CBCS SCHEME

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BEC402

Fourth Semester B.E./B.Tech. Degree Examination, June/July 2024
Principles of Communication Systems

Time: 3 hrs. Max. Marks: 100

Note: 1. Answer any FIVE full questions, choosing ONE full question from each module.
2. M : Marks , L: Bloom's level , C: Course outcomes.

		Module – 1	M	L	C
Q.1	a.	What is conditional probability? Prove that $P\left(\frac{B}{A}\right) = P\left(\frac{A}{B}\right) \cdot P(B)/P(A)$	05	L2	CO1
	b.	Define the autocorrelation and cross correlation. Discuss the properties of autocorrelation.	10	L2	CO1
	c.	Develop a program to generate the probability density function of Gaussian distribution function.	05	L3	CO1
		OR			
Q.2	a.	Define auto-covariance, random variable, cumulative distribution function and probability distribution function.	08	L1	CO1
	b.	The random variable its plot is given as $f_x(x) = 2.e^{-2x}$ for $x \geq 0$. Find the probability that it will take value between 1 and 3.	04	L3	CO1
	c.	Define probability with an example. Discuss their properties (axioms).	08	L2	CO1
		Module – 2			
Q.3	a.	Explain amplitude modulation with necessary equations and sketches in time domain and frequency domain.	08	L3	CO2
	b.	Define modulation index and percentage of modulation. Explain over modulation and distortion.	06	L2	CO2
	c.	Derive the expression for Amplitude Modulation (AM) power in terms of modulation index.	06	L2	CO1
		OR			
Q.4	a.	Explain a general block diagram of a frequency division multiplexing.	06	L1	CO2
	b.	Explain the working principle of lattice type balanced modulator with circuit diagram.	07	L1	CO2
	c.	With neat diagrams, explain high level collector modulator.	07	L2	CO2
		Module – 3			
Q.5	a.	With a neat block diagram, explain converting a phase modulated signal into a frequency modulated signal.	07	L1	CO3
	b.	Determine the frequency modulated signal $v_{FM} = V_C \sin(2\pi f_c t + m_f \sin 2\pi f_m t)$ interms of Bessel functions. Write the amplitude of sideband frequencies (J_n) interms of modulation index (m_f).	06	L3	CO3
	c.	Identify the noise suppression of frequency modulated signal.	07	L2	CO3
		OR			
Q.6	a.	What is the maximum bandwidth of an FM signal with a deviation of 30 kHz and a maximum modulating signal of 5 kHz. (i) Using number of sidebands N = 9 (ii) Using Carson's rule	04	L2	CO3
	b.	Define phase locked loop. Explain with neat circuit diagram of FM demodulator using the IC 565.	08	L2	CO3
	c.	With neat block diagram, explain the concept of frequency modulation with an IC voltage controlled oscillator (IC NE566)	08	L2	CO3

		Module – 4			
Q.7	**a.**	Why digitize the analog signals? Explain the different processes used to convert the analog signal to digital signal.	**06**	**L2**	**CO4**
	b.	What is quantization process? Explain the different types of quantization with their important characteristics.	**07**	**L2**	**CO4**
	c.	Explain the concept of Time division multiplexing with a neat block diagram.	**07**	**L2**	**CO4**
		OR			
Q.8	**a.**	Define PCM (Pulse Code Modulation). Explain the basic elements of a PCM system with neat diagrams.	**06**	**L2**	**CO4**
	b.	For the data stream 01101001. Draw the following line code waveforms: (i) Unipolar NRZ (ii) Polar NRZ (iii) Unipolar RZ (iv) Bipolar RZ (v) Manchester code (vi) Differential coding	**09**	**L3**	**CO4**
	c.	State and prove the sampling theorem. Explain with neat sketches and equations.	**05**	**L2**	**CO4**
		Module – 5			
Q.9	**a.**	Develop a code to generate and plot eye diagram.	**06**	**L3**	**CO5**
	b.	Define noise factor and noise figure. Also explain noise in cascade connection.	**06**	**L2**	**CO5**
	c.	Define Inter Symbol Interference (ISI). Outline baseband binary data transmission system with neat block diagram and equations.	**08**	**L1**	**CO5**
		OR			
Q.10	**a.**	Explain bandwidth requirements of TI systems.	**06**	**L1**	**CO5**
	b.	Write short notes on: (i) Signal to noise ratio (ii) External noise (iii) Internal noise	**08**	**L1**	**CO5**
	c.	An RF amplifier has an S/N ratio of 8 at the input and an S/N ratio of 6 at the output. What are the noise factor, noise figure and noise temperature?	**06**	**L3**	**CO5**

* * * * *

Module 1

1 a)

What is conditional probability? Prove that $P(B/A) = P(A/B) \cdot P(B)/P(A)$

Conditional Probability

- **Conditional probability** is **defined** as the likelihood of an event or outcome occurring, based on the occurrence of a previous event or outcome.
- Let **P[B|A]** denote the probability of event B, given that event A has occurred. The probability **P**[B|A] is called the conditional probability of B given A.
- P[B|A] is defined by $\mathbf{P[B|A]} = \frac{P[A \cap B]}{P[A]}$ where $P[A \cap B]$ is the joint probability of A and B.
- The conditional probability occurs only for dependent events.

Joint Probability

- **Joint probability** is a statistical measure that calculates the likelihood of two events occurring together and at the same point in time. **Joint probability** is the **probability** of event Y occurring at the same time that event X occurs.
- The joint probability of two events may be expressed as the product of the conditional probability of one event given the other, and the elementary probability of the other.
- We may write

$$P[A \cap B] = P[B|A]\ P[A]$$

Or

$$P[A \cap B] = P[A|B]\ P[B]$$

Bayes' rule for conditional probability

- Statement: It states that, the conditional probability,

$$P[B|A] = \frac{P[A|B]\ P[B]}{P[A]}$$

- Proof: we know that,

$P[B|A] = \frac{P[A \cap B]}{P[A]} \Rightarrow P[A \cap B] = P[B|A]\ P[A]$........(1)

$P[A \cap B] = P[A|B]\ P[B] \ldots \ldots (2)$

Equating eqns (1) and (2),

$$P[B|A] = \frac{P[A|B]P[B]}{P[A]}$$

1b)

Define the autocorrelation and cross correlation. Discuss the properties of autocorrelation.

Autocorrelation function

- The autocorrelation function of the process X(t) as the expectation of the product of two random variables $X(t_1)$ and $X(t_2)$, obtained by observing X(t) at times t_1 and t_2 respectively.
    - $R_X(t_1, t_2) = E[X(t_1)X(t_2)]$
    - $= \int_{-\infty}^{\infty}\int_{-\infty}^{\infty} x_1 x_2 f_{X(t_1),X(t_2)}(x_1, x_2)dx_1 dx_2$
- Autocorrelation function gives correlation of a signal with itself but delayed in time
- Autocorrelation function, $r_x(k, l)$ is a function of time difference (k-l).
- Properties of auto correlation function are
    - Let $R_x(\tau)$ be the auto correlation function of random variable 'X'
    - The mean-square value is therefore equivalent to the average power of the process.

$$R_X(0) = E[X(t)X(t)]$$

$$R_X(0) = E[X^2(t)]$$

    - The autocorrelation of a real valued process has even symmetry.

$$R_X(\tau) = R_X(-\tau)$$

    - The autocorrelation function is maximum at the origin.

i.e., $R_x(0) \geq R_x(\tau)$ for any value of τ

1c)

Develop a program to generate the probability density function of Gaussian distribution function.

```
%Matlab Program to Simulate Gaussian PDF
clc
clear all
close all

%Gaussian PDF with Mean 0 and Variance 1
mean=0
variance=1
x=-10:0.001:10;
Gaussian_PDF=(1/sqrt(2*pi*variance))*exp(-((x-mean).^2)/(2*variance));

subplot(2,1,1)
plot(x,Gaussian_PDF)
```

```
grid on
xlabel('Values of x')
ylabel('Probability of x')
title('Gaussian PDF with mean 0 and variance 1')

%Gaussian PDF with Mean 4 and Variance 0.5
mean=4
variance=0.5
x=-10:0.001:10;
Gaussian_PDF=(1/sqrt(2*pi*variance))*exp(-((x-mean).^2)/(2*variance));

subplot(2,1,2)
plot(x,Gaussian_PDF)
grid on
xlabel('Values of x')
ylabel('Probability of x')
title('Gaussian PDF with mean 4 and variance 0.5')
```

2a)

Define auto-covariance, random variable, cumulative distribution function and probability distribution function.

Covariance: It gives joint expectation of two random variables X and Y. It is denoted by symbol 'λ_{XY}'

- $\text{cov}(X,Y) = \lambda_{XY} = E[(X - m_x)(Y - m_y)]$ where
  m_x is the mean od the random variable 'X' and $m_y$is the mean of the random variable 'Y'

$$
\begin{aligned}
\text{cov}(X,Y) = \lambda_{XY} &= E[(X - m_x)(Y - m_y)] \\
&= E[XY - Xm_y - Ym_x + m_x m_y] \\
&= E[XY] - m_y E[X] - m_x E[Y] + m_x m_y \\
&= E[XY] - m_y m_x - m_x m_y + m_x m_y \\
&= E[XY] - 2m_x m_y + m_x m_y \\
&= E[XY] - m_x m_y
\end{aligned}
$$

3.3 Random Variables

- Random variable is a real valued function, which can take any value in the sample space and its range is set of real numbers.
- Random variable can be classified as
  - Continuous Random variable(CRV)
  - Discrete Random variable(DRV)

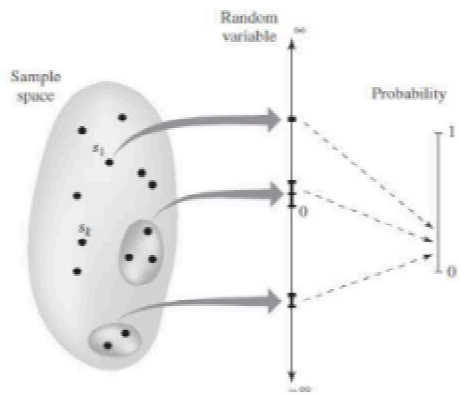


Fig: Illustration of the relationship between sample space, random variable and probability

Probability Mass function

- Let X be a discrete random variable and $[x_1, x_2, x_3, \ldots, x_n]$ be the values of 'X' can take, then $P[X \le x]$ is called probability mass function.
- Probability Mass Function is associated only with Discrete Random Variable
- Eg: Coin tossed twice
- Random variable: Counting no. of heads

- **A function which maps random values to some probabilities is called Probability Mass Function (PMF).**

Probability Distribution Function

- For any random variable 'X' probability distribution function is denoted by $F_X(x)$.
- It is related to PMF as
- $F_X(x) = P[X \le x]$,
- where X is a random variable whose values are $[x_1, x_2, \ldots, x_n]$
- The $F_X(x)$ is a function of 'x' and not a function of random variable 'X'
- Properties of distribution function
  - $F_X(x) \ge 0 \,; -\infty \le x \le \infty$
  - $\mathbf{F_X(-\infty) = 0}$ **and** $\mathbf{F_X(\infty) = 1}$
  - $P[b < X \le a] = F_X(a) - F_X(b); a > b$
- The derivative of probability distribution function is called as **probability density function(pdf)**
- It is denoted by, $f_X(x)$
- Mathematically: $f_X(x) = \frac{dF_X(x)}{dx}$ Where $F_X(x)$is the probability distribution function and X is the random variable that can take any real value 'x'.
- The total area under probability density curve is always equal to 1 (unity) i.e., $\int_{-\infty}^{\infty} f_X(x)dx = 1$

2b)

The random variable its plot is given as $f_x(x) = 2.e^{-2x}$ for $x \geq 0$. Find the probability that it will take value between 1 and 3.

2c)

Define probability with an example. Discuss their properties (axioms).

3.2 Probability

- Probability of event A, is denoted by P(A): This is a function that assigns a non-negative number to an event A in the sample space and satisfies the following properties(axioms):
  - $0 \leq P \leq 1$
  - $P[S] = 1$ (total probability of all events in sample space is 1)
  - If A and B are two mutually exclusive events, then

$$P[A \cup B] = P[A] + P[B] \equiv (P[A \cup B] = P[A \text{ or } B])$$

And

$$P[A \cap B] = 0 \equiv (P[A \text{ and } B] = 0)$$

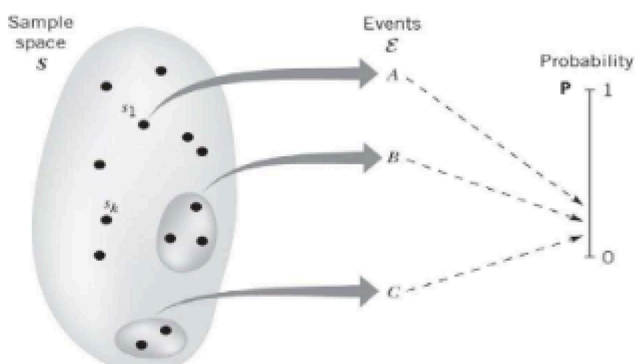


Fig: Illustration of the relationship between sample space, events and probability

- The following properties of probability measure **P** may be derived from the above axioms:
  - $P[\overline{A}] = 1 - P[A]$ When events A and B are not mutually exclusive:
  - $P[A \cup B] = P[A] + P[B] - P[A \cap B]$ Where $P[A \cap B]$ is the probability of joint event "A and B"
  - If $A_1, A_2, A_3, \ldots\ldots A_m$ are mutually exclusive events that include all possible outcomes of the random experiment, then $P[A_1] + P[A_2] + \cdots + P[A_m] = 1$

Module 2:

2a)

Definition: Amplitude modulation is the process of varying the amplitude of a periodic waveform, called the carrier signal, in proportion to the instantaneous amplitude of the modulating signal that typically contains information to be transmitted (1M)

The standard form of AM in the time domain:(4M)

Signals are defined as follows

Message: m(t)

Carrier: $c(t) = Vc\mathrm{Sin}\ (2\pi f_c t)$

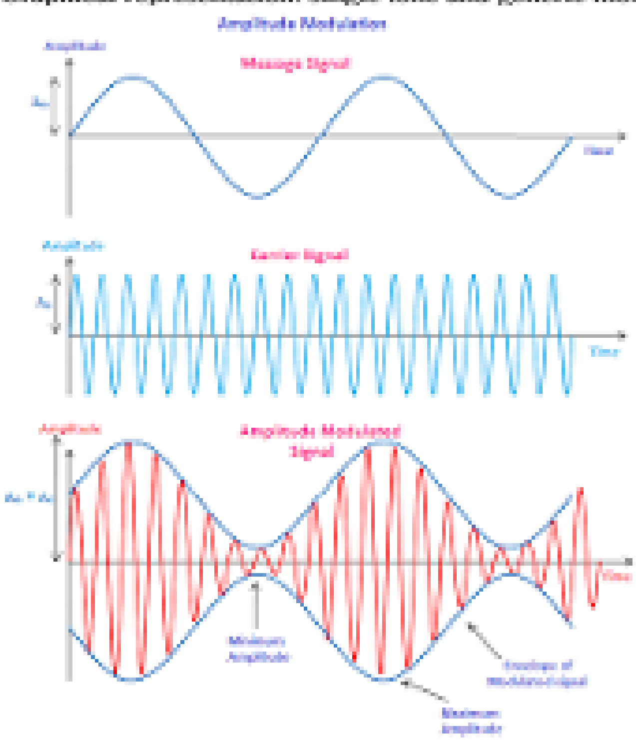
Modulated Signal: The amplitude of the carrier signal changes proportional to m(t) and frequency and phase remain the same, which can be mathematically expressed as

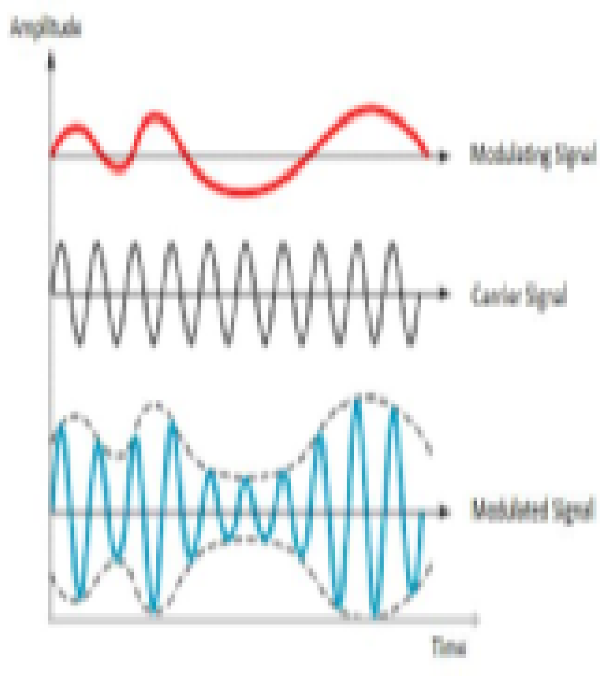
$s(t) = \left[V_c + x * m(t)\right] Sin\ (2\pi f_c t)$; typical value x=1. The expression can be rewritten as

$s(t) = Vc[1 + k_a m(t)\ ]\mathrm{Sin}\ (2\pi f_c t)$

- $k_a\ is\ a\ constant,\ Amplitude\ sensitivity$
- $k_a\ |m(t)|_{max} is\ termed\ as\ Modulation\ index/\ factor,\ \mu$/m
- m=μ = Vm/Vc or $k_a\ |m(t)|_{max}$ or k_a Vm

Graphical representation: single tone and generic modulating signal: (1M)





Frequency Domain Representation: (2M+2M)

To arrive at frequency domain representation, take the Fourier transform of s(t)

$$S(f) = \int_{-\infty}^{\infty} s(t)e^{-j2\pi ft}\ dt$$

$$S(f) = F(V_c\,[1 + k_a m(t)]Sin\,(2\pi f_c t))$$

$$S(f) = F[A_c\,Sin\,(2\pi f_c t)] + F[A_c\,k_a m(t)Sin\,(2\pi f_c t)] \text{---------(1)}$$

Note: magnitude spectrum is considered

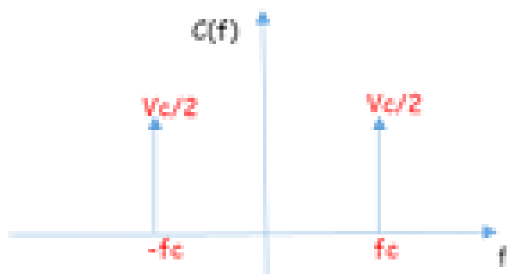
By applying Fourier results

$F\big(Sin(\,2\pi f_c t)\big) = \frac{1}{2}[\delta(f - f_c) + \delta(f + f_c)]$ and $e^{j2\pi ft}g(t) \leftrightarrow G(f - f_c)$ $given$ $g(t) \leftrightarrow G(f)$

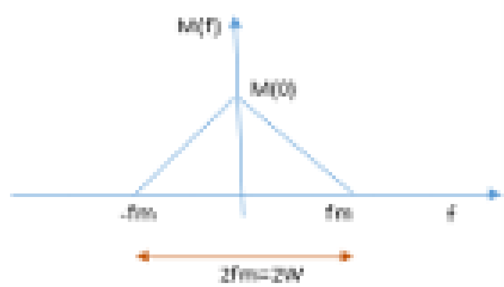
Equation 1 can be rewritten as

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

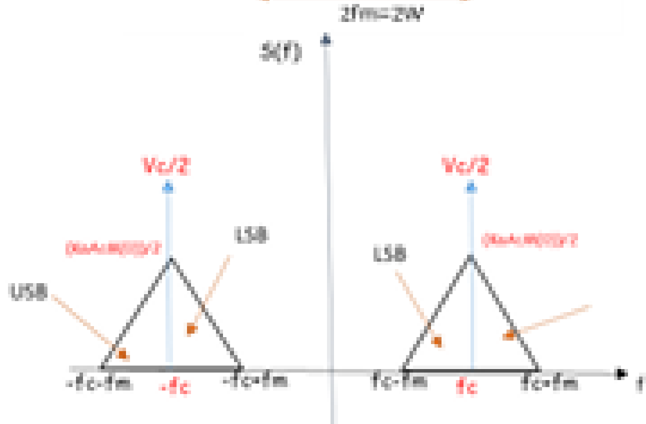
Spectrum of carrier signal



Assuming M(f), the spectrum of the modulating signal to be



Spectrum of S(t)



2b)

Define modulation index and percentage of modulation. Explain over modulation and distortion.

Modulation factor or index is the ratio of change in amplitude of the carrier wave to the amplitude of the unmodulated carrier wave

Time Domain Representation

Time Domain Representation:

Message $\boxed{m(t)}$

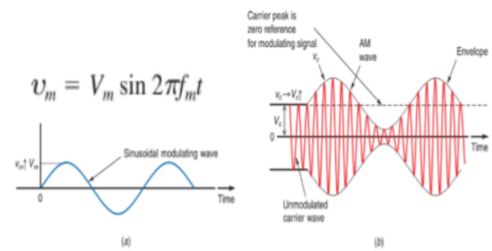
Carrier $\boxed{c(t) = Vc\text{Sin}\,(2\pi f_c t)}$

Modulated Signal

$s(t) = [V_c + x * m(t)\,]Sin\,(2\pi f_c t)$; typical value x=1

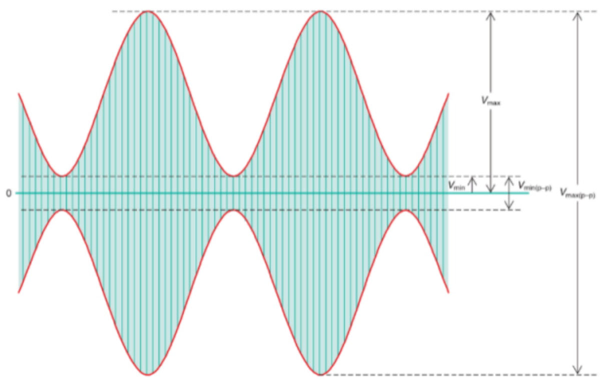
$$\boxed{s(t) = Vc[1 + k_a m(t)\,]\text{Sin}\,(2\pi f_c t)}$$

- k_a *is a constant, Amplitude sensitivity*
- $k_a\,|m(t)|_{max}$ *is termed as Modulation index/ factor,* μ/m
- m=μ = Vm/Vc or $k_a\,|m(t)|_{max}$ or k_a Vm



Modulation factor or index is the ratio of change in amplitude of the carrier wave to the amplitude of the unmodulated carrier wave

Modulation Index-Computation

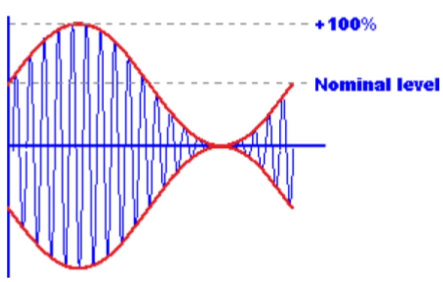


$$V_m = \frac{V_{\max} - V_{\min}}{2}$$

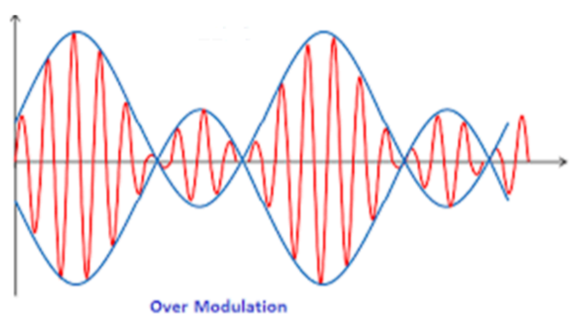
$$V_c = \frac{V_{\max} + V_{\min}}{2}$$

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

|Kam(t)|=m =1



|Kam(t)| or m > 1



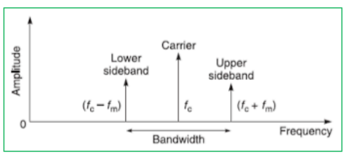
2c)

Derive the expression for Amplitude Modulation (AM) power in terms of modulation index.

Transmitted Power

$$S(f) = F[V_c \, Sin\,(2\pi f_c t)] + F[\frac{Vm}{2}\{Sin(2\pi(f_c + f_m)t) + Sin\,(2\pi(f_c - f_m)t)\}]$$

$$S(f) = \frac{V_c}{2}[\delta(f - f_c) + \delta(f + f_c)] - \frac{V_m}{4}\Big[\big(\delta(f - (f_c + f_m)) + \delta\big(f + (f_c + f_m)\big)\big) + \big(\delta(f - (f_c - f_m)) + \delta\big(f + (f_c - f_m)\big)\big)\Big]$$



$$\boxed{Power = V_{rms}{}^2 R = \frac{V_m{}^2}{2}, R=1}$$

$$\boxed{Power = \frac{V_c{}^2}{2} + \frac{(V_c m)^2}{8} + \frac{(V_c m)^2}{8}}$$

$$P_T = \frac{V_c^2}{2R}\left(1 + \frac{m^2}{4} + \frac{m^2}{4}\right)$$

$$P_T = P_c\left(1 + \frac{m^2}{2}\right)$$

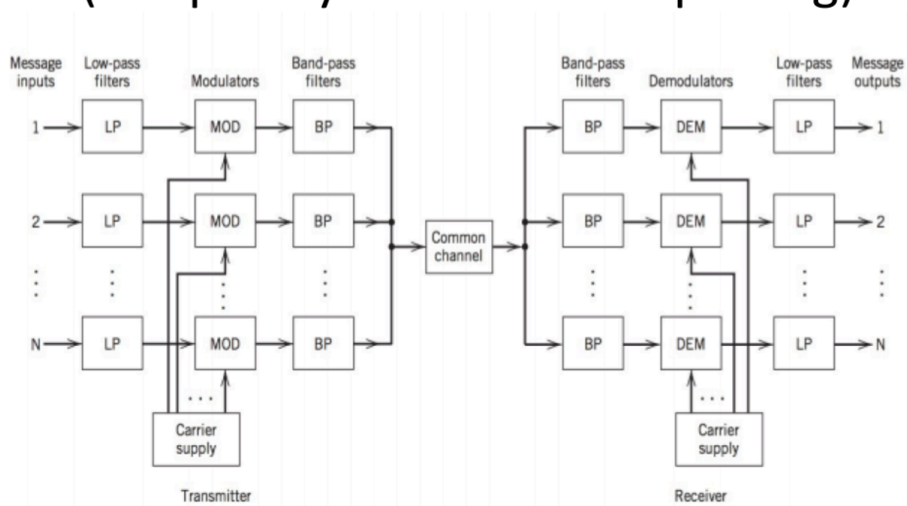
$$P_T = I_T{}^2 R$$

$$I_T = I_c\sqrt{(1 + m^2/2)}. \qquad m = \sqrt{2\left[\left(\frac{I_T}{I_c}\right)^2 - 1\right]}$$

3a)

Explain a general block diagram of a frequency division multiplexing.

FDM(Frequency Division Multiplexing)



The block diagram of FDM-system is shown in figure1.

↳ N-Incoming independent message signals are modulated by mutually Exclusive carriers supplied from carrier source at each modulator. The Modulated signals are passed through the BPF to select any one side band. Therefore BPF's produces SSB-signals and are separated in frequency and combined into a composite signal. and this process is called Frequency division multiplexing.

↳ Multiplexed signal is transmitted over the communication channel.

↳ Total Bandwidth required to N-SSB Modulated signals without any Guard band is

$$BW_T = N \times f_m \text{ ; } N = \text{number of Input signals}$$

↳ At the receiver side N-independent message signals are recovered by passing the composite signal through the BPF followed by Demodulator and LPF.

Advantages of FDM:-

1. A Large Number of signals can be transmitted simultaneously
2. FDM does not requires synchronization between Transmitter & receiver.
3. Demodulation of FDM is easy

3b)

Explain the working principle of lattice type balanced modulator with circuit diagram.

↳ Ring Modulator is a product Modulator used for Generating DSBSC-Modulated signal.

Circuit diagram:

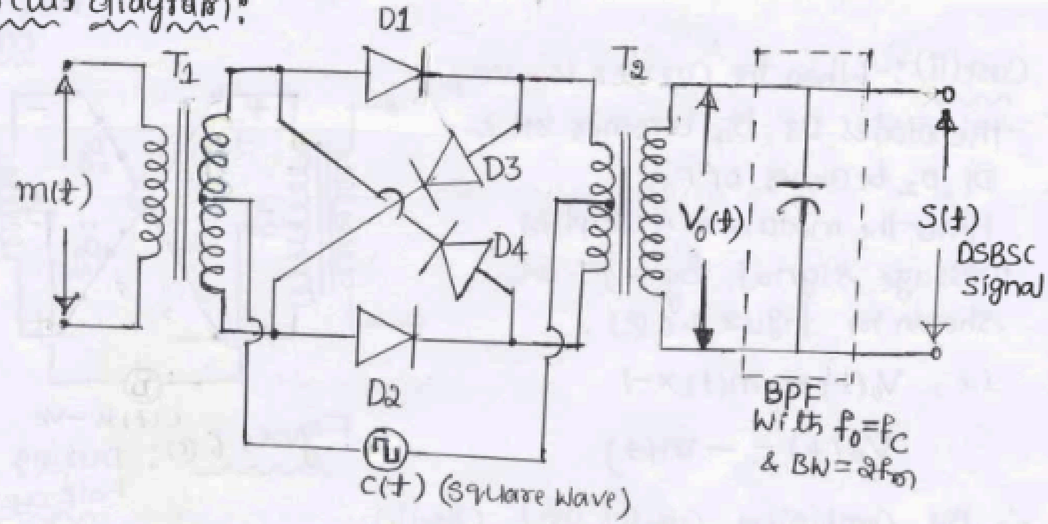


Figure 1.6(a): circuit diagram of Ring Modulator

↳ The circuit diagram of Ring modulator is shown in figure 1.6(a) Consists of two Center-tapped transforms T1, T2 and Four diodes D1, D2, D3 and D4 Connected in bridge circuit and a BPF With Center frequency 'f_c', $BW=2f_m$.

↳ the Carrier signal is applied to the Center taps of the input (T_1) and output (T_2) transformers. Modulating signal is applied to the input transformer T_1.

↳ The output voltage appears across the secondary of the transformer, T_2 (After passing through BPF).

↳ The Diodes Connected in the bridge circuit (Ring) acts like switches and their switching is Controlled by carrier signal (square wave).

Circuit operation:-

Case(i): When the Carrier is +ve, the Diodes D1, D2 becomes ON & Diodes D_3, D_4 becomes OFF.
Hence the Modulator Multiplies message signal $m(t)$ by $+1$.

i.e., $V_0(t) = m(t) \times (+1) = m(t)$

Equivalent circuit is shown in Figure 1.6(b)

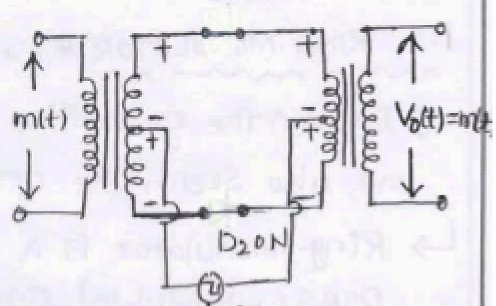


Figure 1.6(b): During +ve half cycle of C(t)

Case(ii) :- When the carrier is -ve, the Diodes D3, D_4 becomes ON & D_1, D_2 becomes OFF. Hence the modulator multiplies message signal by "-1" as shown in figure 1.6 (c).

i.e., $V_0(t) = m(t) \times -1$

$$V_0(t) = -m(t)$$

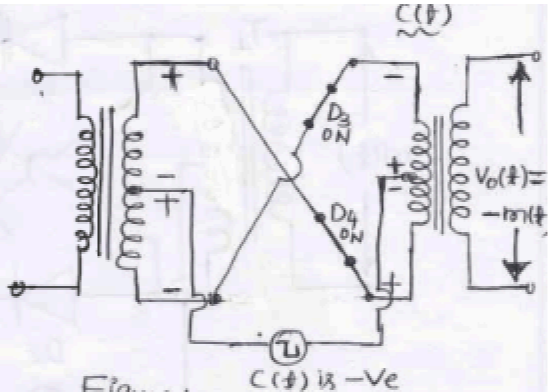


Figure 1.6 (c): During -ve half cycle of C(t)

$\therefore$ By combining Case(i) and Case(ii) The Ring Modulator output at the Secondary of transformer T_2 is given by

$$V_0(t) = m(t) \times C(t) \quad \text{---(1)}$$

The square wave carrier C(t) can be represented by a fourier series as:

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

$$\therefore C(t) = \frac{4}{\pi}\left[\cos 2\pi f_c t - \frac{1}{3}\cos 6\pi f_c t + \cdots\right] \quad \text{---(2)}$$

$\therefore$ Substitute equation (2) in $V_0(t)$ equation (1) we get

$$V_0(t) = m(t) \times \frac{4}{\pi}\left[\cos 2\pi f_c t - \frac{1}{3}\cos 6\pi f_c t + \cdots\right] \quad \text{---(3)}$$

When $V_0(t)$ is passed through BPF having center frequency 'f_c' and Bandwidth '$2f_m$' we get DSBSC signal,

$$\boxed{S(t) = \frac{4}{\pi} m(t) \cos 2\pi f_c t} \Leftarrow \text{DSBSC Wave generated from RING Modulator}$$

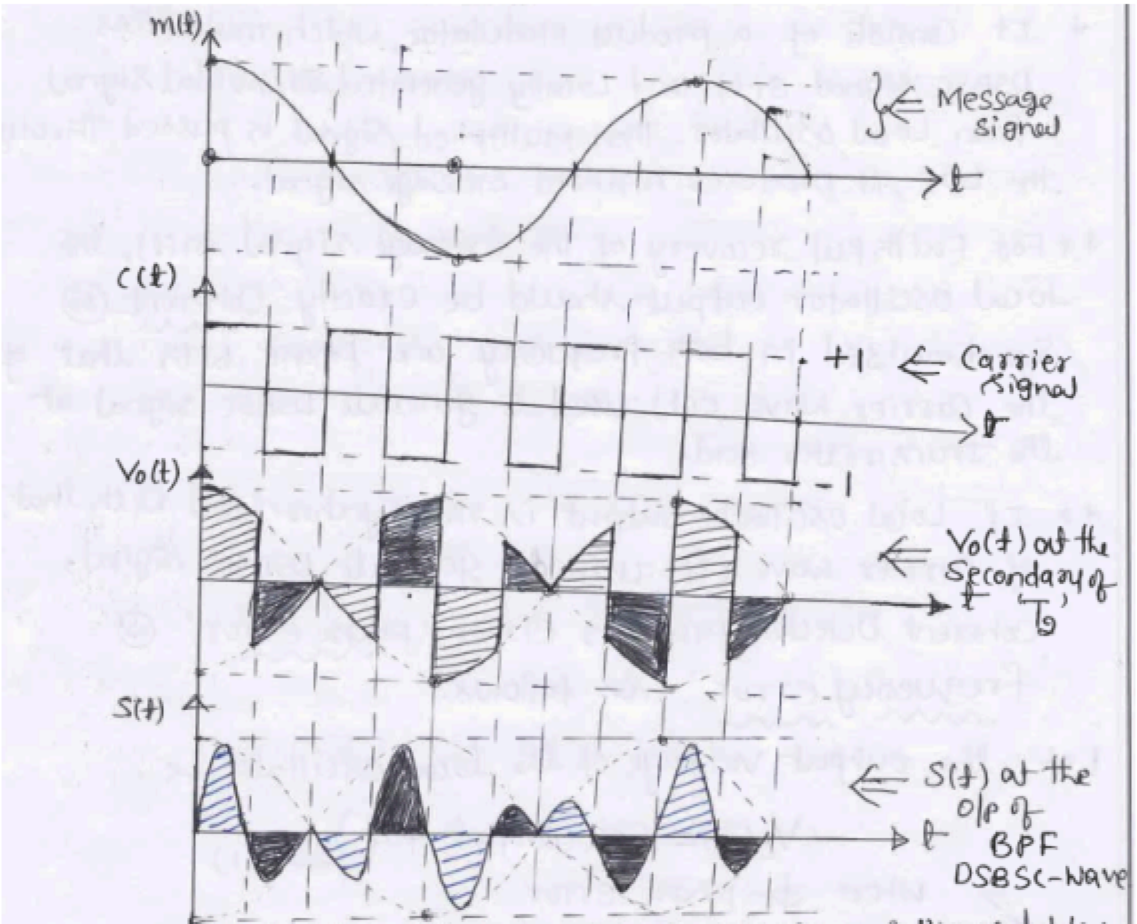
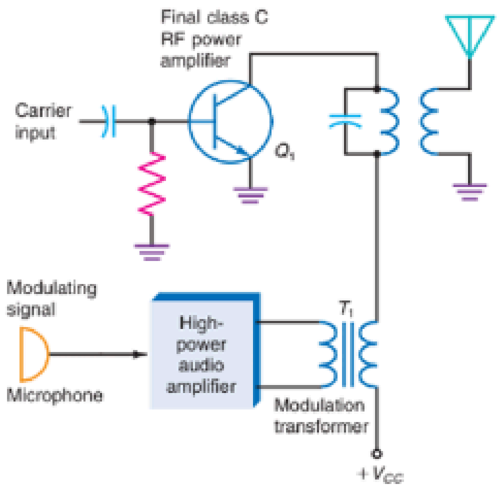


Figure 1.6(d): Time Domain Waveforms of Ring Modulator:-

3c)

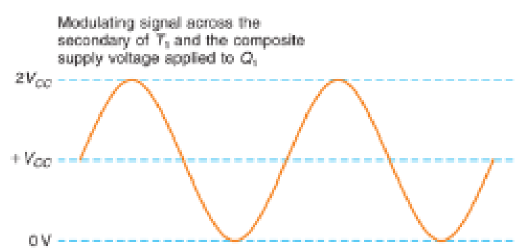
With neat diagrams, explain high level collector modulator.

The high-level modulator varies the voltage and power in the final RF amplifier stage of the transmitter. The result is high efficiency in the RF amplifier with high-quality performance. The circuit diagram of the transistor-based high-level modulator is as follows. The carrier signal is applied to the base of a high-power class C-tuned amplifier. The modulating signal is coupled using a transformer and is superimposed with the collector supply.



With a zero-modulation input signal, there is zero-modulation voltage across the secondary of T1, the collector supply voltage is applied directly to the class C-amplifier, and the output carrier is a steady sine wave.
When the modulating signal occurs, the AC voltage of the modulating signal across the secondary of the modulation transformer is added to and subtracted from the DC collector supply voltage. This varying supply voltage is then applied to the class C amplifier, causing the amplitude of the current pulses through transistor Q1 to vary. As a result, the amplitude of the carrier sine wave varies with

the modulated signal. When the modulation signal goes positive, it adds to the collector supply voltage, thereby increasing its value and causing higher current pulses and a higher amplitude carrier. When the modulating signal goes negative, it subtracts from the collector supply voltage, decreasing it. For that reason, the class C amplifier current pulses are smaller, resulting in a lower-amplitude carrier output. Thus the output amplitude varies in proportion to the message signal, or in other words the circuit produces AM,



A major disadvantage of collector modulators is the need for a modulation transformer that connects the audio amplifier to the class C amplifier in the transmitter. The higher the power, the larger and more expensive the transformer. for 100 percent modulation, the power supplied by the modulator must be one-half the total class C amplifier input power.

Fourth Semester B.E./B.Tech. Degree Examination, June/July 2024

Principles of Communication Systems

MODULE 1 & 2 :

		... explain high level collector modulator.	07	L2	CO2
		Module – 3			
Q.5	a.	With a neat block diagram, explain converting a phase modulated signal into a frequency modulated signal.	07	L1	CO3
	b.	Determine the frequency modulated signal $v_{FM} = V_C \sin(2\pi f_c t + m_f \sin 2\pi f_m t)$ interms of Bessel functions. Write the amplitude of sideband frequencies (J_n) interms of modulation index (m_f).	06	L3	CO3
	c.	Identify the noise suppression of frequency modulated signal.	07	L2	CO3

5a)

Case (i): Implementation of FM Using PM :-

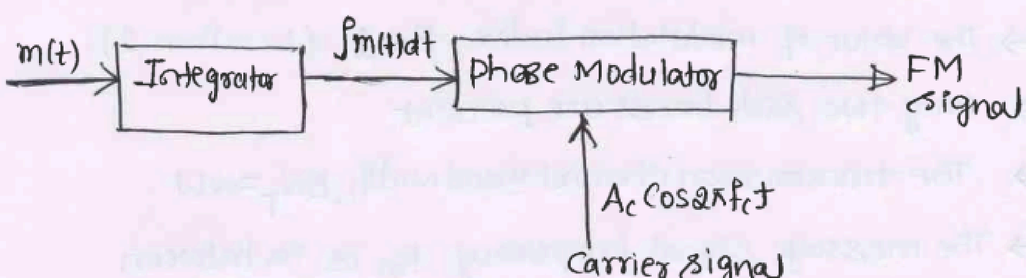
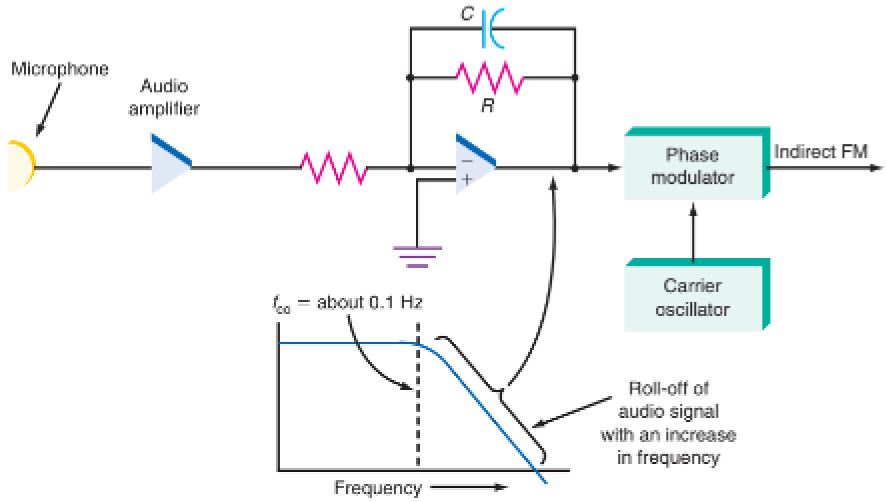


Figure 1: Generation of FM-signal Using Phase Modulator

↳ By changing phase modulator input signal to $\int m(t)\,dt$ we can generate FM signal as shown in figure 1.

Figure 5-5 Using a low-pass filter to roll off the audio modulating signal amplitude with frequency.



To make PM compatible with FM, the deviation produced by frequency variations in the modulating signal must be compensated for. This can be done by passing the intelligence signal through a low-pass *RC* network, as illustrated in Fig. 5-5. This low-pass filter,

called a *frequency-correcting network, predistorter,* or *1/f filter,* causes the higher modulating frequencies to be attenuated. Although the higher modulating frequencies produce a greater rate of change and thus a greater frequency deviation, this is offset by the lower amplitude of the modulating signal, which produces less phase shift and thus less frequency deviation. The predistorter compensates for the excess frequency deviation caused by higher modulating frequencies. The result is an output that is the same as an FM signal. The FM produced by a phase modulator is called *indirect FM.*

5b)

Determine the frequency modulated signa $v_{FM} = V_C \sin(2\pi f_c t + m_f \sin 2\pi f_m t)$ interms of Bessel functions. Write th amplitude of sideband frequencies (J_n) interms of modulation index (m_f).

ii) Wide-band FM :-

In Wide-band signal

↳ The Value of modulation index, $\beta \gg 1$. (Greater than 1)

↳ Infinite number of side bands are present.

** ↳ The message signal frequency, f_m is in between 30 HZ to 15 KHZ.

*** The Bandwidth of Wide band FM signal can be calculated from Carson's Rule shown in equation (1)

$$\boxed{BW_T = 2f_m + 2\Delta f_{max}} \Leftarrow$$ CARSON'S Rule to find Band width of wide band FM-Signal.

↳ The Maximum frequency deviation is 75 KHZ.

Applications of Wide band FM:-

* Wide-band FM technique is mainly used in High Quality Music signal transmission.

  Example: FM-channels

Comparison

Given the modulation index, the number and amplitudes of the significant sidebands can be determined by solving the basic equation of an FM signal. This equation is solved with a complex mathematical process known as Bessel functions.

$$s(t) = A_C \operatorname{Sin}(2\pi f_c t - \beta Cos \quad (2\pi f_m t))$$

OR

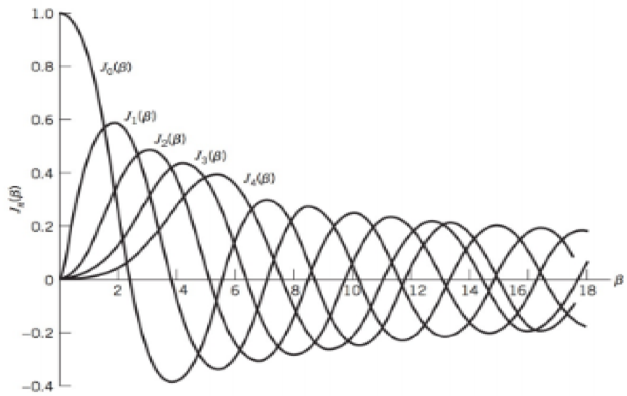
$$s(t) = A_C \cos(2\pi f_c t + \beta \sin(2\pi f_m t))$$

Using expansion and bessel approx

$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos[2\pi(f_c + nf_m)t]$$

$$S(f) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta)[\delta(f - f_c - nf_m) + \delta(f + f_c + nf_m)]$$

Plot of Bessel function of first kind



Bessel function table

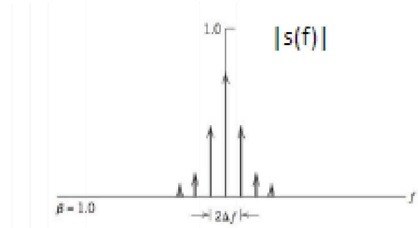
Modulation Index	Carrier	Sidebands (Pairs)															
		1st	2d	3d	4th	5th	6th	7th	8th	9th	10th	11th	12th	13th	14th	15th	16th
0.00	1.00	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—
0.25	0.98	0.12	—	—	—	—	—	—	—	—	—	—	—	—	—	—	—
0.5	0.94	0.24	0.03	—	—	—	—	—	—	—	—	—	—	—	—	—	—
1.0	0.77	0.44	0.11	0.02	—	—	—	—	—	—	—	—	—	—	—	—	—
1.5	0.51	0.56	0.23	0.06	0.01	—	—	—	—	—	—	—	—	—	—	—	—
2.0	0.22	0.58	0.35	0.13	0.03	—	—	—	—	—	—	—	—	—	—	—	—
2.5	−0.05	0.50	0.45	0.22	0.07	0.02	—	—	—	—	—	—	—	—	—	—	—
3.0	−0.26	0.34	0.49	0.31	0.13	0.04	0.01	—	—	—	—	—	—	—	—	—	—
4.0	−0.40	−0.07	0.36	0.43	0.28	0.13	0.05	0.02	—	—	—	—	—	—	—	—	—
5.0	−0.18	−0.33	0.05	0.36	0.39	0.26	0.13	0.05	0.02	—	—	—	—	—	—	—	—
6.0	0.15	−0.28	−0.24	0.11	0.36	0.36	0.25	0.13	0.06	0.02	—	—	—	—	—	—	—
7.0	0.30	0.00	−0.30	−0.17	0.16	0.35	0.34	0.23	0.13	0.06	0.02	—	—	—	—	—	—
8.0	0.17	0.23	−0.11	−0.29	−0.10	0.19	0.34	0.32	0.22	0.13	0.06	0.03	—	—	—	—	—
9.0	−0.09	0.24	0.14	−0.18	−0.27	−0.06	0.20	0.33	0.30	0.21	0.12	0.06	0.03	0.01	—	—	—
10.0	−0.25	0.04	0.25	0.06	−0.22	−0.23	−0.01	0.22	0.31	0.29	0.20	0.12	0.06	0.03	0.01	—	—
12.0	−0.05	−0.22	−0.08	0.20	0.18	−0.07	−0.24	−0.17	0.05	0.23	0.30	0.27	0.20	0.12	0.07	0.03	0.01
15.0	−0.01	0.21	0.04	0.19	−0.12	0.13	0.21	0.03	−0.17	−0.22	−0.09	0.10	0.24	0.28	0.25	0.18	0.12

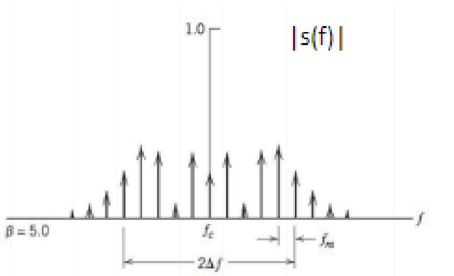
Spectrum of FM

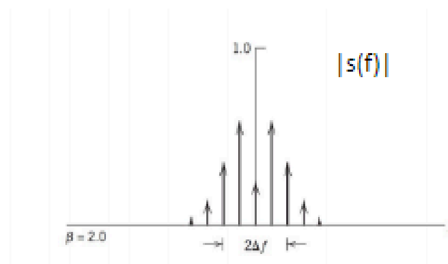
$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos[2\pi(f_c + nf_m)t]$$

$$S(f) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta)[\delta(f - f_c - nf_m) + \delta(f + f_c + nf_m)]$$

fm is fixed the amplitude varies:
Deviation varies





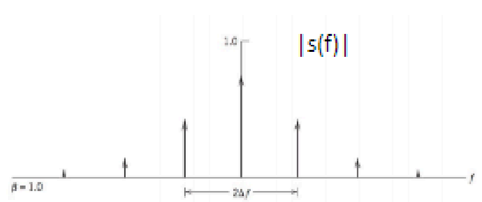


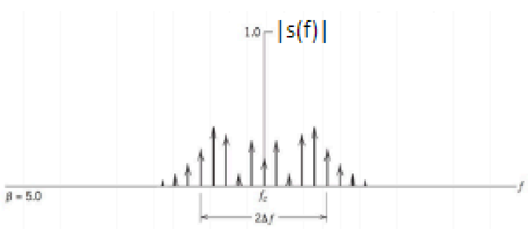
Spectrum of FM

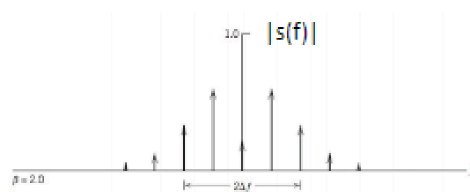
$$s(t) = A_c \sum_{n=-\infty}^{\infty} J_n(\beta)\cos[2\pi(f_c + nf_m)t]$$

$$S(f) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta)[\delta(f - f_c - nf_m) + \delta(f + f_c + nf_m)]$$

amplitude is fixed the fm varies:
Deviation is fixed. The bandwidth
will be limited to 2ΔF







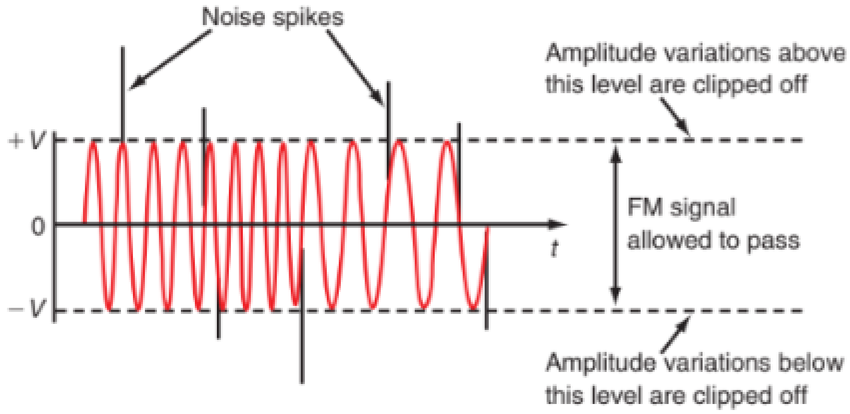
$$S(f) = \frac{A_c}{2} \sum_{n=-\infty}^{\infty} J_n(\beta)\left[\delta(f - f_c - nf_m) + \delta(f + f_c + nf_m)\right]$$

$n = 0, \pm 1, \pm 2, \pm 3$

5c)

Identify the noise suppression of frequency modulated signal.

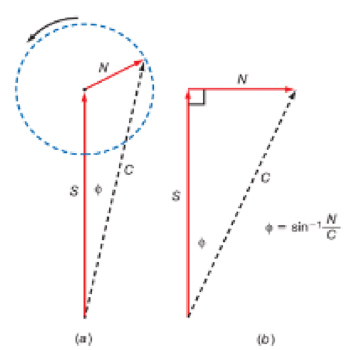
• Constant amplitude: Limiter



Noise and Phase Shift

• The noise amplitude added to an FM signal introduces a small frequency variation, or phase shift, which changes or distorts the signal.

How noise introduces a phase shift.



It is possible to determine just how much of a frequency shift a particular phase shift produces by using the formula

$$\delta = \phi(f_m)$$

where δ = frequency deviation produced by noise
ϕ = phase shift, rad
f_m = frequency of modulating signal

- The overall effect of the shift depends upon the maximum allowed frequency shift for the application. If very high deviations are allowed, i.e., if there is a high modulation index, the shift can be small and inconsequential.
- If the total allowed deviation is small, then the noise-induced deviation can be severe.

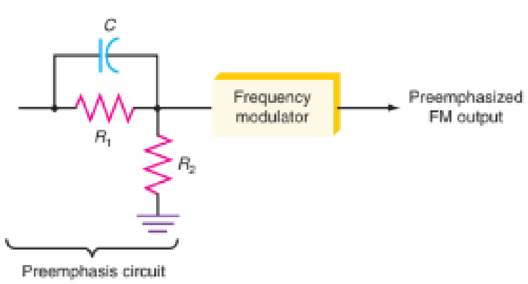
$$\frac{\text{Frequency deviation produced by noise}}{\text{Maximum allowed deviation}}$$

- Remember that the noise interference is of very short duration; thus, the phase shift is momentary, and intelligibility is rarely severely impaired. With heavy noise, human speech might be temporarily garbled, but so much that it could not be understood.

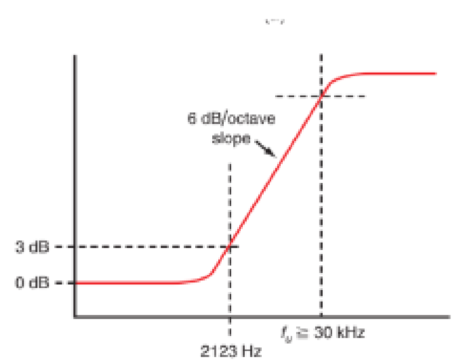
Pre-emphasis and De-emphasis

- Pre-emphasis and De-emphasis : to offset high frequency noise interference
- Noise can interfere with an FM signal, and particularly with the high-frequency components of the modulating signal. Since noise is primarily sharp spikes of energy, it contains a lot of harmonics and other high-frequency components.
- These frequencies can be larger in amplitude than the high-frequency content of the modulating signal, causing frequency distortion that can make the signal unintelligible.
- Audio (Low freq < 3k) and musical instruments (Higher)

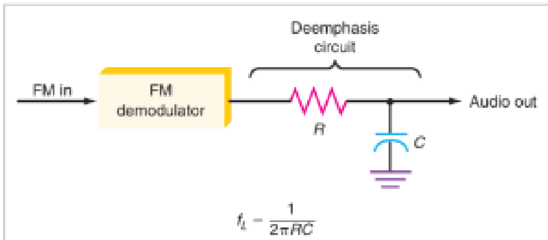
Pre-emphasis: amplifies high frequency component

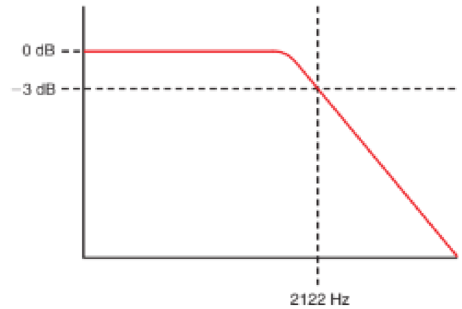


$$f_u = \frac{R_1 + R_2}{2\pi R_1 R_2 C}$$



De-emphasis-receiver





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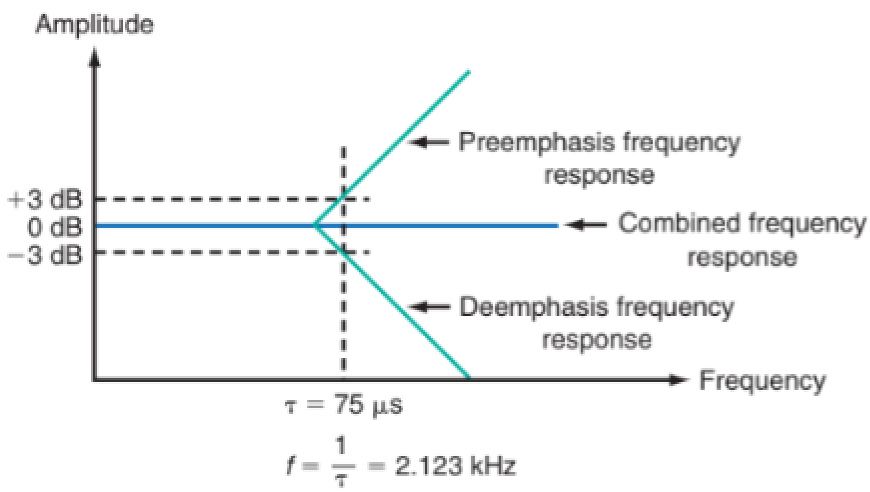


Q.6	a.	What is the maximum bandwidth of an FM signal with a deviation of 30 kHz and a maximum modulating signal of 5 kHz. (i) Using number of sidebands N = 9 (ii) Using Carson's rule	04	L2	CO3
	b.	Define phase locked loop. Explain with neat circuit diagram of FM demodulator using the IC 565.	08	L2	CO3
	c.	With neat block diagram, explain the concept of frequency modulation with an IC voltage controlled oscillator (IC NE566)	08	L2	CO3

a)

a. $m_f = \dfrac{f_d}{f_m} = \dfrac{30 \text{ kHz}}{5 \text{ kHz}} = 6$

Fig. 5-8 shows nine significant sidebands spaced 5 kHz apart for $m_f = 6$.

$$BW = 2f_m N = 2(5 \text{ kHz})\,9 = 90 \text{ kHz}$$

b. $BW = 2[f_{d\,(\max)} + f_{m\,(\max)}]$

$= 2(30 \text{ kHz} + 5 \text{ kHz})$

$= 2(35 \text{ kHz})$

$BW = 70 \text{ kHz}$

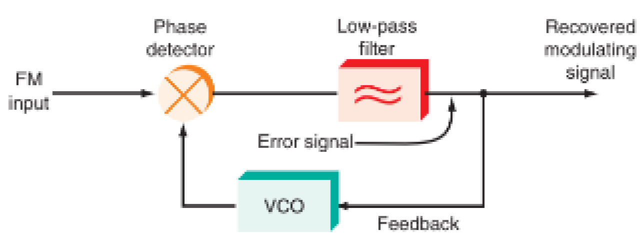
b)

A *phase-locked loop* (PLL) is a frequency- or phase-sensitive feedback control circuit used in frequency demodulation, frequency synthesizers, and various filtering and signal detection applications. All phase-locked loops have the three basic elements, shown in Fig. 6-17.

1. A phase detector is used to compare the FM input, sometimes referred to as the *reference signal,* to the output of a VCO.
2. The VCO frequency is varied by the dc output voltage from a low-pass filter.
3. The low-pass filter smoothes the output of the phase detector into a control voltage that varies the frequency of the VCO.

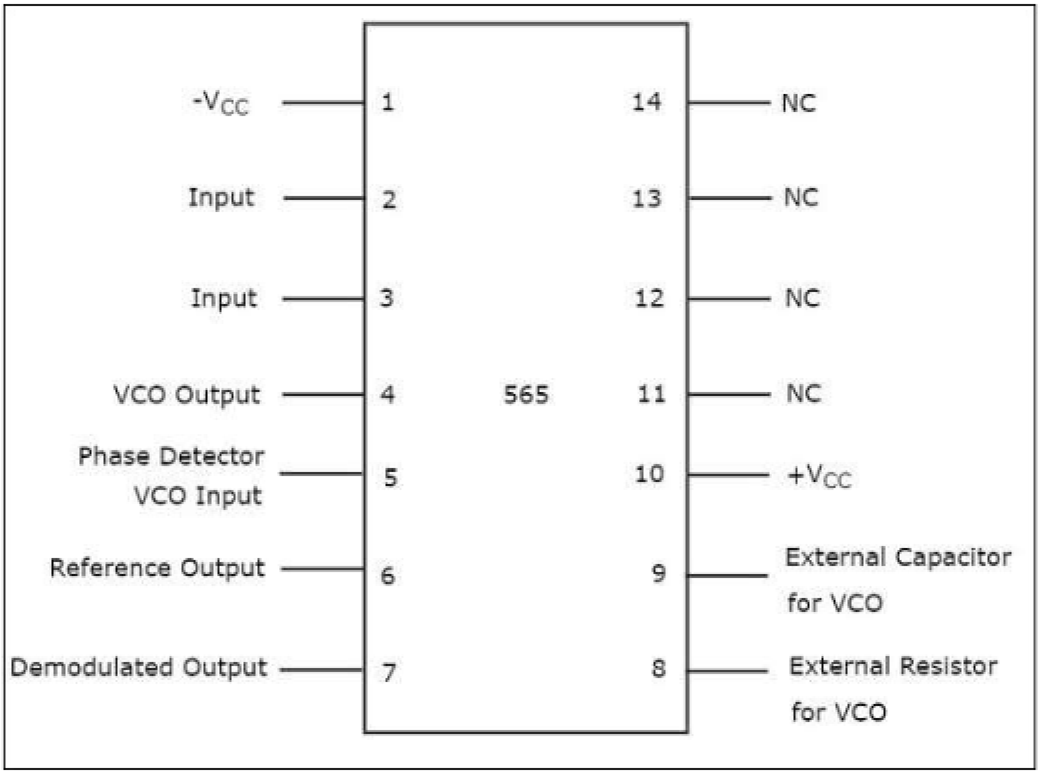
The primary job of the phase detector is to compare the two input signals and generate an output signal that, when filtered, will control the VCO. If there is a phase or frequency difference between the FM input and VCO signals, the phase detector output varies in proportion to the difference. The filtered output adjusts the VCO frequency in an attempt to correct for the original frequency or phase difference. This dc control voltage, called the *error signal,* is also the feedback in this circuit.

Figure 6-17 Block diagram of a PLL.



IC 565

IC 565 is the most commonly used phase locked loop IC. It is a 14 pin Dual-Inline Package (DIP). The **pin diagram** of IC 565 is shown in the following figure −



The purpose of each pin is self-explanatory from the above diagram. Out of 14 pins, only 10 pins (pin number 1 to 10) are utilized for the operation of PLL. So, the remaining 4 pins (pin number 11 to 14) are labelled with NC (No Connection).

The **VCO** produces an output at pin number 4 of IC 565, when the pin numbers 2 and 3 are grounded. Mathematically, we can write the output frequency, f_{out} of the VCO as.

$$f_{out} = \frac{0.25}{R_V C_V}$$

where,

R_V is the external resistor that is connected to the pin number 8

C_V is the external capacitor that is connected to the pin number 9

- By choosing proper values of R_V and C_V, we can fix (determine) the output frequency, f_{out} of VCO.
- **Pin numbers 4 and 5**are to be shorted with an external wire so that the output of VCO can be applied as one of the inputs of phase detector.
- IC 565 has an internal resistance of $3.6K\Omega$. A capacitor, C has to be connected between pin numbers 7 and 10 in order to make a **low pass filter** with that internal resistance.

Note that as per the requirement, we have to properly configure the pins of IC 565.

c)

Frequency modulation is a process of changing the frequency of a carrier wave in accordance with the slowly varying base band signal. The main advantage of this modulation is that it can provide better discrimination against noise.

A VCO is a circuit that provides an oscillating signal whose frequency can be adjusted over a control by Dc voltage. VCO can generate both square and triangular Wave signal whose frequency is set by an external capacitor and resistor and then varied by an applied DC voltage. IC 566 contains a current source to charge and discharge an external capacitor C1 at a rate set by an external resistor. R1 and a modulating DC output voltage.

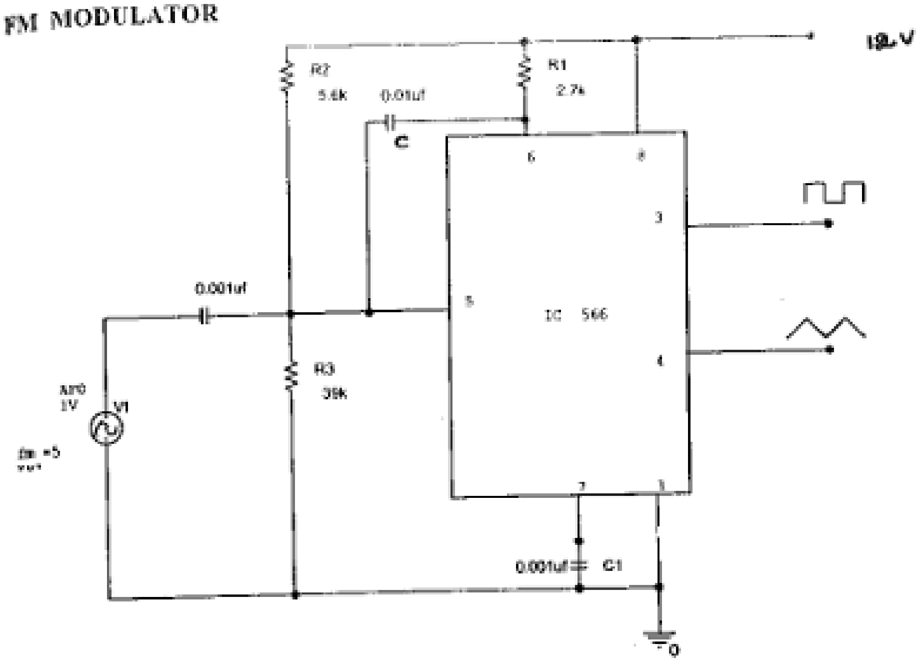
The Schmitt trigger circuit present in the IC is used to switch the current source between charge and discharge capacitor and triangular voltage developed across the capacitor and the square wave from the Schmitt trigger are provide as the output of the buffer amplifier.

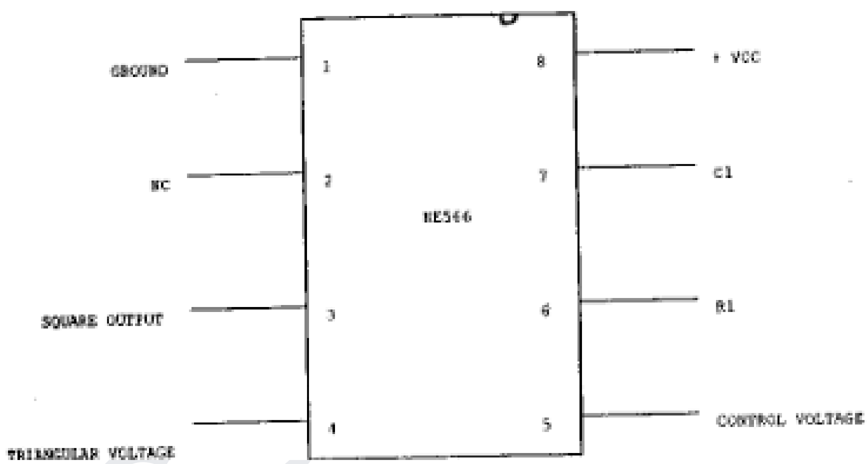
The R2 and R3 combination is a voltage divider, the voltage VC must be in the range ¾ VCC < VC < VCC. The modulating voltage must be less than ¾ VCC the frequency Fc can be calculated using the formula

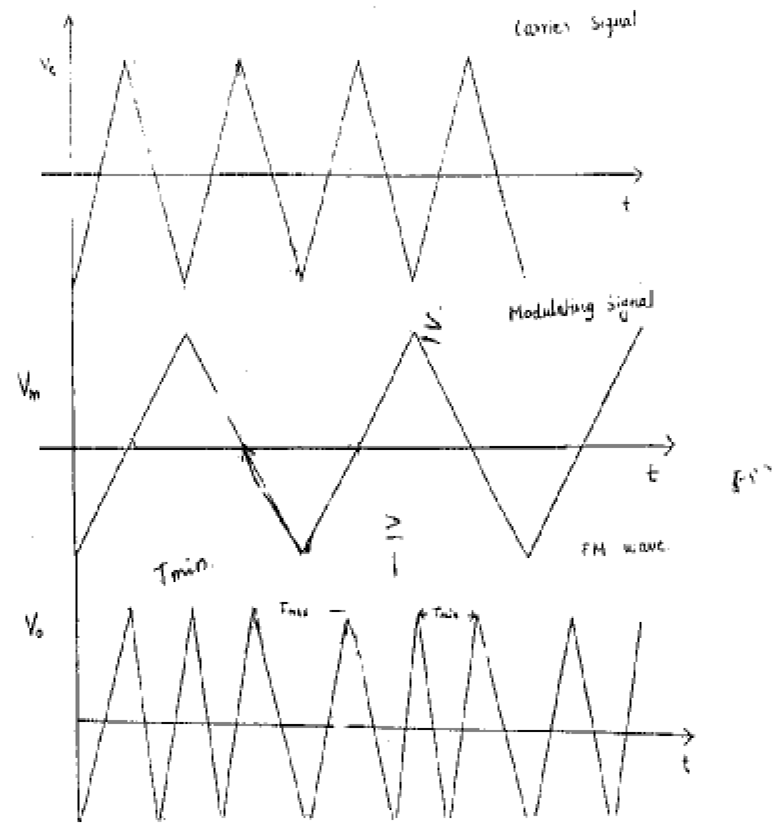
$$F_o = 2\ (V_{cc}-V_c)\ R1\ C1\ V_{cc}$$

For a fixed value of Vc and a constant C1 the frequency can be varied at 10:1 similarly for a constant R! C1 product value the frequency modulation can be done at 10:1 ratio

FM MODULATOR







		Module – 4			
Q.7	a.	Why digitize the analog signals? Explain the different processes used to convert the analog signal to digital signal.	06	L2	CO4
	b.	What is quantization process? Explain the different types of quantization with their important characteristics.	07	L2	CO4
	c.	Explain the concept of Time division multiplexing with a neat block diagram.	07	L2	CO4

a)

* WHY DIGITIZE ANALOG SOURCES :

There are many advantages that the transmission of digital information has over analog.

1) Digital systems are less sensitive to noise than analog.

2) With digital systems, it is easier to integrate different services. For example, video and the accompanying sound track, into the same transmission scheme.

3) The transmission scheme can be relatively independent of the source. For example, a digital transmission scheme that transmits voice at 10kbps could also be used to transmit computer data at 10kbps.

4) Circuitry for handling digital signals is easier to repeat and digital circuits are less sensitive to physical effects such as vibration and temperature.

5) Digital signals are simpler to characterize in terms of bits 1 and 0 and do not have variability as analog signals. This makes the associated hardware easier to design.

6) Various media sharing strategies known as multiplexing techniques are more easily implemented with digital transmission strategies.

7) Digital techniques make it easier to specify complex standards that [illegible] worldwide basis.

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8) The techniques such as equalization, especially adaptive versions, are easier to implement with digital transmission techniques.

The process involved in converting analogue signals into digital in physics is called digitisation, which entails two steps: sampling and quantisation. In sampling, the analogue signal is measured at regular intervals, and in quantisation, these measured values are approximated to the nearest value within a set range.

b)

* THE QUANTIZATION PROCESS :

The process of transforming sampled amplitude values of a message signal into a discrete amplitude value (levels) is referred to as quantization.

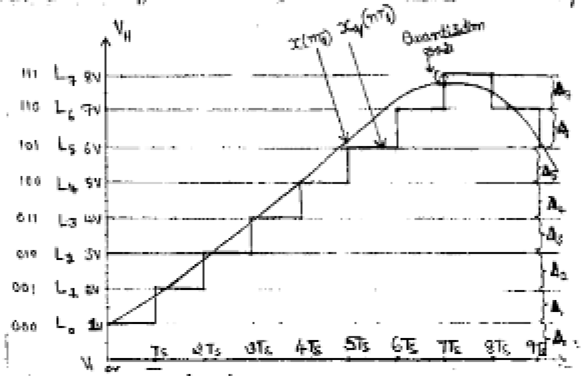


Fig: Quantization Process

* The signal $x(t)$ whose excursion is confined to the range from V_L to V_H being divided into 8-equal levels.

* Step size is denoted by 'Δ' and is given by

$$\Delta = \frac{V_H - V_L}{L}$$

where $L = 2^R$ and R is no. of bits.

$$\therefore \quad \Delta = \frac{V_H - V_L}{2^R}$$

If the step size 'Δ' is maintained same through the process of quantization, then it is called "uniform Quantization".

* The difference between the continuous amplitude sample level and quantized signal level is known as quantization error.

$$e(t) = x_q(nT_s) - x(nT_s)$$

where Quantization error varies from $+\Delta/2$ to $-\Delta/2$.

* The random errors due to quantization process produce a noise at the output of the quantizer and this noise is referred to as Quantization noise.

* Consider fig(A), the sampling, Quantizing and coding of an analog signal is as follows.

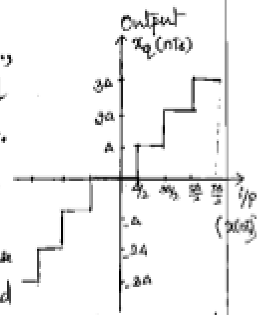
Sampled values of an analog signal	1.7V	2.7V	3.9V	5V	6.2V	7.2V	7.7V	7.4V
Nearest Quantizer level	L_1	L_2	L_3	L_4	L_5	L_6	L_7	L_8
Quantizer level voltage	2V	3V	4V	5V	6V	7V	8V	7V
Binary Code	001	010	011	100	101	110	111	110

* There are two types of quantizer they are
  - Mid-tread type quantizer
  - Mid-riser type quantizer

1) Mid-Tread type Quantizer :

* In mid-tread quantizer, the decision threshold of the quantizers are located at $\pm\frac{\Delta}{2}, \pm\frac{3\Delta}{2}, \ldots$, and the representation levels are located at $0, \pm\Delta, \pm 2\Delta$, where Δ is the step size.

* A uniform quantizer characterized in this way is referred to as a symmetric quantizer of the midtread type, because the origin lies in the middle of a tread of a staircase graph. Here, Quantization levels are odd number.



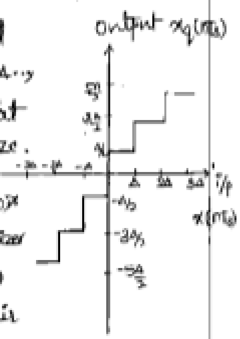
Midtread Quantizer

2) Mid-Riser type Quantizer :

* In midriser quantizer, the decision threshold of the quantizers are located at $0, \pm\Delta, \pm 2\Delta, \ldots$, and the representation levels are located at $\pm\frac{\Delta}{2}, \pm\frac{3\Delta}{2}, \pm\frac{5\Delta}{2}, \ldots$, where Δ is the step size.

* A uniform quantizer characterized in this way is referred to as a symmetric quantizer of the midriser type, because the origin of the lies in the middle of a riser of the staircase graph.

* Here Quantization levels are even number.



Midrise Quntizer

c)

* TIME DIVISION MULTIPLEXING : [TDM]

Time Division Multiplexing is a method of transmitting and receiving independent signals over a common channel by means of synchronised switches at each end of transmission line so that each signal appears on the line only a fraction of time in an alternating pattern.

* Fig(5) shows the block diagram of TDM system.

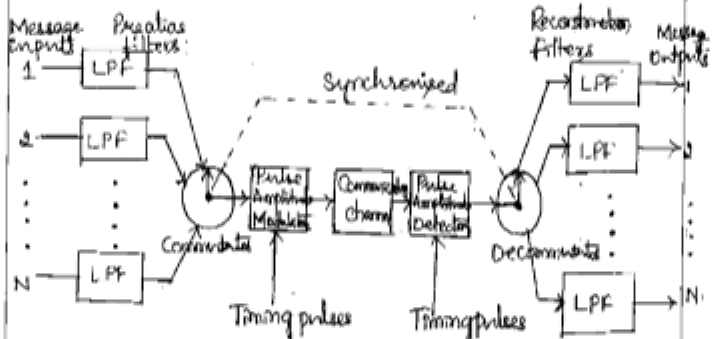


Fig 5 : Block Diagram of TDM System.

* The concept of TDM is illustrated in the fig(5). The Lowpass filters are used to remove high frequency components present in the message signal. The output of the pre-alias filters are then fed to a commutator, which is usually implemented using electronic switching circuitry.

* The function of commutator is as follows:

1) To take a narrow sample of each of the 'N' samples of input at a rate of $f_s \geq 2W$.

2) To sequentially interleave (multiplex) these 'N' samples inside a sampling interval $T_s = 1/f_s$.

* The multiplexed signal is then applied to a pulse amplitude modulator whose purpose is to transform the multiplexed signal into a form suitable for transmission over a common channel.

* At the receiving end, the pulse amplitude demodulator performs the reverse operation of PAM and the decommutator distributes the signals to the appropriate low pass reconstruction filters. The decommutator operates in synchronisation with the commutator.

Q.8	a.	Define PCM (Pulse Code Modulation). Explain the basic elements of a PCM system with neat diagrams.	06	L2	CO4
	b.	For the data stream 01101001. Draw the following line code waveforms: (i) Unipolar NRZ (ii) Polar NRZ (iii) Unipolar RZ (iv) Bipolar RZ (v) Manchester code (vi) Differential coding	09	L3	CO4
	c.	State and prove the sampling theorem. Explain with neat sketches and equations.	05	L2	CO4

a)

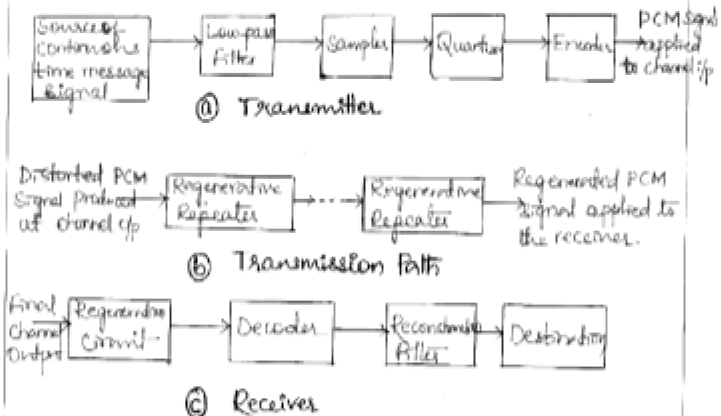
* PULSE CODE MODULATION :

* In pulse code Modulation (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 and amplitude.

* The basic operations performed in the transmitter of a PCM system are sampling, quantizing and encoding as shown in fig 6(a). The Lowpass filter prior to sampling

is included to prevent aliasing of the message signal. The quantizing and encoding operations are usually performed in the same circuit, which is called an analog-to-digital converter.

* The basic operations in the receiver are regeneration of impaired signals, decoding and reconstruction of the train of quantized samples as shown in fig 6(c). Regeneration also occurs at intermediate points along the transmission path as necessary as indicated in fig 6(b).



Fig(6) : The basic elements of a PCM system.

* SAMPLING :

The incoming message signal is sampled with a train of narrow rectangular pulses so as to closely approximate the instantaneous sampling process. In order to ensure perfect reconstruction of the message signal at the receiver, the sampling rate must be greater than or equal to the highest frequency component 'W' of the message signal in accordance with the sampling theorem.

$$f_s \geq 2W.$$

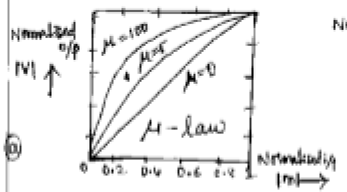
* Quantization :

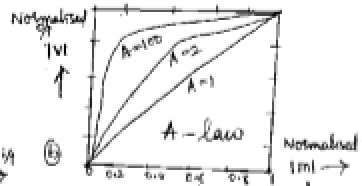
The sampled version of the message signal is then quantized, thereby providing a new representation of the signal that is discrete in both time and amplitude.

* For uniform Quantization, we have mid-tread and mid-rise quantizer and for non-uniform quantization, we have two compression laws μ-law and A law.

* The use of a non-uniform quantizer is equivalent to passing the baseband signal through a compressor and then applying the compressed signal to a uniform quantizer. A particular form of compression law that is used in practice is the so called μ-law, defined by

where m and v are normalized input & output vtg and 'μ' is positive constant.





* Another compression law that is used in practice is the so called 'A'-law as shown above.

$$|v| = \begin{cases} \dfrac{A|m|}{1+\log A}, & 0 \leq |m| \leq \dfrac{1}{A} \\[2ex] \dfrac{1+\log(A|m|)}{1+\log A}, & \dfrac{1}{A} \leq |m| \leq 1 \end{cases}$$

* Encoding :

* In combining the processes of sampling and quantization the specification of a continuous message (baseband) signal becomes limited to a discrete set of values, but not in the form best suited to transmission over a line or radio path.

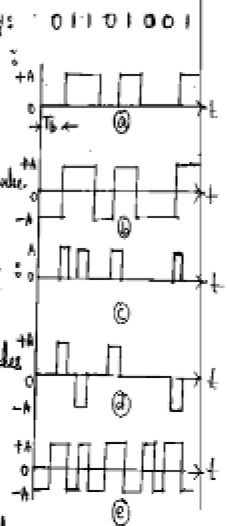
* In a binary code each symbol may be either of two distinct values or kinds, such as the presence or absence of a pulse. The two symbols of a binary code are customarily denoted as 0 and 1.

* **Line code** : It is a line code that a binary stream

of data takes on an electrical representation. The five line codes are illustrated in fig(7).

1) **Unipolar Nonreturn to zero (NRZ) signaling** :

In this line code symbol '1' is represented by transmitting a pulse of amplitude 'A' and symbol '0' is represented by switching off the pulse.



2) **Polar Non-return to zero (NRZ) signalling** :

In this line code, symbol 1 and 0 are represented by transmitting pulses of amplitudes +A and -A respectively.

3) **Unipolar Return to zero (RZ) signalling** :

Here, symbol 1 is represented by a rectangular pulse of amplitude A and half-symbol width and symbol 0 is represented by transmitting no pulse.

ⓐ → Unipolar NRZ
ⓑ → Polar NRZ
ⓒ → Unipolar RZ
ⓓ → Bipolar RZ
ⓔ → Manchester code

4) **Bipolar Return to zero (BRZ) signalling** :

this line code uses three amplitude levels as shown in fig. Specifically, the +ve & -ve pulses

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5) Split-phase (Manchester code)

Symbol 1 is represented a positive pulse of amplitude 'A' followed by a negative pulse of amplitude -A with both pulses being a half-symbol wide. For symbol '0', the polarities of these two pulses are reversed.

* Differential Encoding :

This method is used to encode information in terms of signal transitions. In particular, a transition is used to designate symbol 0 in the incoming binary data stream, while no transition is used to designate symbol 1 as shown in fig.

(a) Original binary data 0 1 1 0 1 0 0 1

(b) Differentially encoded data 1 0 0 0 1 1 0 1 1

(c) Waveform

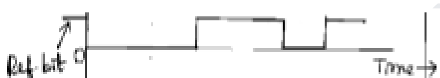


Fig : Differential encoding.

* REGENERATION :

The distorted PCM wave obtained from the transmitter is sent to the amplifier equilizer. The output of equilizer device is passed to the Decision making device to decide the signal in terms of 1 or 0 (coded o/p).

Fig : Block diagram of a regenerative repeater.

* Decoding :

The decoding process involves generating a pulse the amplitude of which is the linear sum of all the pulses in the codeword, with each pulse being weighted by its place value ($2^0, 2^1, 2^2 \ldots, 2^{R-1}$) in the code, where 'R' is the number of bits per sample.

* FILTERING :

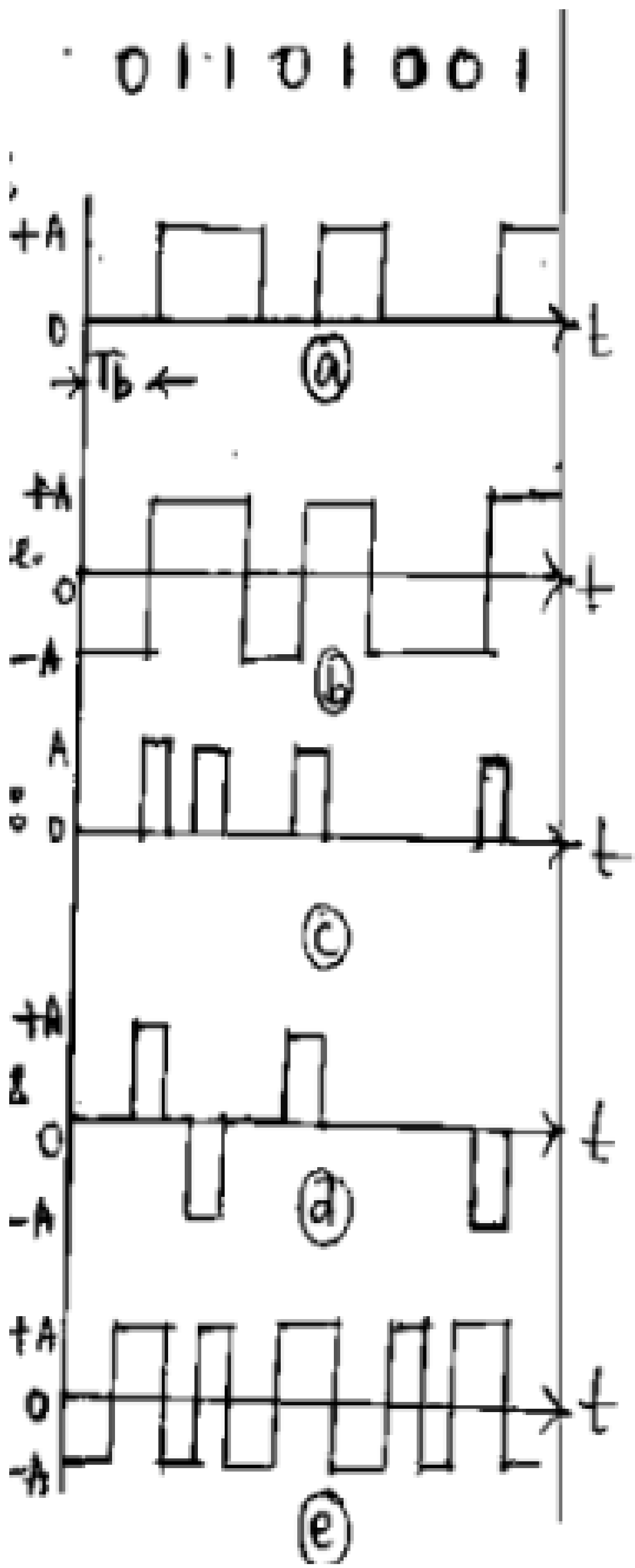
The final operation in the receiver is to recover the message signal wave by passing the decoder output through a lowpass reconstruction filter whose cutoff frequency is equal to the message bandwidth 'W'.

* MULTIPLEXING :

In applications using PCM, it is natural to multiplex different message sources by time division whereby each source keeps its individuality throughout the journey from the transmitter to receiver.

* This individuality accounts for the comparative ease with which message sources may be dropped or reinserted in a time division multiplex system.

b) 01101001


0 1 1 0 1 0 0 1
+A
0
Tb
t
a
+A
0
-A
b
A
0
c
+A
0
-A
d
+A
0
-A
e

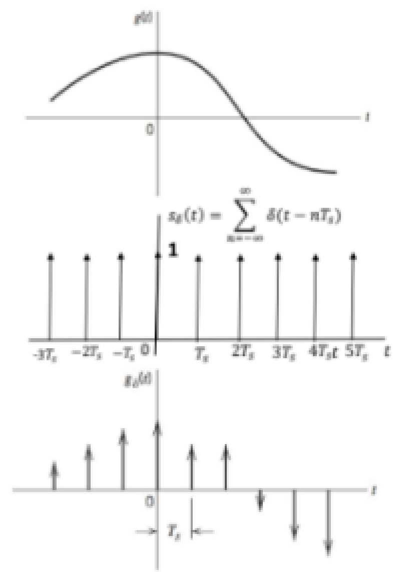
c)

Sampling Theorem

Sampling theorem states that any continuous time signal can be completely represented in its periodic samples and can be recovered back if the sampling frequency is greater than or equal to twice the highest frequency component of base band (message)signal.

$$f_s \geq 2W$$

$f_s = Samping\ frequency$ and W= bandwidth of message signal



Consider an arbitrary signal $g(t)$ of finite energy, which is specified for all time.
Consider a train of unit impulses separated at a distance T_s and represented by $s_\delta(t)$
Multiplying $g(t)$ with $s_\delta(t)$, yields a ideal sampled signal $g_\delta(t)$

Ideal Sampled signal thus can be written as

$$g_\delta(t) = g(t)s_\delta(t) \quad (1)$$

$$g_\delta(t) = g(t)\sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$\boxed{g_\delta(t) = \sum_{n=-\infty}^{\infty} g(nT_s)\,\delta(t - nT_s)} \quad (2)$$

From equation (1)

$$g_\delta(t) = g(t)s_\delta(t)$$

$\downarrow$ F.T.

$$G_\delta(f) = [G(f)] * [S_\delta(f)]$$

$$G_\delta(f) = [G(f)] * [f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s)]$$

($\because$ Fourier Transform of periodic impulse train is a periodic impulse train with change in amplitude)

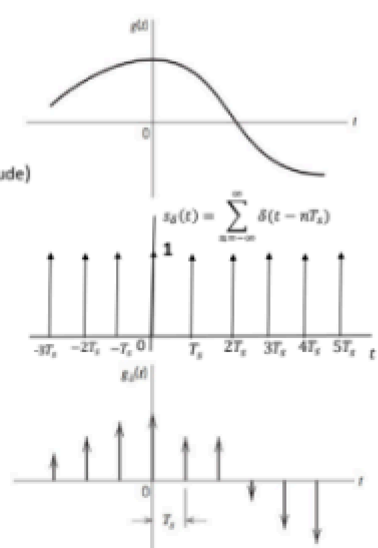
$$G_\delta(f) = f_s \sum_{n=-\infty}^{\infty} G(f - nf_s) \quad (3)$$

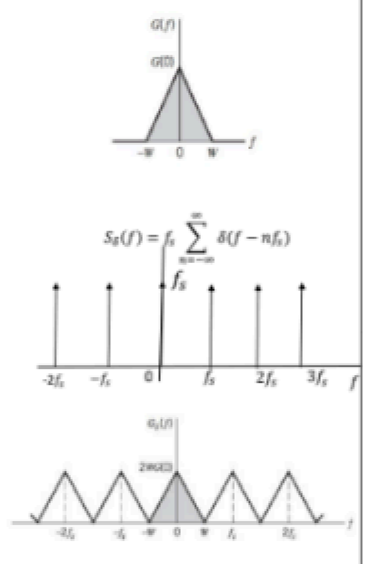
($\because$ by convolution property of impulse $G(f) * \delta(f - nf_s) = G(f - nf_s)$)

We can rewrite equation (3) as

$$G_\delta(f) = f_s G(f) + \sum_{\substack{n=-\infty \\ n\neq 0}}^{\infty} G(f - nf_s) \quad (4)$$
$$n = 0$$

$$where\ f_s = 2W$$





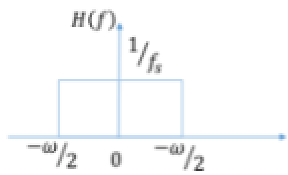
Reconstruction of original signal through samples

Signal $g(t)$ can be reconstructed from ideal sampled signal $g_\delta(t)$ using a reconstruction filter $h(t)$.

The characteristic of reconstruction filter is given as

1) The amplitude of the reconstruction filter must be $1/f_s$
2) It's bandwidth must be equivalent to ω Hz

$$H(f) = \begin{cases} 1/f_s ; & -\omega/2 \le f \le -\omega/2 \\ 0; elsewhere & \end{cases} \quad (5)$$



Taking inverse Fourier transform of equation (5)

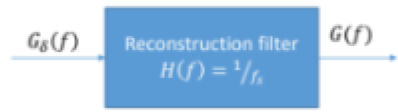
$$h(t) = sinc(2\omega t) \quad (6)$$

Passing equation (4) from a LPF

$$G_\delta(f) = f_s G(f)$$

$$G(f) = \frac{1}{f_s} G_\delta(f) \quad (7)$$

It is like passing $G_\delta(f)$ from reconstruction flter



$$G(f) = H(f) G_\delta(f) \quad (8)$$

Taking inverse Fourier transform

$$g(t) = h(t) * g_\delta(t)$$

$$g(t) = sinc(2\omega t) * \left[\sum_{n=-\infty}^{\infty} g(nT_s)\,\delta(t - nT_s) \right]$$

$$g(t) = \left[\sum_{n=-\infty}^{\infty} g(nT_s) sinc(2\omega(t - nT_s)) \right] \quad (9)$$

Equation (9) is known as interpolation formula

		Module – 5			
Q.9	**a.**	Develop a code to generate and plot eye diagram.	**06**	**L3**	**CO5**
	b.	Define noise factor and noise figure. Also explain noise in cascade connection.	**06**	**L2**	**CO5**
	c.	Define Inter Symbol Interference (ISI). Outline baseband binary data transmission system with neat block diagram and equations.	**08**	**L1**	**CO5**

a) NEED TO INCLUDE CODE FOR EYE DIAGRAM

b)

Expressing Noise Levels

- The noise quality of a receiver can be expressed as in terms of noise figure, noise factor, noise temperature, and SINAD.
- Noise Factor (Noise ratio) and Noise Figure:

$$NR = \frac{S/N \text{ input}}{S/N \text{ output}}$$

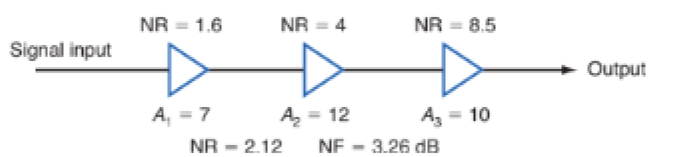
When the noise factor is expressed in decibels, it is called the noise figure (NF):

NF = 10 log NR dB

Noise in Cascaded Stages

- The noise performance of a receiver is determined in the very first stage of the receiver, usually an RF amplifier or mixer. Design of these circuits must ensure the use of very low noise components, taking into consideration current, resistance, bandwidth, and gain figures in the circuit. Beyond the first and second stages, noise is basically no longer a problem.

Noise in cascaded stages of amplification.



The formula used to calculate the overall noise performance of a receiver or of multiple stages of RF amplification, called *Friis' formula*, is

$$NR = NR_1 + \frac{NR_2 - 1}{A_1} + \frac{NR_3 - 1}{A_1 A_2} + \cdots + \frac{NR_n - 1}{A_1, A_2, \cdots, A_{n-1}}$$

where NR = noise ratio

NR_1 = noise ratio of input or first amplifier to receive the signal
NR_2 = noise ratio of second amplifier
NR_3 = noise ratio of third amplifier, and so on
A_1 = power gain of first amplifier
A_2 = power gain of second amplifier
A_3 = power gain of third amplifier, and so on

$$NR = 1.6 + \frac{4 - 1}{7} + \frac{8.5 - 1}{(7)(12)} = 1.6 + 0.4286 + 0.0893 = 2.12$$

e noise figure is

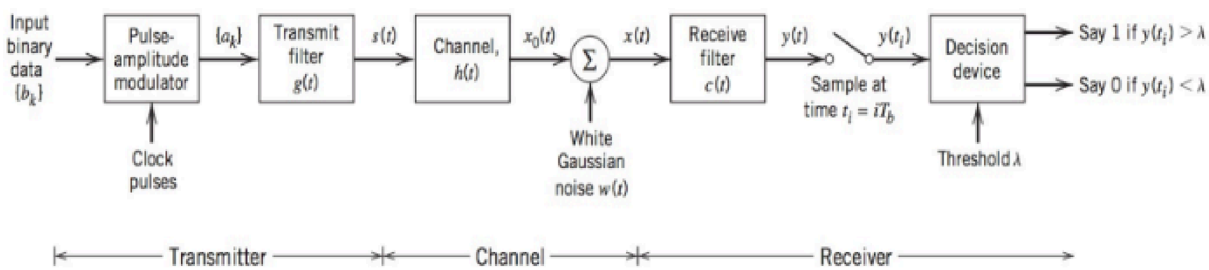
$$NF = 10 \log NR = 10 \log 2.12 = 10(0.326) = 3.26 \text{ dB}$$

What this calculation means is that the first stage controls the noise performance for the whole amplifier chain.

c)

Intersymbol Interference- baseband binary PAM system

- Arises when the communication channel is dispersive- the channel has a frequency-dependent amplitude spectrum, for eg band-limited channel
- The baseband transmission of digital data, the use of discrete pulse-amplitude modulation (PAM) is the most efficient one in terms of power and bandwidth utilization



$$a_k = \begin{cases} +1 & \text{if symbol } b_k \text{ is } 1 \\ -1 & \text{if symbol } b_k \text{ is } 0 \end{cases}$$

$$s(t) = \sum_k a_k g(t - kT_b)$$

- The signal s(t) is modified as result of transmission through the channel of impulse response h(t).In addition, the channel adds random noise to the signal at the receiver input. The noisy signal x(t) is then passed through a receive filter of impulse response c(t).
- Double convolution involving the impulse response g(t) of the transmit filter, the impulse response h(t) of the channel, and the impulse response c(t) of the receive filter, results in
- µ as a scaling factor to account for amplitude changes incurred in the course of signal transmission through the system.

$$\mu p(t) = g(t) \star h(t) \star c(t)$$

$$y(t) = \mu \sum_k a_k p(t - kT_b) + n(t)$$

$$p(0) = 1$$

$$\mu P(f) = G(f)H(f)C(f)$$

Contd...
at receiver end

- The term n(t) is the noise produced at the output of the receive filter due to the additive noise w(t) at the receiver input. It is customary to model w(t) as a white Gaussian noise of zero mean.
- The resulting filter output y(t) is sampled synchronously with the transmitter, with the sampling instants being determined by a clock or timing signal that is usually extracted from the receive-filter output.
- Finally, the sequence of samples thus obtained is used to reconstruct the original data sequence using a decision device.
- Specifically, the amplitude of each sample is compared to a threshold.
  - If the threshold is exceeded, a decision is made favoring symbol 1.
  - If the threshold is not exceeded, a decision is made favoring symbol 0.
  - If the sample amplitude equals the threshold exactly, the receiver simply makes a guess.

- The receive filter output y(t) is sampled at time ti = iTb (with i taking on integer values), yielding

$$
\begin{aligned}
y(t_i) &= \mu \sum_{k=-\infty}^{\infty} a_k p[(i-k)T_b)] + n(t_i) \\
&= \mu a_i + \mu \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} a_k p[(i-k)T_b)] + n(t_i)
\end{aligned}
$$

Without ISI

$$y(t_i) = \mu a_i$$

- The first term represents the contribution of the ith transmitted bit.
- The second term represents the residual effect of all other transmitted bits on the decoding of the ith bit; this residual effect due to the occurrence of pulses before and after the sampling instant t=ti is called intersymbol interference (ISI).
- The last term n(ti) represents the noise sample at time ti

- Under these ideal conditions, the ith transmitted bit is decoded correctly.
- The unavoidable presence of ISI and noise in the system, however, introduces errors in the decision device at the receiver output.
- Therefore, in the design of the transmit and receive filters, the objective is to minimize the effects of noise & ISI and deliver the digital data to its destination with the smallest error rate possible.
- When the signal-to-noise ratio is high, as is the case in a telephone system, for example, the operation of the system is largely limited by ISI rather than noise; we may ignore n(t)

		OR			
Q.10	a.	Explain bandwidth requirements of TI systems.	06	L1	CO5
	b.	Write short notes on: (i) Signal to noise ratio (ii) External noise (iii) Internal noise	08	L1	CO5
	c.	An RF amplifier has an S/N ratio of 8 at the input and an S/N ratio of 6 at the output. What are the noise factor, noise figure and noise temperature?	06	L3	CO5

a)

Signal-to-Noise Ratio(SNR)

- The stronger the signal and the weaker the noise, the higher the S/N ratio.
- Signals can be expressed in terms of voltage or power.

$$\frac{S}{N} = \frac{V_s}{V_n} \quad \text{or} \quad \frac{S}{N} = \frac{P_s}{P_n}$$

where V_s = signal voltage
V_n = noise voltage
P_s = signal power
P_n = noise power

- that the signal voltage is 1.2 μV and the noise is 0.3 μV. The S/N ratio is 1.2/0.3 = 4.
- if the signal power is 5 μW, the power is 125 nW, the S/N ratio is 5 μW/ 125 nW = 40

For voltage: $\text{dB} = 20 \log \frac{S}{N} = 20 \log 4 = 20(0.602) = 12 \text{ dB}$

For power: $\text{dB} = 10 \log \frac{S}{N} = 10 \log 40 = 10(1.602) = 16 \text{ dB}$

External Noise

- industrial, atmospheric, or space.
- shows up as a random ac voltage and can be seen on an oscilloscope. The amplitude varies over a wide range and the frequency too.
- Noise in general contains all frequencies, varying randomly. Called white noise.
- The key to reliable communication, then, is simply to generate signals at a high enough power to overcome external noise.
- Industrial Noise
- Atmospheric Noise
- Extraterrestrial Noise

- Industrial Noise
- Ex:automotive ignition systems, electric motors, and generators, gas-fi lled lights.
- Any electrical equipment that causes high voltages or currents cause noise.
- Atmospheric Noise
- earth's atmosphere – static -Static usually comes from lightning-megawatt power- impact on signals at frequencies below 30 MHz
- Extraterrestrial Noise
- solar and cosmic -sources – SUN- noise sprectrum- big amount of noise that causes tremendous radio signal interference and makes many frequencies unusable for communication.
- Noise generated by stars –cosmic -distances between those stars and earth –less impact.

Internal Noise

- Electronic components in a receiver- resistors, diodes, and transistors are major sources of internal noise.
- Types: Thermal noise,
- semiconductor noise,
- and intermodulation distortion- some design control.

- Thermal noise :phenomenon known as thermal agitation, the random motion of free electrons in a conductor caused by heat. Increasing the temperature causes this atomic motion to increase.
- Thermal agitation is often referred to as white noise or Johnson noise – just as white light. Filtered or band-limited noise is referred to as pink noise.
- The noise power is proportional to the bandwidth of any circuit to which it is applied. Filtering can reduce the noise level, but does not eliminate it entirely.

The amount of open-circuit noise voltage appearing across a resistor or the input impedance to a receiver can be calculated according to Johnson's formula

$$v_n = \sqrt{4kTBR}$$

where v_n = rms noise voltage
k = Boltzman's constant (1.38×10^{-23} J/K)
T = temperature, K (°C + 273)
B = bandwidth, Hz
R = resistance, Ω

What is the open-circuit noise voltage across a 100-kV resistor over the frequency range of direct current to 20 kHz at room temperature (25°C)?

$$v_n = \sqrt{4kTBR}$$
$$= \sqrt{4(1.38 \times 10^{-23})(25 + 273)(20 \times 10^3)(100 \times 10^3)}$$
$$v_n = 5.74\ \mu V$$

- Since noise voltage is proportional to resistance value, temperature, and bandwidth, noise voltage can be reduced by reducing resistance, temperature, and bandwidth or any combination to the minimum level acceptable for the given appl ication.

Thermal noise can also be computed as a power level. Johnson's formula is then

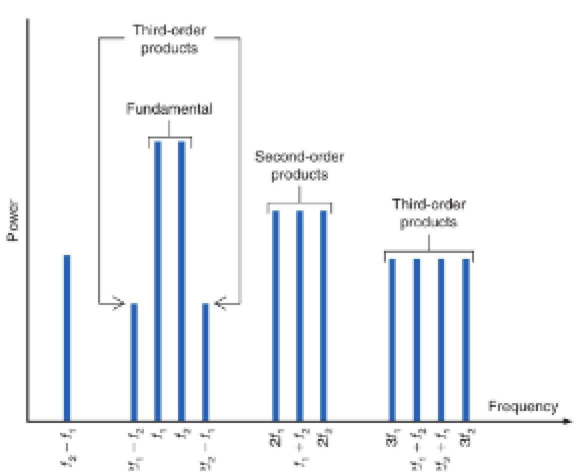
$$P_n = kTB$$

where Pn is the average noise power in watts.

Semiconductor Noise : diodes and transistors are major contributors of noise. In addition to thermal noise, semiconductors produce **shot noise, transit-time noise, and flicker noise.**

- transit-time noise: transit time refers to how long it takes for a current carrier such as a hole or electron to move from the input to the output.
- Flicker noise or excess noise: occurs in resistors and conductors. Due to minute random variations of resistance in the semiconductor material. It is directly proportional to current and temperature.

- Intermodulation Distortion :generation of new signals and harmonics caused by circuit nonlinearities.
- Nonlinearities produce modulation or heterodyne effects. Many frequencies – large no: of sum and difference frequencies.
- Falls within BW. Cant filter it. Get added as noise.
- The key to minimizing these extraneous intermodulation products is to maintain good linearity through biasing and input signal level control.



b)

c)