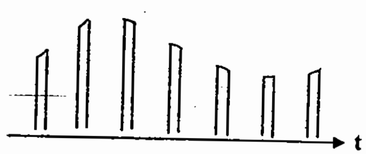


Here, Aliasing effect occurs and we can't get back the original message signal.

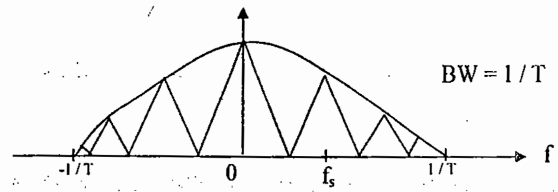
Practical Sampling :



Spectrum of sampling signal



$$= (AT/T_s) \sum_{n=-\infty}^{\infty} \text{sinc}\,(nT/T_s)\, M(f - n/T_s)$$



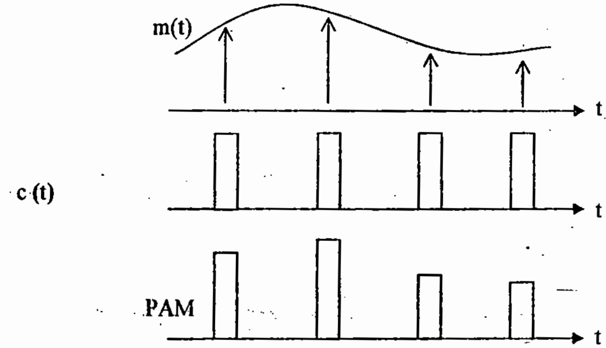
Here message signal can be recovered without distortion

The distortion is minimum whenever the pulse width is very small. After sampling, the sampled value is held constant using sample and hold circuit.

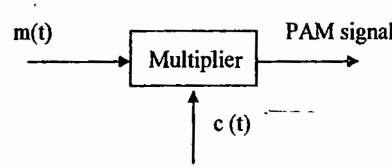
Baseband data transmission:

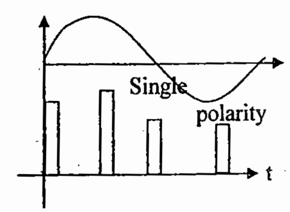
Pulse Analog Modulation (PAM) :

The carrier is high frequency periodic rectangular pulses. In PAM, the amplitude of the pulses is changed according to the sampled value.



Generation of PAM :

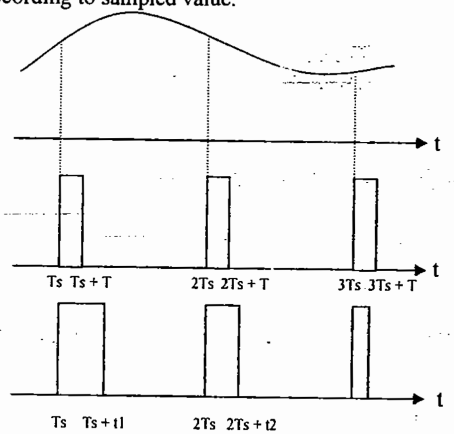




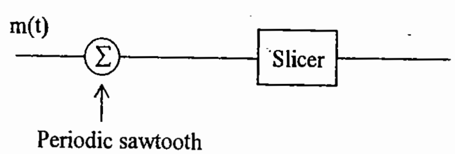
By adding a d.c. level to double polarity PAM, we can get single polarity PAM.

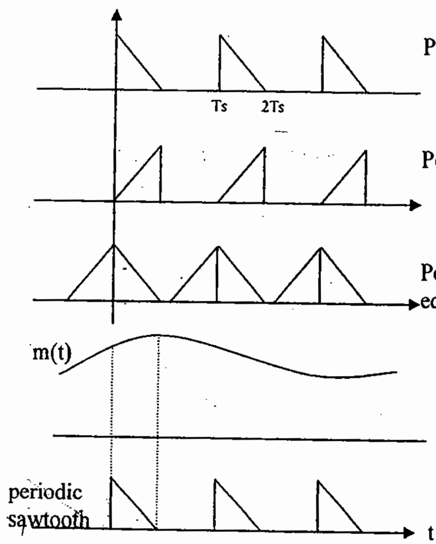
Pulse Width Modulation (PWM) :

The width changes according to the sampled value. We have to change the position of leading edge or trailing edge according to sampled value.



Generation of PWM :

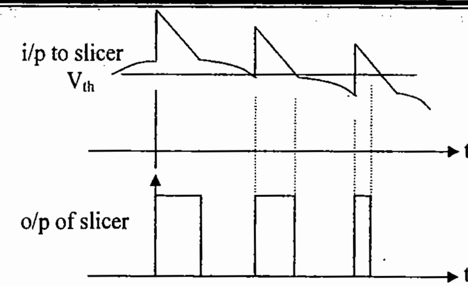




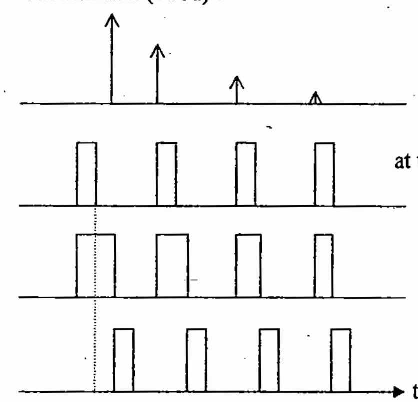
Position of Trailing edge is changed

Position of Leading edge is changed

Positions of both Trailing and Leading edge are changed.



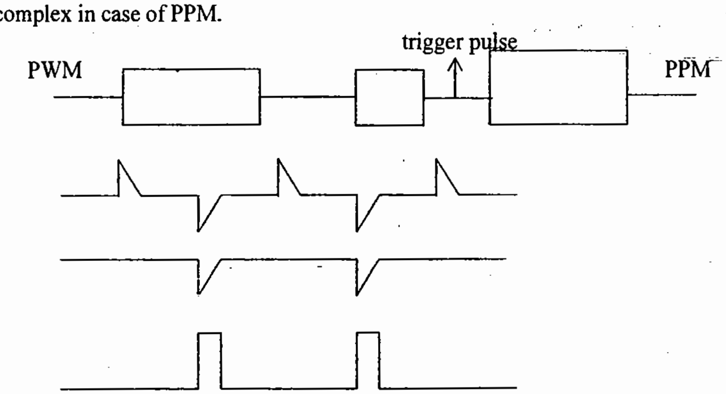
Pulse Position Modulation (PPM) :



PPM waveform should start at trailing edge of PWM waveform

PPM is used in satellite communication.

PAM, PPM, PWM are used in telemetry applications. Generation and demodulation is complex in case of PPM.



If leading edges are modulated, we have to use a forward biased diode and multivibrator with positive trigger pulse.

Pulse Digital Communications :

There are four types of pulse digital communication techniques.

1. PCM 2. DPCM 3. DM 4. ADM

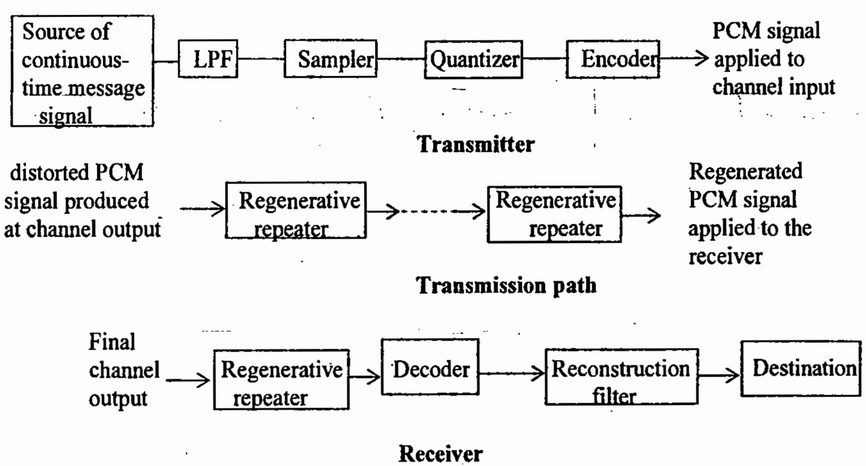
Pulse Code Modulation (PCM) :

In PCM, *a message signal is represented by a sequence of coded pulses, which is accomplished by representing the signal in discrete form in both time an amplitude.* The LPF prior to sampling is included to prevent aliasing of the message signal. The basic operations performed in the transmitter of a PCM sampling system are sampling, quantizing, and encoding. The quantizing and encoding operations are usually performed in the same circuit, which is called an *analog-to-digital converter.*

The incoming message signal is sampled with a train of narrow rectangular pulses so as to closely approximate the instantaneous sampling process. To ensure perfect reconstruction of the message signal at the receiver, the sampling rate must be greater than twice the highest frequency component W of the message signal in accordance with the sampling theorem.

Quantizer converts a continuous amplitude signal but discrete in time to a discrete amplitude discrete time signal.

The *encoding* process is used to translate the discrete set of sample values to a more appropriate form of signals.



Without quantization, it is not possible to have a unique code for each sampled value. Quantizer rounds off the each sampled value to nearest quantizer level.

Regeneration of data is possible in case of digital communications. This is the major advantage over Analog communications.

For n-bit PCM,

Number of quantization levels $L = 2^n$

$$\text{Step size } \delta = \frac{V_{max} - V_{min}}{L}$$

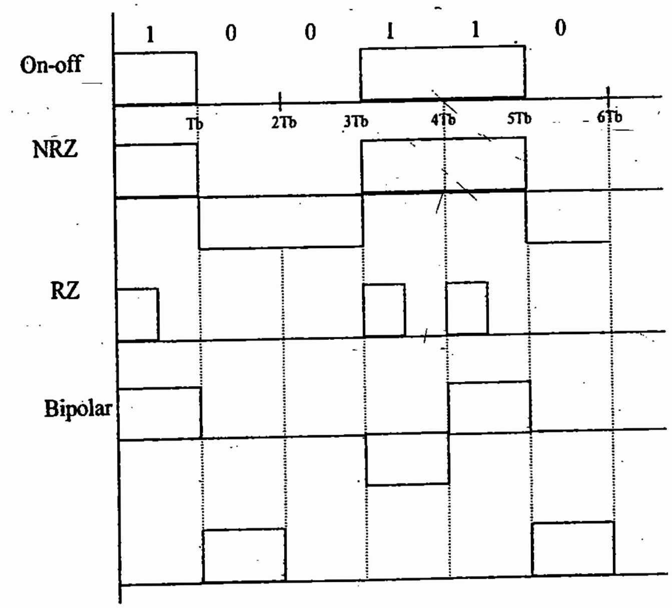
$$\text{Bit rate } (r_b) = \text{Sampling rate} \times n = \frac{1}{T_s} \times n = \frac{n}{T_s}$$

$$\text{Bit duration } (T_b) = \frac{T_s}{n} = \frac{1}{r_b}$$

$$\text{Max. B.W} = \frac{n}{T_s} = \frac{1}{T_b}$$

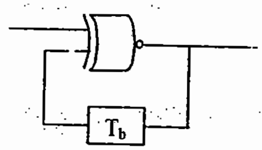
Electrical Representation of Binary data :

(1) On-off signalling
(2) NRZ (Non-return to zero)
(3) RZ (Return to zero)
(4) Bipolar signalling
(5) Differential encoding
(6) Manchester (split phase) encoding



Differential Encoding :

The binary data is encoded first and is represented in on-off signalling.

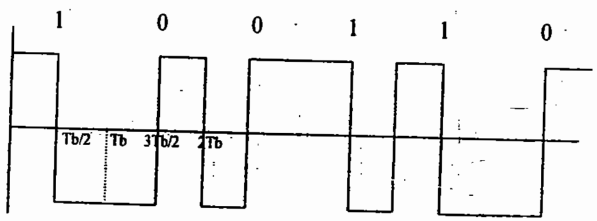


1 – previous state
0 – previous level is complemented

Input			1	0	0	1	1	0
Output	0	1	0	1	0	0	0	1

Manchester Coding :

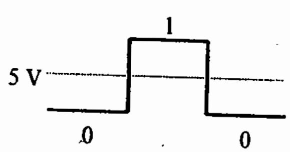
1	$T_b/2$	+Ve
	$T_b/2$	–Ve
0	$T_b/2$	–Ve
	$T_b/2$	+Ve



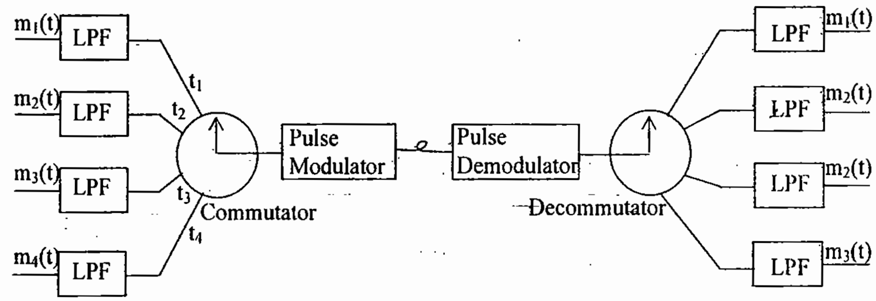
For NRZ signaling, the probability of error is minimum. Thus for PCM, NRZ signaling is generally used.

Regenerative Repeater :

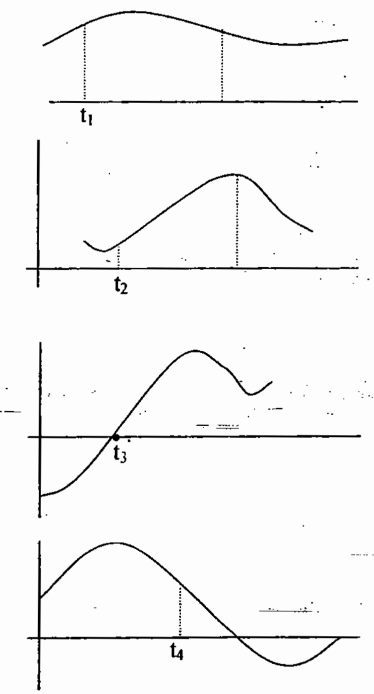
It is used to eliminate the effect of channel noise. It is nothing but one bit quantizer.

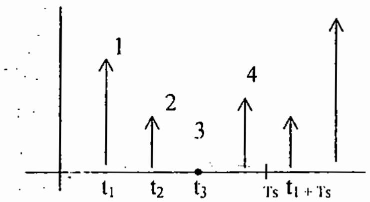


Time division Multiplexing (TDM) :



For bandlimiting of signals, Low pass filters are used at the input. The purpose of commutator is to sample the signals. The time taken to complete one full revolution is T_s.





T_s should be fixed in such a way that the scheme should be capable of reproducing all the signals at the receiver faithfully.

Synchronization :

Number of signals = N

Number of bits in frame = n N

For each frame some extra bits are added for synchronization purpose.

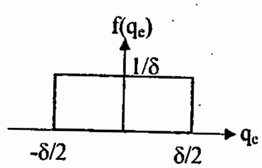
In T_s duration, we get n N + a bits, where a = No. of bits used for synchronization

$$\text{Bit duration } T_b = \frac{T_s}{n} \quad \text{without multiplexing}$$

$$= \frac{T_s}{n\,N} \quad \text{with multiplexing}$$

$$= \frac{T_s}{n\,N + a} \quad \text{with multiplexing \& synchronization}$$

Signal to Quantization Noise Ratio :



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

$$\text{Signal Power} = A_m^2 / 2$$

$$\text{Noise Power} = \int_{-\infty}^{\infty} x^2 f(x)\, dx$$

$$\text{Quantization Noise Power} = \int_{-\delta/2}^{\delta/2} q_e^2 \,.\, f(q_e)\, dq_e$$

$$= \int_{-\delta/2}^{\delta/2} q_e^2 \,.\, (1/\delta_e)\, dq_e = \frac{\delta^2}{12}$$

$$\text{SQNR} = \frac{A_m^2/2}{\delta^2/12}$$

$$\delta = \frac{2\,A_m}{2^n}$$

$$\text{SQNR} = \frac{A_m^2/2}{(4\,A_m^2/2^{2n})\,12} = (3/2)\,2^{2n}$$

$$\text{SQNR}_{dB} = 1.8 + 6n$$

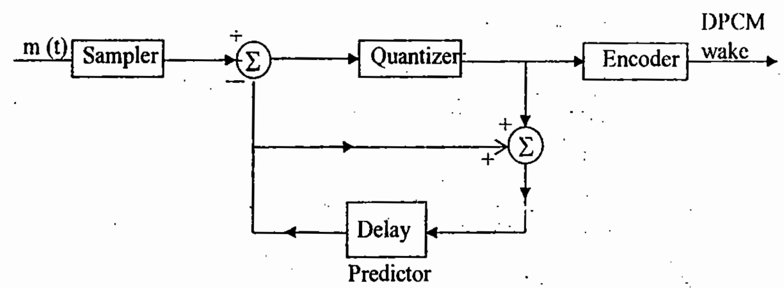
Example : In PCM system, the minimum signal to quantization noise ratio should be 23 dB. Calculate the number of quantization levels required.

Sol : $\text{SQNR} = 1.8 + 6n = 23$

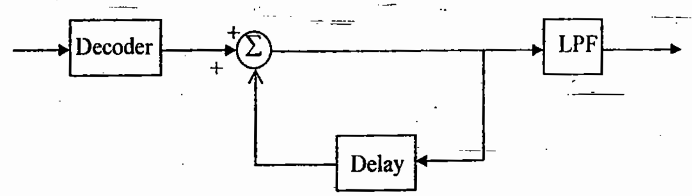
$\Rightarrow \quad n = 4$

$\therefore$ No. of quantization levels $= 2^n = 2^4 = 16$

DPCM (Differential Pulse Code Modulation) :



Transmitter



Receiver

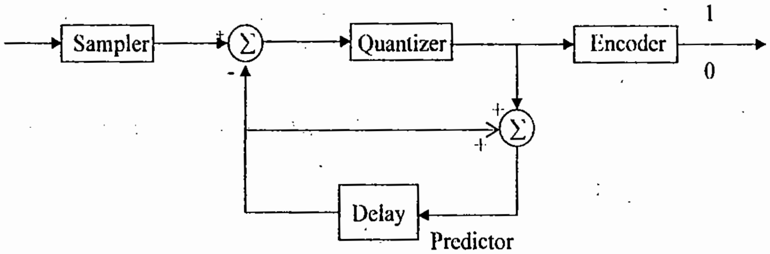
In case of delay, input to the quantizer is present sample and previous sample.
In case of predictor, input to the quantizer is sampled value and its predicted value.

Predictor : It consists of delay elements. By observing the previous sampled values, it predicts the present value.

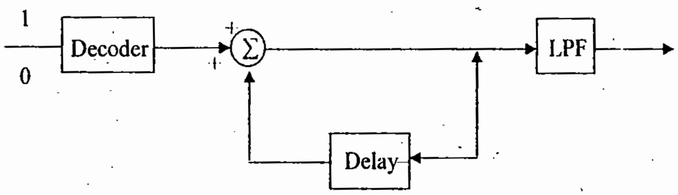
$$\text{Bit rate} = \frac{n}{T_s} = \text{BW}$$

Delta Modulation (DM) :

It is used to decrease the bandwidth. Block schematic is similar to DPCM but the encoder is 1 bit encoder.



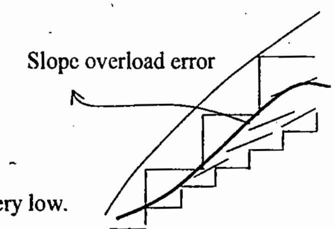
Transmitter



Receiver

(1) Slope Overload error

(2) Granular noise



The input to the LPF is in multiples of δ.
Slope overload error occurs when the step size is very low.
Granular noise occurs when the step size is high.

When the slope of the transmitted waveform and slope of the reconstructed waveform are same then there will be no error.

$$\text{Slope of reconstructed waveform} = \frac{\delta}{T_s}$$

For no slope overloading error,

$$\frac{d}{dt} m(t) = \frac{\delta}{T_s}$$

If the slope of the reconstructed signal is low or high compared to d/dt m(t) then slope overload error (or Granular noise) occurs.

$$\frac{\delta}{T_s} < \frac{d}{dt} m(t) \qquad \text{Slope Overload error}$$

$$\frac{\delta}{T_s} > \frac{d}{dt} m(t) \qquad \text{Granular Noise}$$

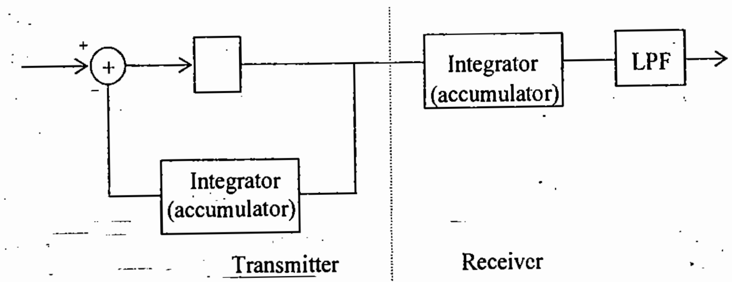
For $m(t) = A_m \cos(2\pi f_m t)$

$$\frac{d}{dt} m(t) = -A_m 2\pi f_m \sin 2\pi f_m t$$

$$\left.\frac{d}{dt} m(t)\right|_{max} = A_m 2\pi f_m$$

$$\boxed{\frac{\delta}{T_s} = A_m 2\pi f_m}$$

Adaptive Delta Modulation (ADM) : ADM involves additional hardware designed to provide *variable step size*, thereby reducing slope-overload effects without increasing the granular noise.



Chapter – 5A **Objective Questions** **SET - A**

01. The maximum permissible duration between two samples of a 2 KHz signal is
   a) 1000 μ sec b) 500 μ sec
   c) 250 μ sec d) None

02. Pick the odd man out
   a) PWM b) PPM
   c) PDM d) PLM

03. The main advantage of TDM over FDM is that it
   a) needs less power
   b) needs less bandwidth
   c) needs simple circuitry
   d) gives better S/N ratio

04. The PWM needs
   a) more power than PPM
   b) more samples per second than PPM
   c) more bandwidth than PPM
   d) None of the above

05. The PAM signal can be detected by
   a) bandpass filter
   b) bandstop filter
   c) high pass filter
   d) low pass filter

State True or False

06 The guard time between pulses increases transmission efficiency
   (a) True (b) False

07. Noise can be reduced by increasing sampling rate
   (a) True (b) False

08. Pulse stuffing is used in
   a) synchronous TDM
   b) synchronous TDM
   c) any TDM
   d) None of the above

09. In communications, the sampling technique leads to
   a) Better efficiency
   b) Highest speed of communication
   c) Less costly equipment
   d) None

10. The minimum sampling frequency is called
   a) Carlson frequency
   b) Pulse sampling rate
   c) Nyquist sampling rate
   d) None

11. The sampling rate is always between
   a) 0 and W b) W and 2W
   c) 2W and 4W d) None

12. As the sampling frequency is increased, the guard band becomes
   a) Smaller
   b) Remains same
   c) Larger
   d) None

13. Which of the following pulse modulation systems has no carrier-wave equivalent ?
   a) PPM b) PDM
   c) PCM d) None

14. So far as noise is concerned, as compared to direct baseband transmission, PAM is
   a) Better b) Similar
   c) Worst d) Same

15. The sampling in PDM is
   a) Uniform
   b) Non-unform
   c) Dependent on the nature of message signal
   d) None

16. The PDM signal is converted into PPM with the help of a
   a) Monostable
   b) Flip-flop
   c) Timer
   d) None

17. In PDM, the bandwidth requirement is a function of
   a) the position of the pulse
   b) Maximum pulse width
   c) Minimum pulse width
   d) None

18. Which of the following circuits cannot be used to generate PDM ?
   a) Monostable multivibrator
   b) Bistable multivibrator
   c) 555 timer
   d) None

19. Apart from a dc component, the PPM spectrum consists of
   a. AM and PM sidebands of all multiples of 'f'
   b. Spectrum of message derivative
   c. Both of these
   d. d) None

20. To generate PDM, the circuit needed is
   a) Sawtooth generator
   b) clipping and squaring circuit
   c) Both of these
   d) None

Key :

01. c	02. b	03. c	04. a	05. d
06. b	07. b	08. b	09. b	10. c
11. c	12.c	13. c	14. b	15. b
16. b	17. c	18. b	19. c	20.c

Chapter - 5A Objective Questions Set - B

01. Adaptive Delta Modulation is preferred over Delta Modulation because its step size changes as per the requirement
(a) True (b) False

02. In PCM, the quantization noise depends on
a) sampling rate
b) number of quantization levels
c) signal power
d) None of the above

03. Which of the following modulation is digital in nature?
a) PAM b) PPM
c) DM d) None of the above

04. Which of the following modulation is analog in nature?
a) PCM
b) DPCM
c) D M
d) None of the above

05. In PCM, if the number of quantization levels is increased from 4 to 64, then the bandwidth requirement will approximately be
a) 3 times b) 4 times
c) 8 times d) 16 times

06. Quantization noise occurs in
a) PAM b) PPM
c) DM d) None of the above

07. Companding is used in PCM to
a) reduce bandwidth
b) reduce power
c) increase S/N ratio
d) get almost uniform S/N ratio

08. The main advantage of PCM is
a) less bandwidth
b) less power
c) better S/N ratio
d) possibility of multiplexing

09. The main disadvantage of PCM is
a) large bandwidth
b) large power
c) complex circuitry
d) quantization noise

10. The main advantage of DM over PCM is
a) less bandwidth
b) less power
c) better S/N ratio
d) simple circuit

11. In a DM system, the granular noise occurs when modulating signal
a) increases rapidly
b) decreases rapidly
c) changes within the step size
d) has high frequency component

12. In DM, slope overload occurs when the modulating signal changes
a) Less rapidly than the encoder can follow
b) More rapidly than the encoder can follow
c) None of these

13. In DM, in order to keep S/N ratio nearly constant, the step size 'a' is kept small but increases as the input signal
a) Increases b) decreases
c) Remains unaffected d) None

14. As compared to message bandwidth, the PCM bandwidth is
a) much larger b) same
c) much smaller d) None

15. In PCM, the high noise immunity is achieved with
a) increased bandwidth
b) decreased bandwidth
c) None

16. In PCM, quantisation noise appears as
a) thermal noise
b) Shot noise
c) White noise

17. For noise reduction in PCM, the exchange of bandwidth for S/N ratio is
a) Linear
b) Exponential
c) Uniform

18. The bandwidth required for transmitting 4 KHz signal using PCM with 128 quantising levels is
a) 8 KHz b) 16 KHz
c) 24 KHz d) 28 KHz

19. To generate PCM, the signal is sampled and converted into
a) PWM
b) PPM
c) PAM

20. In PCM system, if the bandwidth of a channel is increased, the S/N ratio would
a) Decrease
b) Remain the same
c) Increase

Key :

01. a	02. b	03. c	04. d	05. a	06. c
07. d	08. c	09. a	10. d	11. c	12. b
13. a	14. a	15. a	16. c	17. b	18. c
19. c	20. c				

Chapter – 5A **Objective Questions** **Set - C**

01. In a delta modulation scheme, the step height is 75 mV and step width is 1.5 ms. The maximum slope that the staircase can track is
a) 50 V/s b) 55 V/s c) 60 V/s d) 65 V/s

02. The ramp signal m(t) = a t is applied to a delta modulator with sampling period T_s and step size 'δ'. Slope over load distortion would occur if
a) $\delta < a$ b) $\delta > a$ c) $\delta < a\,T_s$ d) $\delta > a\,T_s$

03. In delta modulation which of the following drawbacks are existing?

1. Slope overload 2. Serration noise 3. Granular noise

a) 1 & 2 only b) 2 & 3 only c) 1 & 3 only d) 1, 2 & 3

04. The signal to quantization noise ratio in an n – bit PCM system

a) depends upon the sampling frequency employed
b) is independent of the value of n
c) decreases with increasing value of n
d) increases with increasing value of n

05. A speech signal occupying the bandwidth of 300 Hz to 3 kHz is converted into PCM format for use in digital communication. If the sampling frequency is 8 kHz and each sample is quantized into 256 levels, then the output bit rate will be
a) 3 kb/s b) 8 kb/s c) 64 kb/s d) 256 kb/s

06. Which of the following pulse communication system is inherently immune to noise?
a) PPM b) PCM c) PWM d) PAM

07. Companding is used
a) to overcome quantising noise in PCM
b) in PWM receivers to reduce impulse noise
c) to protect small signals in PCM from quantising noise
d) None of the above

08. In PCM for 132 standard quantising levels, the maximum error will be
a) (1/66) of the total amplitude range b) (1/132) of the total amplitude range
c) (1/264) of the total amplitude range d) (1/528) of the total amplitude range

09. Quantising noise is produced in
a) FDM b) PCM c) all modulation systems d) all pulse modulation systems

10. For a 10 – bit PCM system, the signal to quantization noise ratio is 62 dB. If the number of bits is increased by 2, then the signal to quantization noise ratio will
a) increase by 6 dB b) increase by 12 dB
c) decrease by 6 dB d) decrease by 12 dB

11. Four voice signals, each limited to 4 kHz and sampled at Nyquist rate, are converted into binary PCM signal using 256 quantization levels. The bit transmission rate for the time – division multiplexed signal will be
a) 8 kbps b) 64 kbps c) 256 kbps d) 512 kbps

12. A message signal band limited to 5 kHz is sampled at the minimum rate as dictated by the sampling theorem. The number of quantization levels is 64. If the samples are encoded in binary form, the transmission rate is
a) 60 kbps b) 50 kbps c) 32 kbps d) 10 kbps

13. The peak-to-peak input to an 8-bit PCM decoder is 2 volts. The signal power-to-quantization noise power ratio (in dB) for an input of 0.5 cos (ω_m t) is
a) 47.8 b) 49.8 c) 95.6 d) 99.6

14. Four independent messages have bandwidths of 100 Hz, 200 Hz, and 400 Hz, respectively. Each is sampled at the Nyquist rate, and the samples are time division multiplexed (TDM) and transmitted. The transmitted sample rate (in Hz) is
a) 1600 b) 800 c) 400 d) 200

15. A TDM link has 20 signal channels and each channel is sampled 8000 times/sec. Each sample is represented by seven binary bits and contains an additional bit for synchronization. The total bit rate for the TDM link is
a) 1180 K bits /sec b) 1280 K bits/sec c) 1180 M bits/sec d) 1280 M bits/sec

16. When the number of quantization levels is increased from 4 to 64, the bandwidth required for the transmission of a PCM signal increases by a factor of
a) 3 b) 4 c) 5 d) 6

Chapter – 5A **Objective Questions** **Set - D**

Common Data for Questions 01, 02 and 03:

A speech signal, band limited to 4 kHz and peak voltage varying between +5V and -5V, is sampled at the Nyquist rate. Each sample is quantized and represented by 8 bits

01. If the bits 0 and 1 are transmitted using bipolar pulses, the minimum bandwidth required for distortion free transmission is

(a) 64 kHz (b) 32 kHz

(c) 8 kHz (d) 4 kHz

02. Assuming the signal to be uniformly distributed between its peak to peak value, the signal to noise ratio at the quantizer output is

(a) 16dB (b) 32 dB

(c) 48 dB (d) 64 dB

03. The number of quantization levels required to reduce the quantization noise by a factor of 4 would be

(a) 1024 (b) 512

(c) 256 (d) 64

04. In delta modulation, the slope overload distortion can be reduced by

(a) decreasing the step size

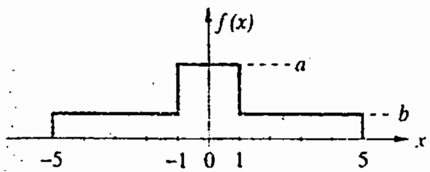
(b) decreasing the granular noise

(c) decreasing the sampling rate

(d) increasing the step size

Statement for Linked Answer Questions 05 & 06:

An input to a 6 – level quantizer has the probability density function f (x) as shown in the figure. Decision boundaries of the quantizer are chosen so as to maximize the entropy of the quantizer output. It is given that 3 consecutive decision boundaries are '– 1', '0' and '1'.



05. The values of a and b are

(a) a = 1/6 and b = 1/12

(b) a = 1/5 and b = 3/40

(c) a = 1/4 and b = 1/16

(d) a = 1/3 and b = 1/24

06. Assuming that the reconstruction levels of the quantizer are the mid-points of the decision boundaries, the ratio of signal power to quantization noise power is

(a) $\frac{152}{9}$ (b) $\frac{64}{3}$

(c) $\frac{76}{3}$ (d) 28

07. A signal m (t) with bandwidth 500 Hz is first multiplied by a signal g (t) where

$$g(t) = \sum_{k=-\infty}^{\infty} (-1)^k \delta(t - 0.5 \times 10^{-4} k)$$

The resulting signal is then passed through an ideal lowpass filter with bandwidth 1 kHz. The output of the lowpass filter would be

(a) δ (t) (b) m (t)

(c) 0 (d) m (t) δ (t)

08. The minimum sampling frequency (in samples / sec) required to reconstruct the following signal from its samples without distortion

$$x(t) = 5\left(\frac{\sin 2\pi 1000t}{\pi t}\right)^3 + 7\left(\frac{\sin 2\pi 1000t}{\pi t}\right)^2$$

would be

(a) 2×10^3 (b) 4×10^3

(c) 6×10^3 (d) 8×10^3

09. The minimum step – size required for a Delta – Modulator operating at 32 K samples /sec to track the signal (here u (t) is the unit – step function)

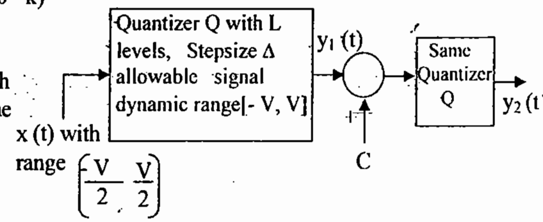
$x(t) = 125t\,(u(t) - u(t-1)) + (250 - 125t)(u(t-1) - u(t-2))$

so that slope –overload is avoided, would be

(a) 2^{-10} (b) 2^{-8}

(c) 2^{-6} (d) 2^{-4}

10. In the following figure the minimum value of the constant "C", which is to be added to y_1 (t) such that y_1 (t) and y_2 (t) are different, is



(a) Δ (b) $\Delta / 2$

(c) $\Delta^2 / 12$ (d) Δ / L

Statement for Linked Answer Questions 11 & 12:

A symmetric three – level midtread quantizer is to be designed assuming equiprobable occurrence of all quantization levels.

11. If the input probability density function is divided into three regions as shown in fig. the value of 'a' in the figure is

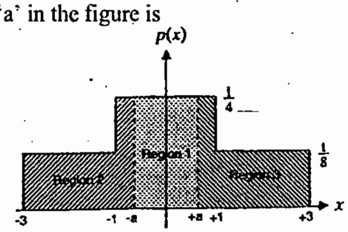


Fig.

(a) 1 /3 (b) 2 / 3

(c) 1 / 2 (d) 1 / 4

12. The quantization noise power for the quantization region between – a and + a in the figure is

(a) 4 / 81 (b) 1 / 9

(c) 5 / 81 (d) 2 / 81

Common Data for Questions 13 and 14:
The amplitude of a random signal is uniformly distributed between -5 V and 5 V.

13. If the signal to quantization noise ratio required in uniformly quantizing the signal is 43.5 dB, the step size of the quantization is approximately
(a) 0.0333 V (b) 0.05 V
(c) 0.0667 V (d) 0.10 V

14. If the positive values of the signal are uniformly quantized with a step size of 0.05 V, and the negative values are uniformly quantized with a step size of 0.1 V, the resulting signal to quantization noise ratio is approximately
(a) 46 dB (b) 43.8 dB
(c) 42 dB (d) 40 dB

15. In a PCM system .if the code word length is increased from 6 to 8 bits, the signal to quantization noise ratio improves by the factor
(a) 8/6 (b) 12 (c) 16 (d) 8

16. In the output of a DM speech encoder, the consecutive pulses are of opposite polarity during time interval $t_1 \le t \le t_2$. This indicates that during this interval
(a) the input to the modulator is essentially constant
(b) the modulator is going through slope overload
(c) the accumulator is in saturation
(d) the speech signal is being sampled at the Nyquist rate

17. A random variable X with uniform density in the interval 0 to 1 is quantized as follows:
If $0 \le X \le 0.3$, $x_q = 0$
If $0.3 \le X \le 1$, $x_q = 0.7$
Where x_q is the quantized value of X. The root –mean value of the quantization noise is
(a) 0.573 (b) 0.198
(c) 2.205 (d) 0.266

18. Choose the correct one from among the alternatives A,B,C, and D after matching an item from group 1 with the most appropriate item in Group 2

Group 1	Group 2
1. FM	P: Slope overload
2. DM	Q: μ-law
3. PSK	R: Envelope detector
4. PCM	S: Capture effect
	T: Hilbert transform
	U: Matched filter

(a) 1-T, 2-P, 3-U, 4-S
(b) 1-S, 2-U, 3-P, 4-T
(c) 1-S, 2-P, 3-U, 4-Q
(d) 1-U, 2-R, 3-S, 4-Q

19. A sinusoidal signal with peak-to-peak amplitude of 1.536 V is quantized into 128 levels using a mid-rise uniform quantizer. The quantization- noise power is
(a) 0.768 V
(b) 48×10^{-6} V^2
(c) 12×10^{-6} V^2
(d) 3.072 V

20. The input to a linear delta modulator having step size $\Delta = 0.628$ is a sine wave with frequency f_m and peak amplitude E_m. If the sampling frequency $f_s = 40$ kHz, the combination of the sine – wave frequency and the amplitude, where slope overload will take place is

E_m	f_m
a) 0.3 V	8 kHz
b) 1.5 V	4 kHz
a) 1.5 V	2 kHz
a) 3.0 V	1 kHz

21. A signal is sampled at 8 kHz and is quantized using 8 – bit uniform quantizer. Assuming SNR_q for a sinusoidal signal, the correct statement for PCM signal with a bit rate of R is
(a) R = 32 kbps, SNRq = 25.8 dB
(b) R = 64 kbps, SNRq = 49.8 dB
(c) R = 64 kbps, SNRq = 55.8 dB
(d) R = 32 kbps, SNRq = 49.8 dB

22. Consider a sampled signal $y(t) = 5 \times 10^{-6}\, x(t)\, \Sigma\, \delta(t - nT_s)$, $n = -\infty$ to ∞ where $x(t) = 10 \cos (8\pi \times 10^3)\, t$ and $T_s = 100$ μsec. When y (t) is passed through an ideal low pass filter with a cutoff frequency of 5 KHz, the ;output of the filter is
(a) $5 \times 10^{-6} \cos (8\pi \times 10^3)\, t$
(b) $5 \times 10^{-5} \cos (8\pi \times 10^3)\, t$
(c) $5 \times 10^{-1} \cos (8\pi \times 10^3)\, t$
(d) $10 \cos (8\pi \times 10^3)\, t$

23. A signal $x(t) = 100 \cos (24\pi \times 10^3)\, t$ is ideally sampled with sampling period of 50μsec and then passed through an ideal low – pass filter with cutoff frequency of 15 KHz. Which of the following frequencies is / are present at the filter output?
(a) 12 KHz only
(b) 8 KHz only
(c) 12 KHz and 9 KHz
(d) 12 KHz and 8 KHz

24. The Nyquist sampling interval, for the signal sinc (700t) + sinc (500t) is
(a) 1 / 350 sec (b) π / 350 sec
(c) 1 / 700 sec (d) π / 175 sec

25. The Nyquist sampling frequency (in Hz) of a signal given by $6 \times 10^{-4}\, \mathrm{sinc}^2 (400)^* \; 10^6\, \mathrm{sinc}^3 (100t)$ is
(a) 200 (b) 300
(c) 500 (d) 1000

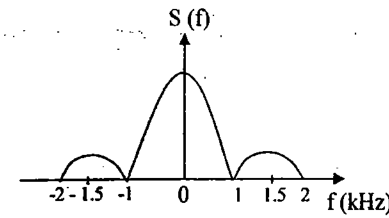
26. The peak – to peak input to an 8 – bit PCM coder is 2 volts. The signal power to quantization noise power ratio (in dB) for an input of $0.5 \cos (\omega_m t)$ is
(a) 47.8 (b) 49.8
(c) 95.6 (d) 99.6

27. In a PCM system with uniform quantization, increasing the number of bits from 8 to 9 will reduce the quantization noise power by a factor of
(a) 9 (b) 8 (c) 4 (d) 2

28. Flat top sampling of low pass signals
(a) gives rise to aperture effect
(b) implies over sampling
(c) leads to aliasing
(d) introduces delay distortion

29. Compression in PCM refers to relative compression of
(a) higher signal amplitudes
(b) lower signal amplitudes
(c) lower signal frequencies
(d) higher signal frequencies

30. A deterministic signal has the power spectrum given in figure. The minimum sampling rate needed to completely represent this signal is



(a) 1 kHz (b) 2 kHz
(c) 3 kHz (d) None of the above

31. The number of bits in a binary PCM system is increased form n to (n + 1). As a result, the signal to quantization noise ratio will improve by a factor
(a) (n + 1) / n
(b) $2^{(n+1)/n}$
(c) $2^{2(n+1)/n}$
(d) Which is independent of n

32. If the number of bits per sample in a PCM system is increased from n to (n + 1) the improvement in signal to quantization noise ratio will be
(a) 3 dB (b) 6 dB
(c) 2n dB (d) n dB

33. A 1.0 kHz signal is flat -- tap sampled at the rate of 1800 samples / sec and the samples are applied to an ideal rectangular LPF with cut – off frequency of 1100 Hz. Then the output of the filter contains
(a) only 800 Hz component
(b) 800 Hz and 900 Hz components
(c) 800 Hz and 1000 Hz components
(d) 800 Hz , 900 Hz and 1000 Hz components

34. The signal to quantization noise ratio in an in – bit PCM system.
(a) depends upon the sampling frequency employed
(b) is independent of the value of ' n'
(c) increases with increasing value of 'n'
(d) decreases with the increasing value of 'n'

Match the following (Q.No. 35. 36.)

35. a) AM system (1) Coherent detection
(b) DSB – SC system (2) Envelope detection
(c) PAM system (3) Correlation detection
(4) PLL
(5) LPF

36. a) AM system 1) B (Bandwidth of the modulating signal)
(b) SSB system 2) 2B
(c) PCM (n bit) system (3) Between B and 2B
(4) 2nB
(5) nB

37. Increased pulse – width in the flat top sampling, leads to
(a) attenuation of high frequencies in reproduction
(b) attenuation of low frequencies in reproduction
(c) greater aliasing errors in reproduction
(d) no harmful effects in reproduction

38. The bandwidth required for the transmission of a PCM signal increases by a factor of _______________ when the number of quantization levels is increased from 4 to 64

39. Six independent low pass signals of bandwidths 3 W, W , 2 W, 3 W and 2 W Hz are to be time – division multiplexed on a common channel using PAM: To achieve this, the minimum transmission bandwidth of the channel should be _______ Hz

40. A signal has frequency components from 300 Hz to 1.8 KHz. The minimum possible rate at which the signal has to be sampled is
____

41. A 4 GHz carrier is DSB SC modulated by a low pass message signal with maximum frequency of 2 MHz. The resultant signal is to be ideally sampled. The minimum frequency of the sampling impulse train should be:
(a) 4 MHz (b) 8 MHz
(c) 8 GHz (d) 8.004 GHz

42. In a BPSK signal detector, the local oscillator has a fixed phase error of 20°. This phase error deteriorates the SNR at the output by a actor of
(a) cos 20° (b) $\cos^2 20°$
(a) cos 70° (b) $\cos^2 70°$

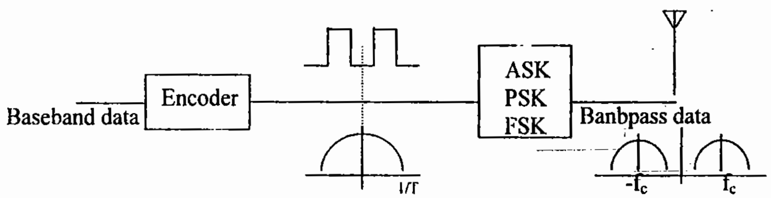
43. A.signal having uniformly distributed amplitude in the interval (- V, + V) is to be encoded using PCM with uniform quantization. The signal – to quantizing noise ration is determined by the
(a) dynamic range of the signal
(b) sampling rate
(c) number of quantizing levels.
(d) power spectrum of the signal.

44. A signal containing only tow frequency components (3kHz and 6 kHz) is sampled at the rate of 8 kHz and then passed through a low pass filter with a cutoff frequency of 8 kHz. The filter output
(a) is an undistorted version of the original signal
(b) contains only the 3 kHz component
(c) contains the 3 kHz component and a spurious component of 2 kHz
(d) contains both the components of the original signal and two spurious components of 2 kHz and 5 kHz

45. Companding in PCM systems leads to improved signal to quantization noise ratio. This improvement is for:
(a) Lower frequency components only
(b) Higher frequency components
(c) Lower amplitudes only
(d) Higher amplitudes only

(B) ASK, PSK, FSK, DPSK :

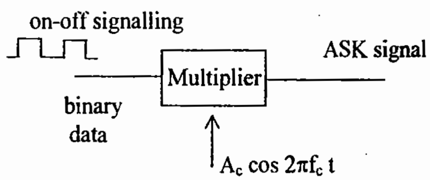
Band pass Data Transmission



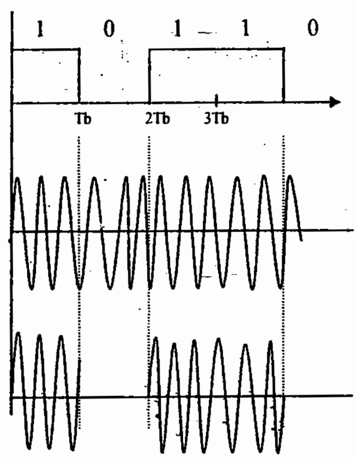
The output of the encoder has significant low frequency components. If it has to radiate through antenna then antenna height will become a problem. So the encoder output is multiplied by an analog carrier.

Digital Carrier Modulation :

(1) **Amplitude Shift Keying (ASK)** .

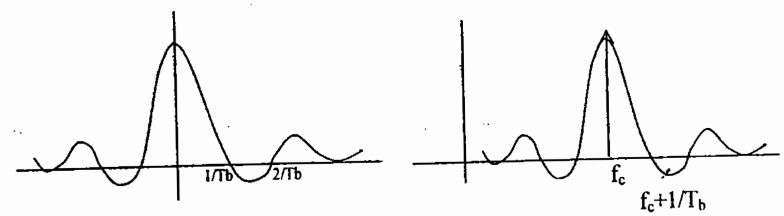


The amplitude of the carrier is modulated according to the message signal.



$s(t) = A_c \cos 2\pi f_c t \qquad 0 \le t \le T_b$

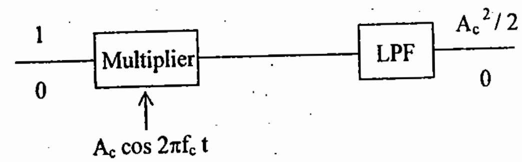
$= 0$



$\text{Bandwidth} = 2 \times 1/T_b$

$= 2 \times \text{bit rate}$

Demodulation :

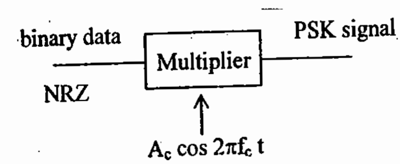


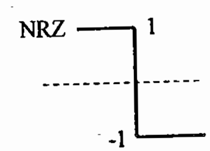
$1 \Rightarrow A_c \cos 2\pi f_c t \times A_c \cos 2\pi f_c t = (A_c^2/2)\,[1 + \cos 4\pi f_c t]$

$\text{Output of LPF} = (A_c^2/2)$

$0 \Rightarrow \text{Output of LPF} = 0$

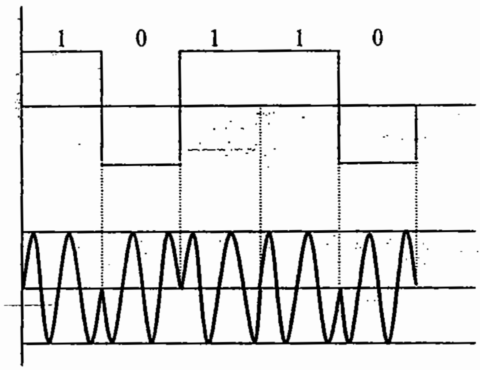
Phase Shift Keying :





$s(t) = A_c \cos 2\pi f_c t \qquad 1$

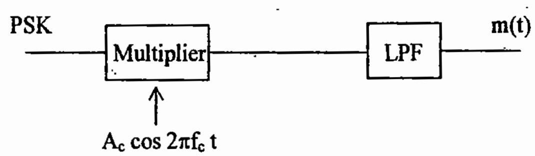
$= -A_c \cos 2\pi f_c t \qquad 0$

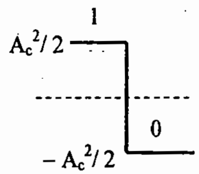


Frequency of the carrier must be a multiple of a bit rate.

$$T_b = n / f_c \quad \Rightarrow f_c = n\, r_b$$

Demodulation :

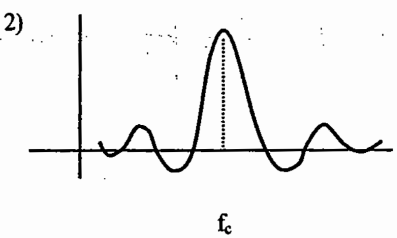




$$1 \Rightarrow A_c \cos 2\pi f_c t \times A_c \cos 2\pi f_c t = (A_c^2/2)\,[1 + \cos 4\pi f_c t\,]$$

Output of LPF $= (A_c^2/2)$

$$0 \Rightarrow \text{Output of LPF} = -(A_c^2/2)$$



Bandwidth $= 2 \times$ bit rate

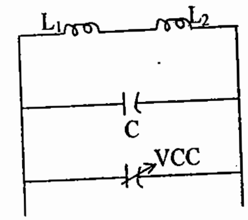
Frequency Shift Keying

$s(t) = A_c \cos 2\pi f_1 t \qquad 1$

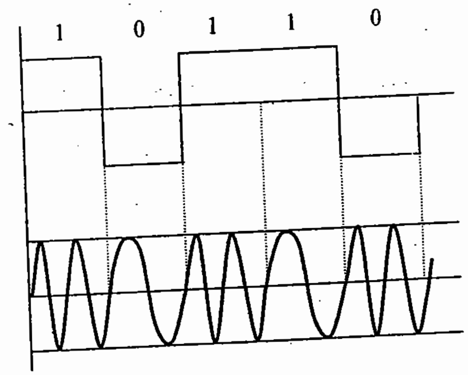
$\quad\;\; = A_c \cos 2\pi f_2 t \qquad 0 \qquad f_1 > f_2$

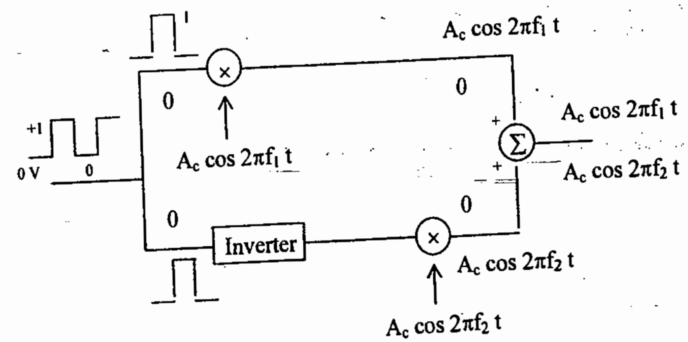
Generation :

VCO :

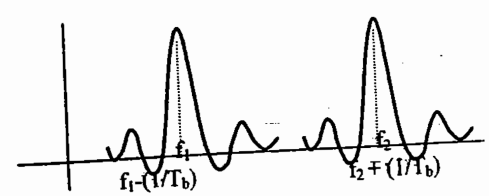


$$f = \frac{1}{2\pi\sqrt{(L_1+L_2)[C + C_1 \text{ or } C_2]}}$$





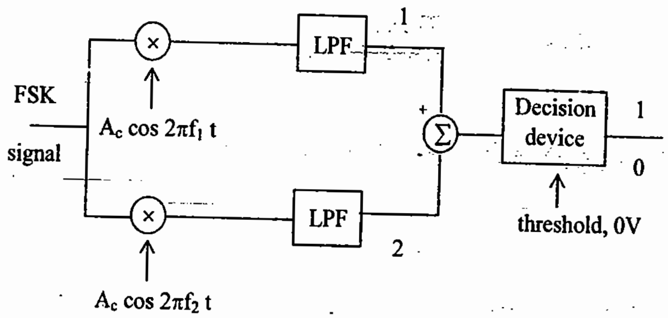
Bandwidth $= 2\,\Delta f + 2\, f_m$



Bandwidth $= f_1 + (1/T_b) - f_2 + (1/T_b)$

$= f_1 - f_2 + (2/T_b)$; $f_1 - f_2 = 2\Delta f$

Demodulation :



Cut-off frequency $< f_1 - f_1$

$1 \Rightarrow A_c \cos 2\pi f_1 t \times A_c \cos 2\pi f_2 t$ 1

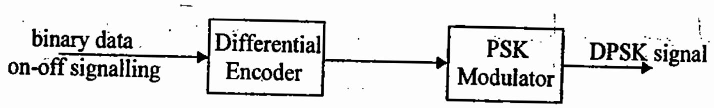
$A_c \cos 2\pi f_1 t \times A_c \cos 2\pi f_2 t$ 2

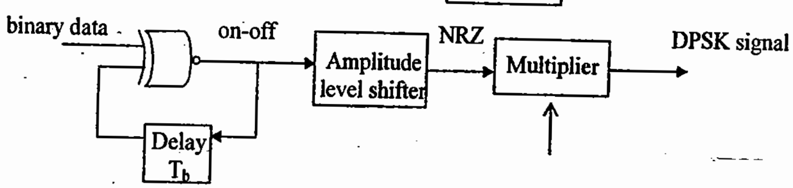
$= (A_c^2/2)\ [\cos 2\pi (f_1 - f_2)\, t + \cos 2\pi (f_1 - f_2)\, t\,]$

$0 \Rightarrow A_c \cos 2\pi f_2 t \times A_c \cos 2\pi f_1 t$ 1

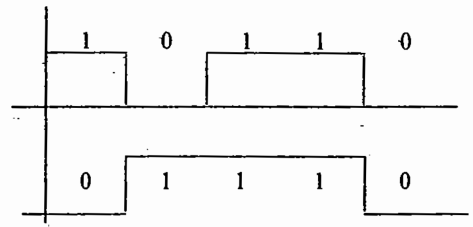
$A_c \cos (2\pi f_2 t) \times A_c \cos (2\pi f_2 t)$ 2

Differential Phase Shift Keying (DPSK) :

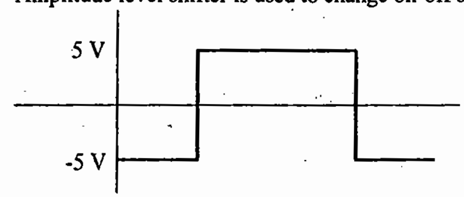




	1	0	1	1	0
0	0	1	1	1	0

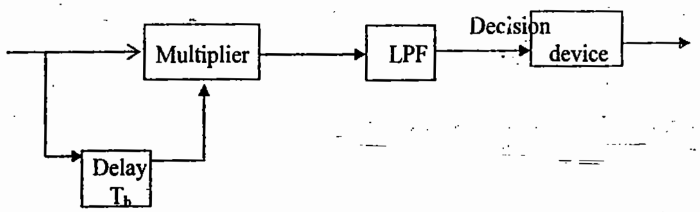


Amplitude level shifter is used to change on-off signalling to NRZ.



By adding a negative dc level, we can convert an on-off signalling to NRZ.

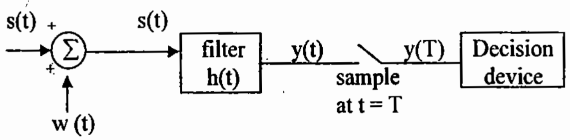
	1	0	1	1	0
0	0	1	1	1	0
180^0	180^0	0^0	0^0	0^0	180^0



Receiver

Advantage of DPSK over PSK is DPSK does not require a coherent carrier for demodulation.

(C) Matched filters / Correlation Receiver, B.W, Probability of error



Input to the filter $\rightarrow$ Signal + White noise

$\rightarrow s(t) + w(t)$

Output of the filter: signal component

$= s(t) \circledast h(t)$

$\rightarrow S(f)\ H(f)$

$$y(t) = \int_{-\infty}^{\infty} S(f)\, H(f)\, e^{j2\pi fT}\, df$$

After sampling

$$y(T) = \int_{-\infty}^{\infty} S(f)\, H(f)\, e^{j2\pi fT}\, df$$

Signal Power

$$|y(T)|^2 = \left| \int_{-\infty}^{\infty} S(f)\, H(f)\, e^{j2\pi ft}\, df \right|^2$$

Schwartz inequality :

$$\left| \int_{-\infty}^{\infty} g_1(t)\, g_2(t)\, dt \right|^2 \leq \int_{-\infty}^{\infty} |g_1(t)|^2\, dt \int_{-\infty}^{\infty} |g_2(t)|^2\, dt$$

$$\text{Signal Power} \leq \int_{-\infty}^{\infty} |H(f)|^2\, df \int_{-\infty}^{\infty} |S(f)|^2\, df$$

$$\text{Max. signal Power} \leq \int_{-\infty}^{\infty} |H(f)|^2\, df \int_{-\infty}^{\infty} |S(f)|^2\, df$$

$$H(f) = S^*(f)\, e^{-j2\pi fT}$$

$$h(t) = \int_{-\infty}^{\infty} S^*(f)\ e^{-j2\pi fT}\ e^{j2\pi ft}\, df$$

Noise PSD at the output of the filter is $\dfrac{N_0}{2}\ |H(f)|^2$

$$\text{Noise Power} = \frac{N_0}{2} \int_{-\infty}^{\infty} |H(f)|^2\, df$$

$$\text{SNR} \leq \frac{2}{N_0} \int_{-\infty}^{\infty} |S(f)|^2\, df$$

$$(\text{SNR})_{\max} = \frac{2}{N_0} (\text{Energy of the signal}) = \frac{2E}{N_0}$$

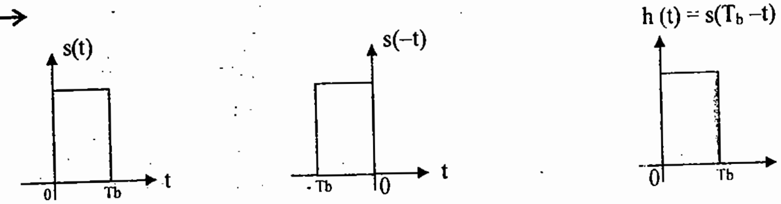
$$h(t) = s^*[-(t - T)] = s^*(T - t)$$

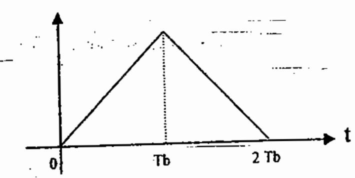
For real signal $\quad s(t) = s^*(t)$

$S^*(f) = S(-f)$

$\therefore\ h(t) = s(T - t)$

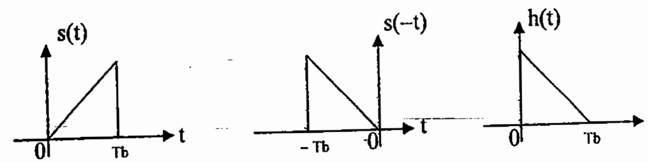
$\rightarrow$

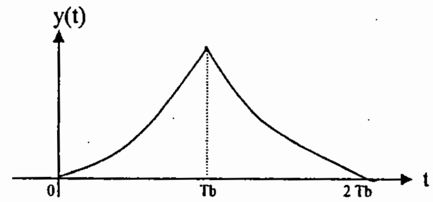




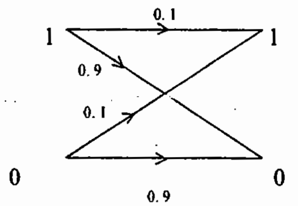
We are sampling the output at T_b only because the signal is max. at T_b

$\rightarrow$



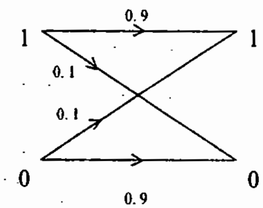


Probability of error :

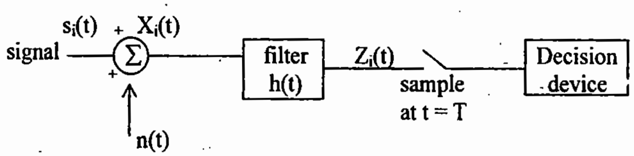


Binary Nonsymmetric channel

$P_e = P(1/0)$ or $P(0/1)$



Binary symmetric channel



$s_i(t) = s_1(t)$ —— '1'

$= s_2(t)$ —— '0'

Input to the filter $= X_i(t) = s_i(t) + n(t)$

Output of the filter $= Z_i(t) = a_i(t) + n_0(t)$

After sampling, $Z_i(T) = a_i(T) + n_0(T)$

$$Z_i = a_i + n_0$$

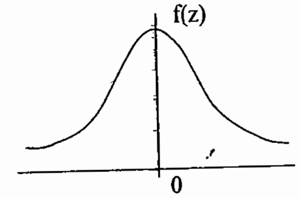
$$Z = a_1 + n_0 \longrightarrow 1$$

$$= a_2 + n_0 \longrightarrow 0$$

Assume the noise as Gaussian noise with mean = 0, when there is no signal.

Z is a Gaussian Random variable with mean = 0

$$f(z) = \frac{1}{\sqrt{2\pi\sigma^2}} \text{Exp}[-z^2/2\sigma^2]$$



$m_1 = 0$

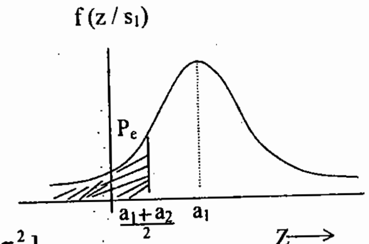
Mean Square Value $= m_2 =$ Noise Power

$\sigma^2 = m_2$ (mean square value)

When the input is '1', $s_1(t)$

$$Z = a_1 + n_0$$

because of n_0, Z is a random variable with mean 'a_1'.

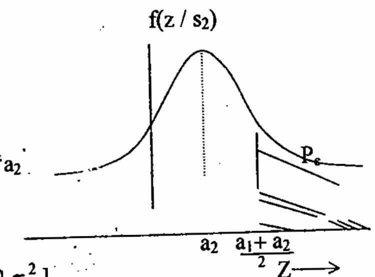


$$f(z/s_1) = \frac{1}{\sqrt{2\pi\sigma^2}} \text{Exp}[-(z-a_1)^2/2\sigma^2]$$

When the input is '0', $s_2(t)$

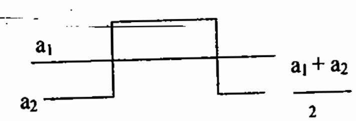
$$Z = a_2 + n_0$$

because of n_0, Z is a random variable with mean 'a_2'.



$$f(z/s_2) = \frac{1}{\sqrt{2\pi\sigma^2}} \text{Exp}[-(z-a_2)^2/2\sigma^2]$$

'0' is decoded as '1' when $(a_2 + n_0) > \dfrac{a_1 + a_2}{2}$



'1' is decoded as '0' when $(a_1 + n_0) < \dfrac{a_1 + a_2}{2}$

$$P[X \le x] = F_X(x) = \int_{-\infty}^{x} f(x)\,dx$$

Probability that 1 is decoded as zero is

$$P_{e(1/0)} = P\left(Z < \frac{a_1 + a_2}{2}\right)$$

$$= \frac{1}{\sqrt{2\pi\sigma^2}} \int_{-\infty}^{(a_1+a_2)/2} \exp\left[-(z-a_1)^2/2\sigma^2\right] dz$$

Probability of error when '0' is transmitted is

$$P_{e(0/1)} = P\left(Z > \frac{a_1+a_2}{2}\right)$$

$$= 1 - \frac{1}{\sqrt{2\pi\sigma^2}} \int_{-\infty}^{(a_1+a_2)/2} \exp\left[-(z-a_2)^2/2\sigma^2\right] dz$$

For binary symmetrical channel $P_{e(1/0)} = P_{e(0/1)}$

$$= \int_{(a_1+a_2)/2}^{\infty} \frac{1}{\sqrt{2\pi\sigma^2}} \exp\left[-(z-a_2)^2/2\sigma^2\right] dz$$

Let $y = \frac{z-a_2}{\sigma}$

$z = \frac{a_1+a_2}{2}$ $\qquad$ $y = \frac{a_1-a_2}{2\sigma}$

$$= \int_{(a_1-a_2)/2\sigma}^{\infty} \frac{1}{\sqrt{2\pi\sigma^2}} \exp\left[-y^2/2\right] \sigma\, dy$$

$$= \frac{1}{\sqrt{2\pi}} \int_{(a_1-a_2)/2\sigma}^{\infty} \exp\left[-y^2/2\right] dy$$

$$P_e = Q\left(\frac{a_1-a_2}{2\sigma}\right)$$

where $Q(x) = \frac{1}{\sqrt{2\pi}} \int_{x}^{\infty} \exp\left[-y^2/2\right] dy$

$$P_e = Q\left(\sqrt{\frac{(a_1-a_2)^2}{4\sigma^2}}\right)$$

$(a_1-a_2)^2$ = Difference signal power, $\frac{(a_1-a_2)^2}{\sigma^2} = \frac{2E_d}{N_0}$

$$P_e = Q\left(\sqrt{\frac{2E_d}{4N_0}}\right) = Q\left(\sqrt{\frac{E_d}{2N_0}}\right)$$

On-off signaling :

$$E_d = \int_{-\infty}^{\infty} (s_1-s_2)^2\, dt$$

Difference signal energy per bit $= \int_0^{T_b} (s_1-s_2)^2\, dt = A^2 T_b$

$$P_e = Q\left(\sqrt{\frac{A^2 T_b}{2N_0}}\right)$$

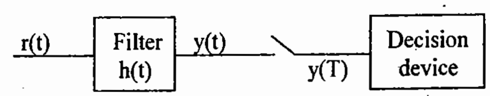
NRZ signaling :

$$\int_0^{T_b} 4A^2\, dt = 4A^2 T_b$$

$$P_e = Q\left(\sqrt{\frac{2A^2 T_b}{N_0}}\right)$$

$$P_{e,\,NRZ} < P_{e,\,on-off}$$

Correlation Receiver :



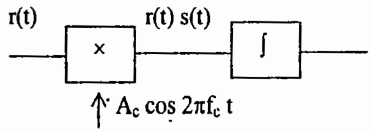
$$y(t) = \int_{-\infty}^{\infty} r(\tau)\, h(t-\tau)\, d\tau$$

$$= \int_{-\infty}^{\infty} r(\tau)\, s[T-(t-\tau)]\, d\tau \qquad h(t) = s(T-t)$$

$$= \int_{-\infty}^{\infty} r(\tau)\, s[T+\tau-t]\, d\tau$$

After sampling ,

$$y(T) = \int_0^{T_b} r(\tau)\, s(\tau)\, d\tau$$



In general , correlation receiver and matched filter are same.

$$\therefore\ P_e = Q\left(\sqrt{\frac{E_d}{2 N_0}}\right)$$

ASK signal :

$$s_1 = A_c \cos 2\pi f_c t$$

$$s_2 = 0$$

$$E_d = A_c^2 \int_0^{T_b} \cos^2 2\pi f_c t$$

$$= A_c^2 \int_0^{T_b} (1 + \cos 4\pi f_c t)/2 \ = A_c^2 [T_b/2] = (A_c^2 T_b)/2$$

$$P_e = Q\left(\sqrt{\frac{A^2 T_b}{4 N_0}}\right)$$

Signal energy when '1' is transmitted $= A_c^2 \int_0^{T_b} \cos^2 2\pi f_c t \ dt = \dfrac{A_c^2 T_b}{2}$

Signal energy when '0' is transmitted $= 0$

$\therefore$ Average energy per bit $= \dfrac{A_c^2 T_b}{4} = E_b$

$$A_c = \sqrt{\frac{4 E_b}{T_b}}$$

$$s(t) = \sqrt{\frac{4 E_b}{T_b}} \cos 2\pi f_c t \qquad \text{'1'}$$

$$= 0 \qquad \text{'0'}$$

$$\therefore\ P_e = Q\left(\sqrt{\frac{E_b}{N_0}}\right)$$

PSK signal :

$$S_1 = A_c \cos 2\pi f_c t$$

$$S_2 = -A_c \cos 2\pi f_c t$$

$$E_d = 4 A_c^2 \int_0^{T_b} \cos^2 2\pi f_c t = \frac{4 A_c^2 T_b}{2} = 2 A_c^2 T_b$$

$$P_e = Q\left(\sqrt{\frac{2 A^2 T_b}{2 N_0}}\right) = Q\left(\sqrt{\frac{A_c^2 T_b}{N_0}}\right)$$

$$P_{e,\,PSK} < P_{e,\,ASK}$$

So generally PSK is preferred.

$$\therefore\ P_e = Q\left(\sqrt{\frac{2 E_b}{N_0}}\right)$$

$$s_1(t) = A_c \cos 2\pi f_c t \qquad \text{'1'}$$

$$s_1(t) = -A_c \cos 2\pi f_c t \qquad \text{'0'}$$

Signal energy when '1' is transmitted $= \dfrac{A_c^2 T_b}{2}$

Signal energy when '0' is transmitted $= \dfrac{A_c^2 T_b}{2}$

Average energy per bit $E_b = \dfrac{A_c^2 T_b}{2}$

$$A_c = \sqrt{\frac{2 E_b}{T_b}}$$

$$S_1(t) = \sqrt{\frac{2E_b}{T_b}}\cos 2\pi f_c t \qquad \text{'1'}$$

$$= -\sqrt{\frac{2E_b}{T_b}}\cos 2\pi f_c t \qquad \text{'0'}$$

FSK signal :

$$S_1(t) = A_c \cos 2\pi f_1 t$$

$$S_2(t) = A_c \cos 2\pi f_2 t$$

$$E_d = A_c^2 \int_0^{T_b} (\cos 2\pi f_1 t - \cos 2\pi f_2 t)^2\, dt = A_c^2 T_b$$

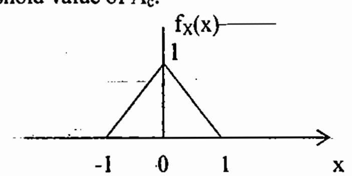
$$P_e = Q\left(\sqrt{\frac{A^2 T_b}{2N_0}}\right)$$

PSK is most preferred compared to ASK and FSK because it has minimum bandwidth and P_e.

Example : Consider an AM system with additive thermal noise having a power spectral density $\eta/2 = 10^{-12}$ W/Hz. Assume that the baseband message signal X(t) has a bandwidth of 4 kHz an the amplitude distribution shown figure. The signal is demodulated by envelope detection and appropriate postdetection filtering. Assume $\mu = 1$.

(a) Find the minimum value of the carrier amplitude A_c that will yield $(S/N_o) \geq 40$ dB.

(b) Find the threshold value of A_c.



Sol : (a) $X_c(t) = A_c[1 + X(t)]\cos\omega_c t$

$$S_X = E[X^2(t)] = \int_{-\infty}^{\infty} x^2 f_X(x)\,dx = 2\int_0^1 x^2(-x+1)\,dx = 1/6$$

$$(S/N)_0 = \frac{S_X}{1+S_X}\gamma = \frac{1/6}{1+1/6}\gamma = (1/7)\gamma \geq 10^4$$

Hence, $\gamma \geq 7(10^4)$

$$\therefore \frac{A_c^2(1+1/6)}{4(10^{-12})(4)(10^3)} \geq 7(10^4)$$

$$A_c \geq 31(10^{-3})\,V = 31\text{ mV}$$

(b) $(S/N) = 10$

Therefore, $\gamma|_{\text{at threshold}} = \gamma_{th} = 2(S/N) = 20$

$$\therefore \frac{A_c^2(1+1/6)}{4(10^{-12})(4)(10^3)} = 20 \Rightarrow A_c = 0.52(10^{-3})\,V = 0.52\text{ mV}$$

Chapter – 5A Objective Questions Set - A

01. Which of the following gives maximum probability of error?
 a) ASK b) FSK
 c) PSK d) DPSK

02. Which of the following gives minimum probability of error ?
 a) ASK b) FSK
 c) PSK d) DPSK

State True or False

03. In DPSK, no synchronous carrier is needed at the receiver
 (a) True (b) False

04. In FSK, no synchronous carrier is needed at the receiver
 (a) True (b) False

05. Probability of error in DPSK is less than PSK
 (a) True (b) False

06. Amplitude Shift Keying is also known as on-off keying
 (a) True (b) False

07. Correlator is a coherent system of signal reception
 (a) True (b) False

08. In differential PSK, the information is coded in terms of
 a) Absolute phase for each symbol
 b) Phase changes between adjacent symbols
 c) None of these

09. Main disadvantage of DPSK is that
 a) Power margin is very low
 b) It requires much large power
 c) It is locked on a specific signaling speed.

10. In FSK, the threshold level is independent of
 a) Carrier amplitude
 b) Carrier frequency
 c) Both of these

11. In ASK, the threshold level is
   a) Independent of carrier amplitude A_c
   b) Function of A_c
   c) None

12. The QAM is a combination of the
   a) ASK and PSK
   b) ASK and FSK
   c) PSK and FSK

13. The disadvantage of ~~coherent FSK~~ detection is that
   a) It leads to high signal fading
   b) It requires two synchronized oscillations
   c) Both of these
   d) None

14. FSK is preferred to ASK in applications where
   a) Fading of signal is prevalent
   b) Synchronous detection is not feasible
   c) Both of these
   d) None

15. The disadvantage of FSK is that
   a) It does not provide sufficient S/N ratio
   b) It does not have low error probability
   c) It is not efficient in its use of spectrum space
   d) None

16. In contrast to analog transmission, digital system can employ
   a) Regeneration
   b) Error control coding
   a) Both of these
   d) None

17. In coherent detection in digital transmission, the received waveform is
   a) Compared with a reference waveform
   b) Compared with the possible transmitted waveform
   c) None of these

18. In ASK, the transmission bandwidth is equal to
   a) Baseband bandwidth
   b) Twice baseband bandwidth
   c) Four times baseband bandwidth
   d) None

19. The total bandwidth required for a raised cosine spectrum is
   a) W / 2 b) W
   c) 2W d) 4W

20. In μ-ary PSK channel, the information rate (binary bits/sec) can easily be increased by
   a) Increasing μ
   b) Decreasing μ
   c) None

Key :

01. a	02. c	03. a	04. b	05. b	06. a
07. a	08. b	09. c	10. a	11. b	12. a
13. b	14. c	15. c	16. c	17. b	18. b
19. c	20. a				

Chapter – 5 B & C Objective Questions Set - B

01. Consider a Binary Symmetric Channel (BSC) with probability of error being p. To transmit a bit, say 1, we transmit a sequence of three 1s. The receiver will interpret the received sequence to represent 1 if at least two bits are 1. The probability that the transmitted bit will be received in error is

(a) $p^3 + 3p^2(1 - p)$ (b) p^3

(c) $(1 - p)^3$ (d) $p3+p^2(1-p)$

02. The raised cosine pulse p(t) is used for zero ISI in digital communications. The expression for p(t) with unity roll-off factor is given by

$$p(t) = \frac{\sin 4\pi Wt}{4\pi Wt(1 - 16W^2t^2)}$$

The value of p(t) at t = 1 / 4W is

(a) -0.5 (b) 0

(c) 0.5 (d) ∞

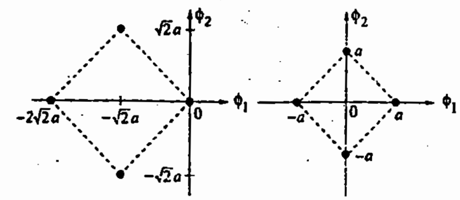
03. During transmission over a certain binary communication channel, bit errors occur independently with probability p. The probability of AT MOST one bit in error in a block of n bits is given by

(a) p^n (b) $1-p^n$

(c) $np(1-p)^{n-1}+(1-p)^n$ (d) $1-(1-p)^n$

Common Data for Questions 04, 05:

Two 4 – ary signal constellations are shown. It is given that ϕ_1 and ϕ_2 constitute an orthonormal basis for the two constellations. Assume that the four symbols in both the constellations are equiprobable. Let $N_0 / 2$ denote the power spectral density of white Gaussian noise.



Constellation 1 Constellation 2

following statements is true?

(a) Probability of symbol error for Constellation 1 is lower

(b) Probability of symbol error for Constellation 1 is higher

(c) Probability of symbol error is equal for both the constellations

(d) The value of N_0 will determine which of the two constellations has a lower probability of symbol error

Common Data for Questions 06,07:

Let $g(t) = p(t) * p(t)$, where * denotes convolution and $p(t) = u(t) - u(t-1)$ with $u(t)$ being the unit step function

Q06. The impulse response of filter matched to the signal $s(t) = g(t) - \delta(t-2) * g(t)$ is given as

(a) $s(1-t)$ (b) $-s(1-t)$

(c) $-s(t)$ (d) $s(t)$

Q07. An Amplitude Modulated signal is given as

$x_{AM}(t) = 100\,(p(t) + 0.5\,g(t))\cos\omega_c t$ in the interval $0 \le t \le 1$.

One set of possible values of the modulating signal and modulation index would be

(a) t, 0.5 (b) t, 1.0

(c) t, 2.0 (d) t^2, 0.5

Q08. A signal as shown in fig. is applied to a matched filter. Which of the following does represent the output of this matched filter?

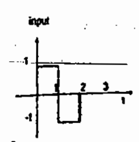
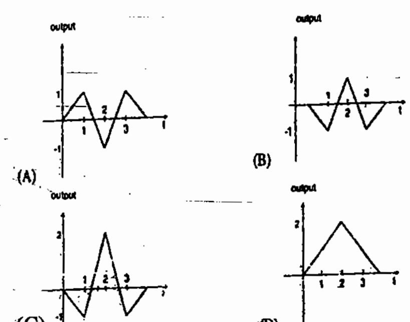


Fig.



09. Consider a binary digital communication system with equally likely 0's and 1's. When binary 0 is transmitted the voltage at the detector input can lie between the levels – 0.25 V and + 0.25 V with equal probability; when binary 1 is transmitted, the voltage at the detector can have any value between 0 and 1 V with equal probability. If the detector has a threshold of 0.2 V (i.e., if the received signal is greater than 0.2 V, the bit is taken as 1), the average bit error probability is

a) 0.15 b) 0.2

c) 0.05 d) 0.5

10. Choose the correct one from among the alternatives A, B, C, and D after matching an item from Group 1 with the most appropriate item in Group 2.

Group 1	Group 2
1. FM	P : Slope overload
2. DM	Q : μ - law
3. PSK	R : Envelope detector
4. PCM	S : Capture effect
	T : Hilbert transform
	U : Matched filter

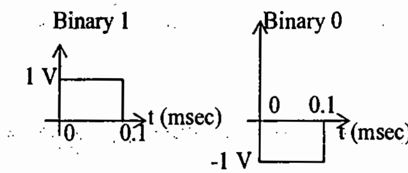
(a) 1 – T, 2 – P, 3 – U, 4 – S

(b) 1 – S, 2 – U, 3 – P, 4 – T

(c) 1 – S, 2 – P, 3 – U, 4 – Q

(d) 1 – U, 2 – R, 3 – S, 4 – Q

11. A source produces binary data at the rate of 10 kbps. The binary symbols are represented as shown in fig.



The source output is transmitted using two modulation schemes, namely Binary PSK (BPSK) and Quadrature PSK (QPSK). Let B_1 and B_2 be the bandwidth requirements of BPSK and QPSK respectively. Assuming that the bandwidth of the above rectangular pulses is 10 kHz, B_1 and B_2 are

a) $B_1 = 20$ kHz, $B_2 = 20$ kHz
b) $B_1 = 10$ kHz, $B_2 = 20$ kHz
c) $B_1 = 20$ kHz, $B_2 = 10$ kHz
d) $B_1 = 10$ kHz, $B_2 = 10$kHz

12. If E_b, the energy per bit of a binary digital signal, is 10^{-6} watt-sec and the one-sided power spectral density of the white noise, $N_o = 10^{-5}$ W/Hz, then the output SNR of the matched filter is

a) 26 dB b) 10 dB
c) 20 dB d) 13 dB

13. If S represents the carrier synchronization at the receiver and ρ represents the bandwidth efficiency, then the correct statement for the coherent binary PSK is

a) ρ = 0.5, S is required
b) ρ = 1.0, S is required
c) ρ = 0.5, S is not required
d) ρ = 1.0, S is not required

14. At a given probability of error, binary coherent FSK is inferior to binary coherent PSK by

(a) 6 dB (b) 3 dB
(c) 2 dB (d) 0 dB

15. For a bit-rate of 8 Kbps, the best possible values of the transmitted frequencies in a coherent binary FSK system are

a) 16 KHz and 20 KHz
b) 20 KHz and 32 KHz
c) 20 KHz and 40 KHz
d) 32 KHz and 40 KHz

16. During transmission over a communication channel, bit errors occur independently with probability 'p'. If a block of n bits is transmitted, the probability of at most one bit error is equal to

a) $1 - (1-p)^n$

b) $p + (n-1)(1-p)$

c) $np(1-p)^{n-1}$

d) $(1-p)^n + np(1-p)^{n-1}$

17. In a digital communication system employing Frequency Shift Keying (FSK), the 0 and 1 bit are represented by sine waves of 10 KHz and 25 KHz respectively. These waveforms will be orthogonal for a bit interval of

a) 45 μsec b) 200 μsec
c) 50 μsec d) 250 μsec

18. The input to a matched filter is given by

$$s(t) = \begin{cases} 10\sin(2\pi \times 10^6 t), & 0 < t < 10^{-4}\text{ sec} \\ 0, & \text{otherwise} \end{cases}$$

The peak amplitude of the filter output is

a) 10 volts b) 5 volts
c) 10 millivolts d) 5 millivolts

19. For a given date rate, the bandwidth B_p of a BPSK signal and the bandwidth B_0 of the OOK signal are related as
   a) $B_p = B_o / 4$ b) $B_p = B_o / 2$
   c) $B_p = B_o$ d) $B_p = 2 B_o$

20. The line code that has zero dc component for pulse transmission of random binary data is
   a) Non-return to zero (NRZ)
   b) Return to zero (RZ)
   c) Alternate Mark Inversion (AMI)
   d) None of the above

21. A PAM signal can be detected by using
   a) an ADC b) an integrator
   c) a band pass filter
   d) a high pass filter

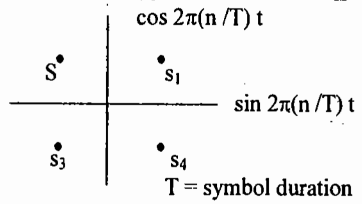
22. Match the following
   a) SSB 1) Envelope detector
   b) AM 2) Integrate and dump
   c) BPSK 3) Hilbert transform
   4) Ratio detector
   5) PLL

23. The bit stream 01001 is differentially encoded using 'Delay and EX OR' scheme for DPSK transmission. Assuming the reference bit as a '1' and assigning phases of '0' and 'π' for 1's and 0's, respectively, in the encoded sequence, the transmitted phase sequence becomes
   a) π 0 π π 0 b) 0 π π 0 0
   c) 0 π π π 0 d) π π 0 π π

24. Coherent demodulation of FSK signal can be effected using
   a) correlation receiver
   b) bandpass filters and envelope detectors
   c) matched filter
   d) discriminator detection.

25. Source encoding in a data communication system is done in order to
   a) enhance the information transmission rate
   b) reduce the transmission errors
   c) conserve the transmitted power
   d) facilitate clock recovery in the receiver.

26. For the signal constellation shown in the figure, the type of modulation is____



27. In binary data transmission DPSK is preferred to PSK because:
   (a) a coherent carrier is not required to be generated at the receiver
   (b) for a given energy per bit, the probability of error is less.
   (c) the 180° phase shifts of the carrier are unimportant
   (d) more protection is provided against impulse noise

28. The message bit sequence input to a DPSK modulator is 1, 1, 0, 0, 1, 1. The carrier phase during the reception of the first two message bits is π, π. The carrier phase for the remaining four message bits is
   (a) π, π, 0, π (b) 0, 0, π, π
   (c) 0, π, π, π (d) π, π, 0, 0

29. In a digital communication system, transmissions of successive bits through a noisy channel are assumed to be independent events with error probability p. The probability of at most one error in the transmission of an 8 – bit sequence is
   (a) $7(1-p)/8 + p/8$
   (b) $(1-p)^8 + 8p(1-p)^7$.
   (c) $(1-p)^8 + (1-p)^7$.
   (d) $(1-p)^8 + p(1-p)^7$

Objective Questions

01. Four messages band limited to W, W, 2W and 3W respectively are to be multiplexed using Time Division Multiplexing (TDM). The minimum bandwidth required for transmission of this TDM signal is.
   (a) W (b) 3W (c) 6W (d) 7W

02. In a GSM system, 8 channels can co-exist in 200 kHz bandwidth using TDMA. A GSM based cellular operator is allocated 5 MHz bandwidth. Assuming a frequency reuse factor of $\frac{1}{5}$, i.e. a five-cell repeat pattern, the maximum number of simultaneous channels that can exist in one cell is
   (a) 200 (b) 40 (c) 25 (d) 5

03. In a Direct Sequence CDMA system the chip rate is 1.2288×10^6 chips per second. If the processing gain is desired to be AT LEAST 100, the data rate
   (a) must be less than or equal to 12.288×10^3 bits per sec
   (b) must be greater than 12.288×10^3 bits per sec
   (c) must be exactly equal to 12.288×10^3 bits per sec
   (d) can take any value less than 122.88×10^3 bits per sec

04. Three analog signals, having bandwidths 1200 Hz, 600 Hz and 600 Hz, are sampled at their respective Nyquist rates, encoded with 12 bit words, and time division multiplexed. The bit rate for the multiplexed signal is
   a) 115.2 kbps b) 28.8 kbps
   c) 57.6 kbps d) 38.4 kbps

05. Four independent messages have bandwidths of 100 Hz, 100 Hz, 200 Hz, and 400 Hz, respectively. Each is sampled at the Nyquist rate, and the samples are time division multiplexed (TDM) and transmitted. The transmitted sample rate (in Hz) is
   a)1600 b)800 c)400 d)200

06. Quadrature multiplexing is
   a) the same as FDM
   b) the same as TDM
   c) a combination of FDM and TDM
   d) quite different from FDM and TDM

IES

07. 12 signals each band – limited to 5 kHz are to be transmitted over a single channel by frequency division multiplexing. If AM – SSB modulation with guard band of 1 kHz is used, then the bandwidth of the multiplexed signal will be

(a) 51 kHz (b) 61 kHz

(c) 71 kHz (d) 81 kHz

08. Time division multiplexing requires
(a) constant data transmission
(b) transmission of data samples
(c) transmission of data at random
(d) transmission of data of only one measured

09. Quadrature multiplexing is
(a) same as FDM
(b) same as TDM
(c) a combination of FDM and TDM
(d) the scheme where same carrier frequency is used for two different signals

10. Four signals each band – limited to 5 kHz are sampled at twice the Nyquist rate. The resulting PAM samples are transmitted over a single channel after time division multiplexing. The theoretical minimum transmission bandwidth of the channel should be equal to

(a) 5 kHz (b) 20 kHz

(c) 40 kHz (d) 80 kHz

11. In a certain '12 channel TDM'. system, it is found that channel No. 3 and channel No. 8 are connected to the same input signal. The technique

(a) wastes the channel capacity
(b) takes care of different sampling rates
(c) is required when different bandwidth signals are to be transmitted
(d) reduces noise

12. In asynchronous TDM, for n signal sources, each frame contains 'm' slots, where m is usually
(a) less than n (b) 2 n
(c) n (d) greater than 2 n

13. In a cellular communications system, path loss between transmitter and receiver is due to
(a) scattering from buildings, trees, vehicles and other structures only
(b) reason at (a) above and due to reflections from ground only
(c) reasons at (a) and (b) above along with reflection from ionosphere only
(d) reasons at (a), (b) and (c) above along with loss due to surface wave phenomenon

14. The number of signalling bits per channel per frame in T1 multiplexer following CCITT hierarchy is
(a) 64000 (b) 128
(c) 4 (d) 400

15 Four signals g_1 (t), g_2 (t), g_3 (t) and g_4 (t) are to be multiplexed and transmitted. g_1 (t) and g_4 (t) have a bandwidth of 4 kHz, and the remaining two signals have bandwidth of 8 kHz. Each sample requires 8 bit for encoding. What is the minimum transmission bit rate of the system?

(a) 512 kbps (b) 16 kbps

(c) 192 kbps (d) 384 kbps

16. A 12 channel TDM system where each channel signal is 4 kHz is sampled at 8 kHz. What is the bandwidth requirement?
(a) 12 kHz (b) 12×4 kHz

(c) 12×8 kHz (d) $12 \times 8 \times 4$ kHz

17. Assertion (A): TDM can be employed to transmit channels having unequal bandwidths.
Reason (R): If sampling theorem is strictly followed, any analog signal can be reconstructed from its samples.

(a) Both A and R are true and R is the correct explanation of A
(b) Both A and R are true but R is NOT the correct explanation of A
(c) A is true but R is false
(d) A is false but R is true

18. In FDM systems used for telephone, which modulation scheme is adopted?
(a) AM (b) DSB – SC
(c) SSB – SC (d) FM

19. Which is the most important sub – system for recovering and reconstructing signals in a TDM system?
(a) Envelope detector followed by a low pass filter
(b) Synchronization circuit for proper timing
(c) Band pass filters to segregate channels
(d) Coherent detector to ensure frequency and phase connection

20. Multiplexing is possible if signals are sampled. Two signals have bandwidths A = 0 to 4 kHz and B = 0 to 8 kHz respectively. The sampling frequency chosen is 12 kHz. Which one of the following is correct? This choice of the sampling frequency

(a) is correct since A and B have an integral relationship of 2
(b) will not lead to aliasing
(c) does not obey sampling theorem
(d) can never lead to multiplexing

21. Which one of the following is correct? In a TDM system each signal is allotted in a frame a unique and fixed

(a) frequency slot (b) time slot

(c) amplitude slot (d) phase slot

"All the best"