KLS Vishwanathrao Deshpande Institute of Technology

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(Approved by AICTE, New Delhi, Affiliated to VTU, Belagavi)
(Recognized Under Section 2(f) by UGC, New Delhi)
Udyog Vidya Nagar, Haliyal - 581 329, Dist.: Uttara Kannada
Phone: 08284 - 220861, 220334, 221409, Fax: 08284 - 220813
www.klsvdit.edu.in | principal@klsvdit.edu.in | hodece@klsvdit.edu.in

DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

University / Model Question Paper Scheme & Solution

Faculty Name	:	Prof. Rohini Kallur
Course Name	:	Principles of communication Systems
Course Code	:	BEC402
Year of Question Paper	:	June / July 2024
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Faculty Member HoD Dean (Acad.)

Head of the Department
Dept. of Electronics & Communication Engg
KLS V.D.I.T. HALIYAL (U.K.)

Dean, Academics

CBCS SCHEME

USN | | | | | | | | | | |

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

$$P(B/A) = P(A/B) \cdot P(B) / P(A) \qquad [05\ \text{Marks}]$$

[Defn - 2M, Proof - 3M]

Ans: Let us consider two events x and y of a random experiment.

The conditional probability, $P[Y/X]$ refers to the probability of Y occuring, given that X has already occured. It is defined as,

$$P[Y/X] = \frac{P[X,Y]}{P[X]} \quad \text{—} (1)$$

where $P[X,Y]$ = Joint Probability of events X, Y

$P[X]$ = probability of event X

$\therefore$ Conditional probability is defined as the ratio of Joint probability to that of probability of known event.

To prove, $$P[B/A] = \frac{P(A/B) \cdot P(B)}{P(A)}$$

Proof: We know that, the conditional probability,

$$P[B/A] = \frac{P[A,B]}{P[B]} \quad \text{—} (1)$$

$$\therefore P[A,B] = P[B/A]\, P[B] \quad \text{—} (2)$$

Similarly, $$P[A/B] = \frac{P[A,B]}{P[B]} \quad \text{—} (3)$$

$$\therefore P[A,B] = P[A/B] \cdot P[B] \quad \text{—} (4)$$

By comparing eqn (3) & (4),

Dk

$$P[B/A] \cdot P[A] = P[A/B]\, P[B]$$

$$\therefore P[B/A] = \frac{P[A/B]\, P[B]}{P[A]}$$

The above equation is called as Bayi's theorem.

1) b) Define Autocorrelation and cross correlation. Discuss the properties of Auto correlation. [10 Marks]
[Defn - 4M Properties - 6M]

Soln: The Auto correlation function of the random variable 'x' is defined as the expectation of the product of two random variables, $x(k)$ and $x(l)$, obtained by observing the random variable 'x' at times 'k' and 'l' respectively.

It is denoted by $r_x(k,l)$ or $r_x(\tau)$

$$\therefore r_x(k,l) = E[x(k) \cdot x(l)] = r_x(k-l)$$

Properties of correlation :-

Let $r_x(\tau)$ be the auto correlation function of Random process 'x' then,

i) $r_x(\tau) = r_x(k-l)$

ii) $r_x(0) = E[x^2]$; when $k=l$

iii) $r_x(\tau)$ is maximum value $\tau=0$

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

iv) $r_x(\tau)$ is a even function of τ

$$r_x(\tau) = r_x(-\tau)$$

Cross Correlation function →

Consider two random variables x and y, observed at time instants 'K' and 'l' respectively, then the cross correlation function between random variables x and y is given by,

$$r_{xy}(K, l) = E[x(k) \cdot y(l)]$$

or

$$r_{yx}(l, K) = E[y(l), x(K)]$$

∴ Cross correlation function gives the correlation between two different random processes

1)c) Develop a program to generate the probability density function of Guassian Distribution Function.

[05 Marks]

[Program - 5M]

Soln:

```
% Parameters
mu = 0;
Sigma = 1;

% Define range of x values
x = linspace (-5, 5, 1000);
% Compute PDF using the normal distribution formula
Pdf = (1/ (sqrt (2*pi)* sigma)) * exp (-(x-mu)^2/
                                         (2 * sigma^2));

% Plot the PDF
plot (x, Pdf, 'b', 'Line width', 2);
title (' Probability Density Function of Guassian
          Distribution');
xlabel('x');
ylabel('PDF');
grid on;
```

2) a) Define Auto-covariance, random variable, cumulative distribution function and Probability distribution function.

[Each Definition – 2M] [08 Marks]

Ans: Auto covariance:

Auto covariance of a sequence expresses the linear statistical dependencies between its samples. It is defined for a real valued signal with a lag of m samples

$$Cov(X, Y) = \lambda_{xy} = E[(X - m_x)(Y - m_y)]$$

Random variables:

Random variables is a real valued function, which can take any value from the sample space and its range is a set of real numbers.

Cumulative Distribution function:

A Cumulative distribution function of a real valued random variable X is the function given by

$$F_X(x) = P(X \leq x)$$

where right hand side represents the probability that the random variable x takes on a value less than or equal to x.

Probability Distribution function:-

For any random variable x, Probability distribution function is denoted by $F_X(x)$.

It is related to probability mass function as,

$$F_X(x) = P[x \leq x]$$

2) b) The variable its plot is given by, $f_X(x) = 2e^{-2x}$ for $x \geq 0$. Find the probability that it will take value between 1 and 3. [04 Marks]

[Each value - 2M]

<u>Soln:</u>

$$f_X(x) = 2e^{-2x}$$

$$P[x > 1] = F_X(1)$$

$$= 2e^{-2} = \cancel{20}\ 0.02$$ — 2M

$$P[1 < x < 3] = F_X(3) - F_X(1)$$

$$= 2e^{-6} - 2e^{-2}$$ — 2M

$$= 0.000002 - 0.02$$

$$= -0.01998$$

2) c) Define probability with an example. Discuss the Properties

[08 Marks]

[Defn – 2M, Properties – 3M, Ex – 3M]

Ans: Probability of event A, is denoted by P(A).

Let a random experiment is repeated n-times. If the event A occurs n_A times, then the probability of event A, i.e P(A) is defined as,

$$P(A) = \lim_{n \to \infty} \left(\frac{n_A}{n}\right)$$

∴ The ratio $\frac{n_A}{n}$ represents the fraction of occurrence of event A.

Axioms

1) $P(A) \geq 0$
2) $P(S) = 1$
3) If A & B are dependent events, then

$$P(A+B) = P(A) + P(B) - P(A \cdot B)$$

where $P(A \cdot B) \rightarrow$ Joint probability of events A, B

Example: 1) A coin is thrown three times, what is the probability that atleast one head is obtained?

Soln: Sample space → 3 coins, each with two Possibilities → H & T.

{HHH, HHT, HTH, HTT, THH, HTT, TTT}

Total no of outcomes – 8

Atleast one head – 7

∴ $P(A) = 7/8$

Module 2

3) a) Explain amplitude modulation with necessary equations and sketches in time domain and frequency domain [Explaination - 4M, Waveforms - 4M] [08 Marks]

Ans: Amplitude modulation is the process of altering the amplitude of carrier signal in accordance with the instantaneous values of message signal by keeping frequency and phase of carrier signal constant.

$$S(t) = A_c\left[1 + K_a m(t)\right]\cos 2\pi f_c t \quad \text{— (1)}$$

Expression for AM signal

The instantaneous value of message signal is given by,

$$m(t) = A_m \cos(2\pi f_m t) \quad \text{—— (2)}$$

The instantaneous value of carrier signal is given by,

$$c(t) = A_c \cos(2\pi f_c t) \quad \text{—— (2)}$$

Substituting m(t) in eqn (1) we get,

$$S(t) = A_c\left[1 + K_a \cdot A_m \cos(2\pi f_m t)\right]\cos(2\pi f_c t)$$

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

where $\mu = K_a A_m \Rightarrow$ Modulation Index.

$$S(t) = \left[A_c + \mu A_c \cos(2\pi f_m t)\right]\cos 2\pi f_c t$$

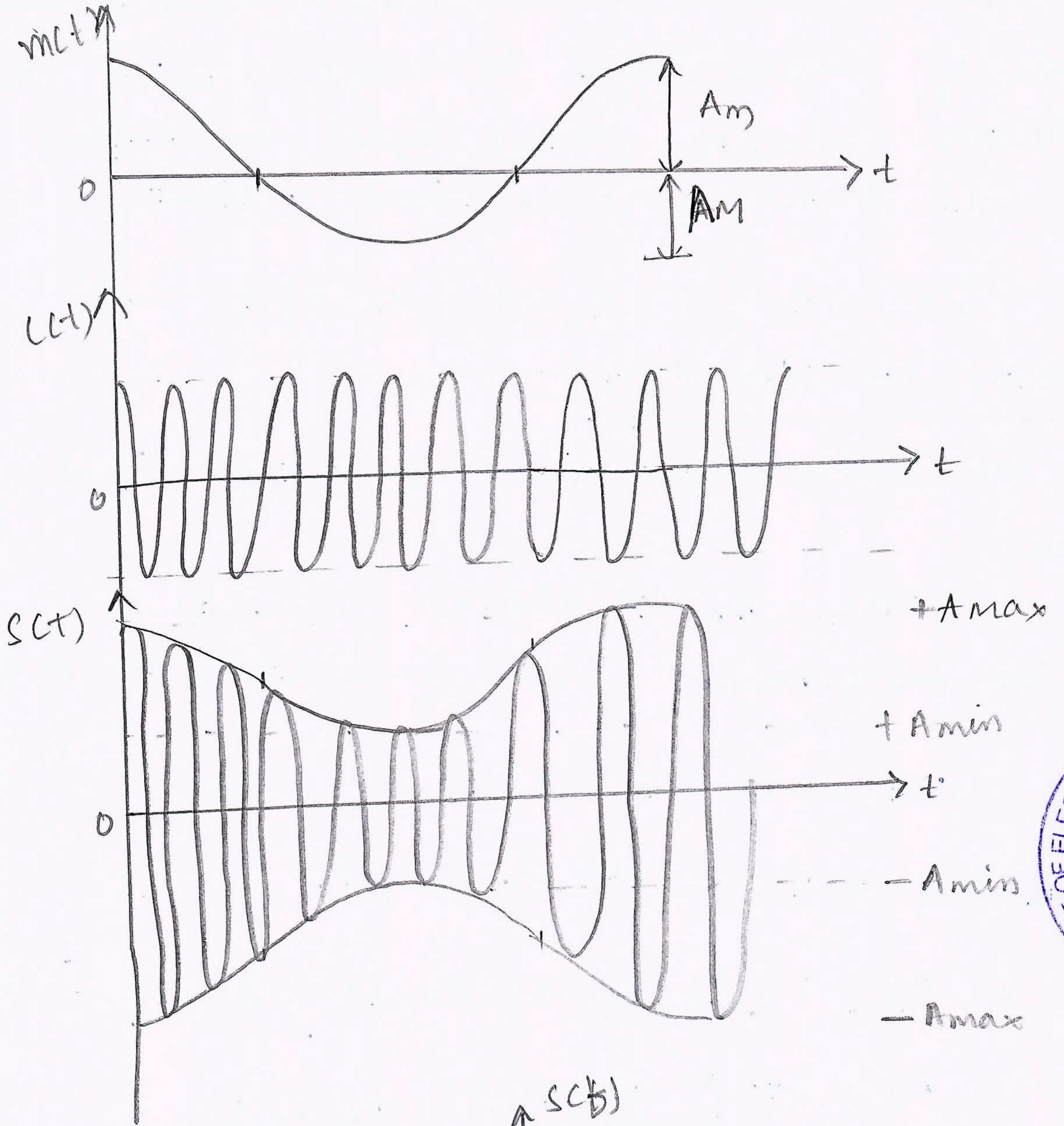
$$= A_c \cos 2\pi f_c t + \mu A_c \cos 2\pi f_m t \cos 2\pi f_c t$$

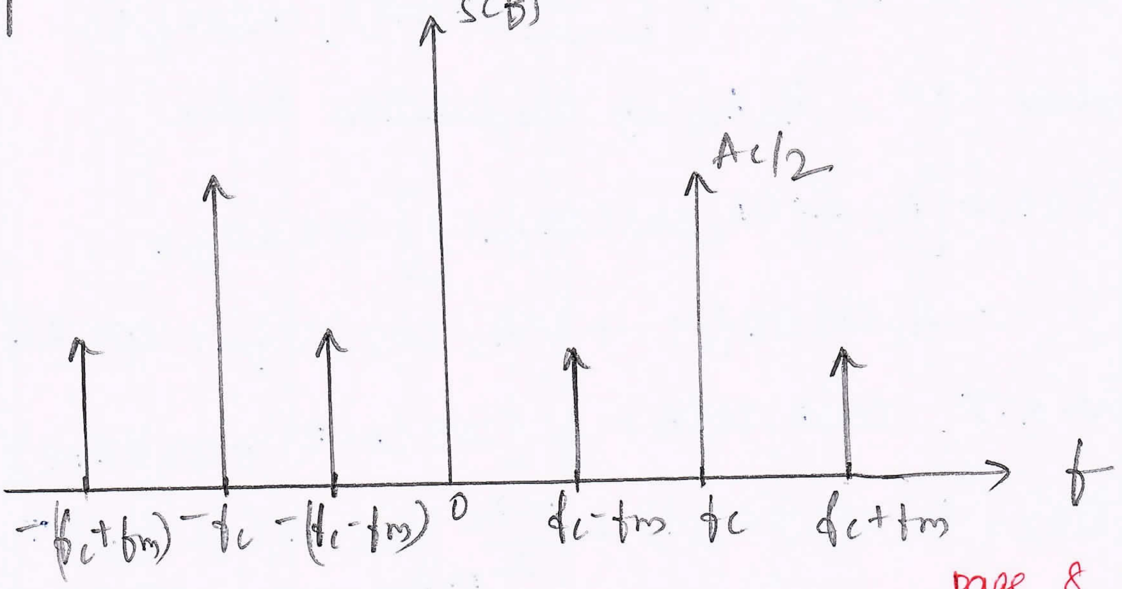
W.K.T $\cos A \cdot \cos B = \frac{1}{2}\left[\cos(A-B) + \cos(A+B)\right]$

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

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

Wave forms in time domain and frequency domain





PL

3) b) Define modulation index and percentage of modulation. Explain ~~mo~~ over modulation and distortion.
[Defn - 3M , Expln - 3M] [06 Marks]

Ans: The ratio of message signal amplitude to that of unmodulated carrier signal amplitude is called '**Modulation index**"

$\therefore$ Modulation index , $\mu = \frac{A_m}{A_c}$

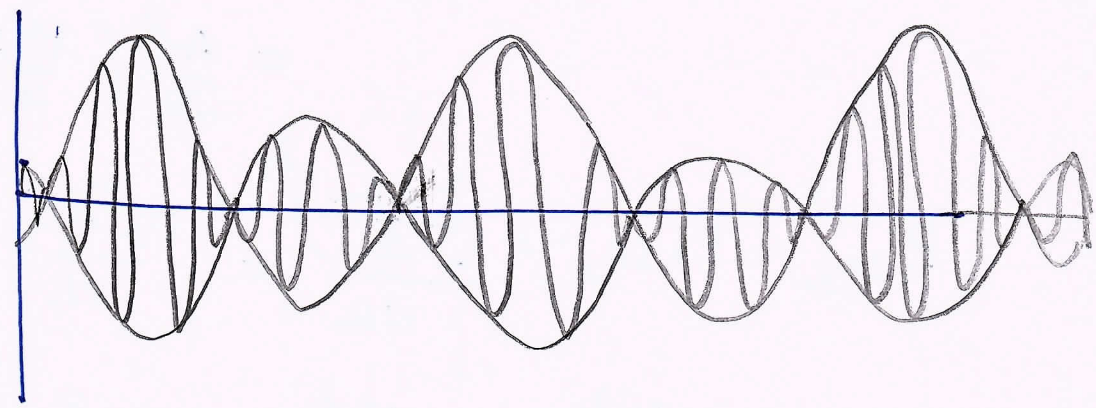
Percentage of Modulation index is,

$$\% \mu = \frac{A_m}{A_c} \times 100 \quad \text{or} \quad \% \mu = k_a \cdot A_m \times 100$$

→ If A_m is greater than A_c, modulation index μ becomes greater than 1, then distortion is introduced into the system.

Over Modulation :-

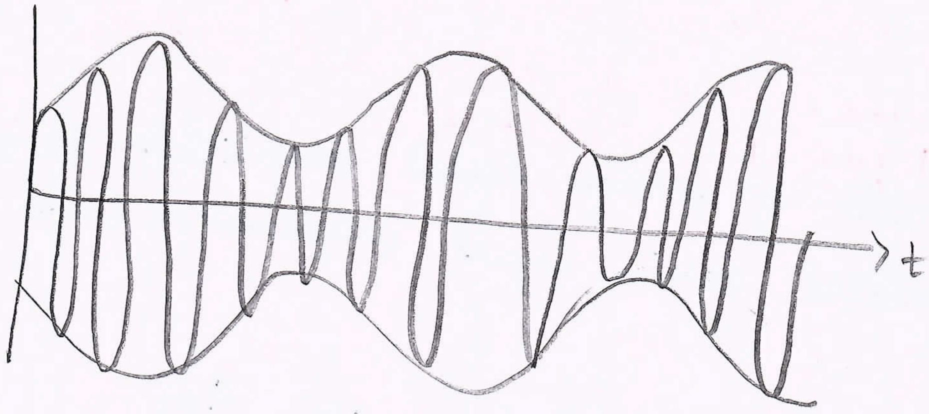
If the value of the modulation index is greater than 1, then the wave will be an over modula-ted wave,



Distortion

If the value of modulation index is less than 1, then the modulated o/p would look like the figure below.

DL



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

[Derivation – 6M]

[06 Marks]

Soln: Let P_T, be the total power in the AM signal. The multi tone AM signal for two modulating signals contains five different frequency components. Its total power is calculated as follows.

$$P_T = P_c + P_{LSB_1} + P_{USB_1} + P_{USB_2} + P_{LSB_2} \quad \text{--- (1)}$$

$$P_c = \frac{A_c^2}{2R}$$

$$P_{LSB_1} = P_{USB_1} = \frac{\left(\frac{\mu_1 A_c}{2}\right)^2}{2R} = \frac{\mu_1^2 A_c^2}{8R}$$

$$= \frac{A_c^2}{2R}\left(\frac{\mu_1}{4}\right) = P_c \frac{\mu_1^2}{4}$$

$$P_{LSB_2} = P_{USB_2} = \frac{\left(\frac{\mu_2 A_c}{2}\right)^2}{2R} = \frac{\mu_2^2 A_c^2}{8R} = \frac{A_c^2}{2R}\left(\frac{\mu_2^2}{4}\right) = P_c \frac{\mu_2^2}{4}$$

$$P_T = P_c + P_c \frac{\mu_1^2}{4} + P_c \frac{\mu_1^2}{4} + P_c \frac{\mu_2^2}{4} + P_c \frac{\mu_2^2}{4}$$

$$P_T = P_c \left[1 + \frac{\mu_1^2}{4} + \frac{\mu_1^2}{4} + \frac{\mu_2^2}{4} + \frac{\mu_2^2}{4} \right]$$

$$P_T = P_c\left[1+\frac{\mu_1^2}{2}+\frac{\mu_2^2}{2}\right]$$

$$= P_c\left[1+\frac{(\mu_1^2+\mu_2^2)}{2}\right]$$

$$\boxed{P_T = P_c\left[1+\frac{\mu_t^2}{2}\right]}$$

4) a) Explain a general block diagram of a frequency division multiplexing? [06 Marks]

[Block dgm – 3M, expln – 03M]

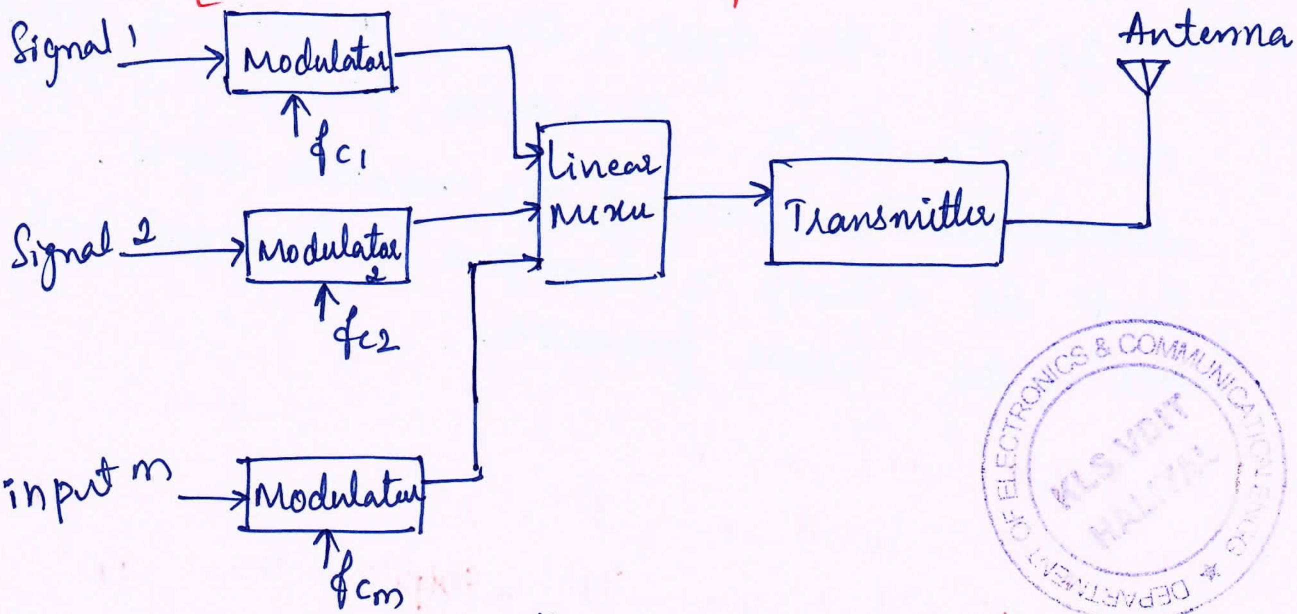


Fig (i) Transmitter

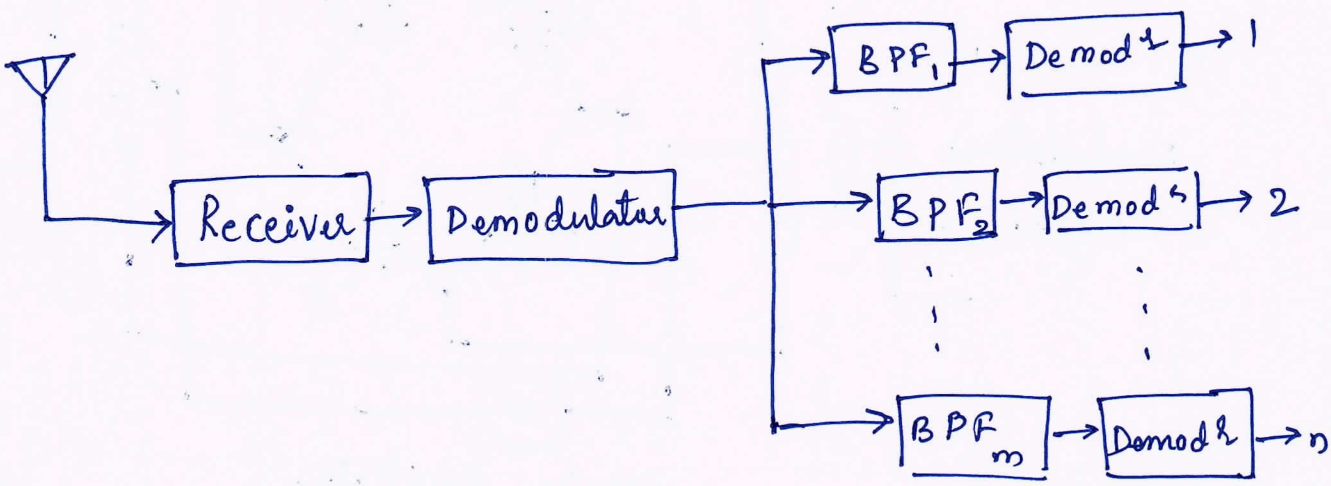


Fig (ii) Receiver

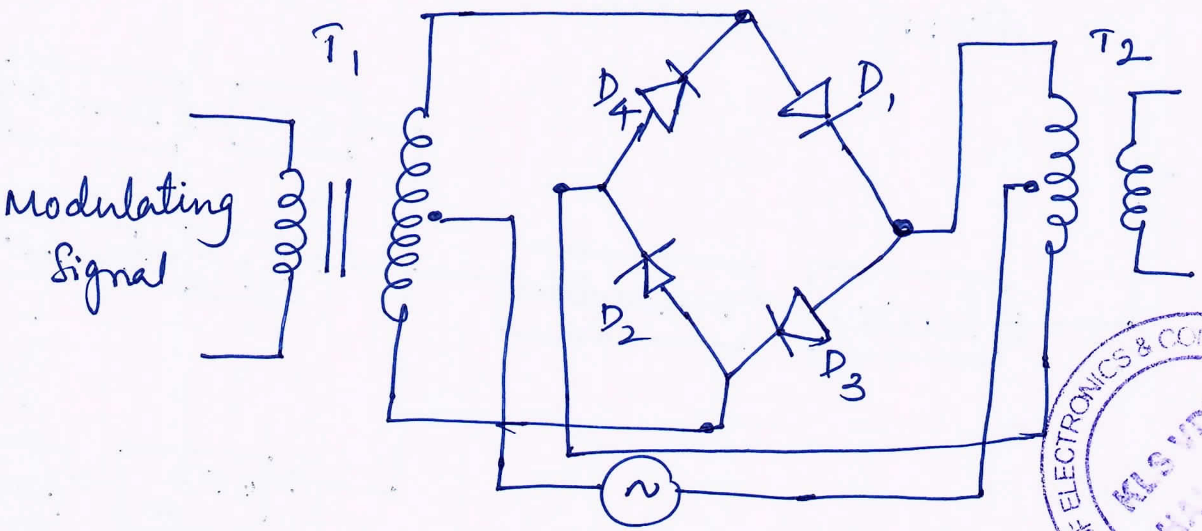
Fig (i) shows a block diagram of an FDM system, each signal to be transmitted feeds a modulator circuit. The carrier for each modulator (f_c) is on a different frequency. The carrier frequencies are usually equally spaced from one another over a specific range. These carriers are referred to as subcarriers.

→ The FDM process divides up to the bandwidth of the single channel into ~~splitter~~ smaller, equally spaced channels each capable of carrying information in sidebands.

In fig (ii) the system shows a receiving portion of an FDM system. A receiver picks up a signal and demodulates it, recovering the composite signal. This is sent to a group of bandpass filters, each centred on one of the carrier frequencies.

4)b) Explain the working principle of lattice type Balanced modulator with circuit diagram. [07 Marks]

Ans: [Circuit Diagram - 3M , Explaination - 4M]



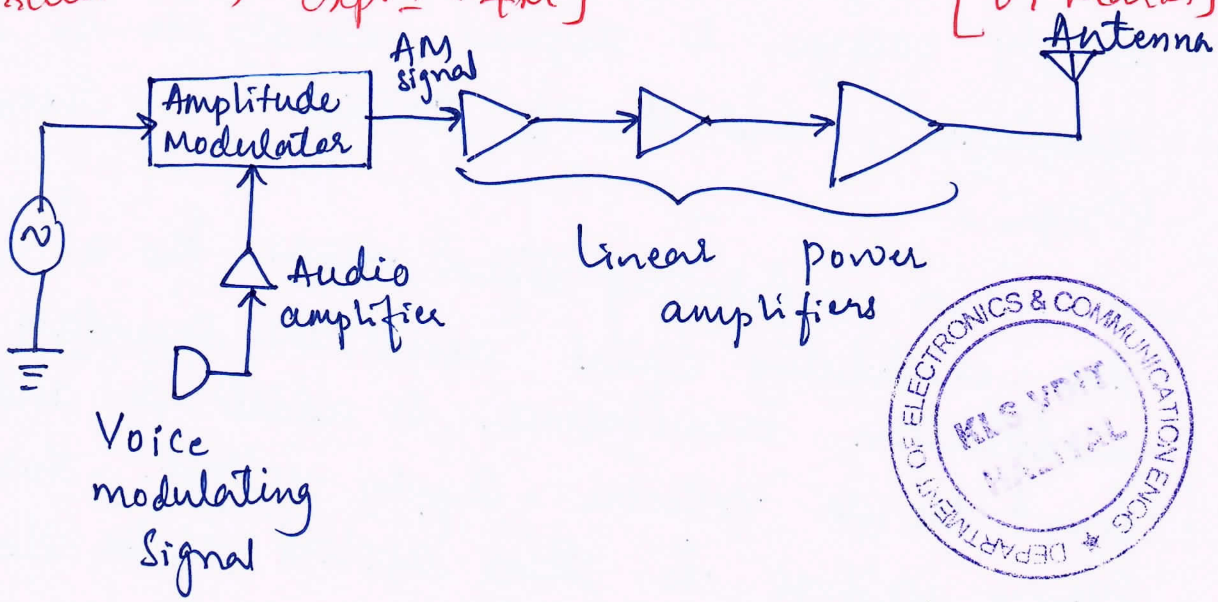
Carrier Oscillator

One of the most popular and widely used balanced modulator is the diode ring or lattice modulator as shown in the figure above.

It consists of an i/p transformer T_1, an o/p transformer T_2 and four diodes connected in a bridge circuit. The carrier signal is applied to the centre taps of the i/p & o/p transformers, and the modulating signal is applied to the input transformer T_1. The o/p appears across the secondary of the o/p transformer T_2.

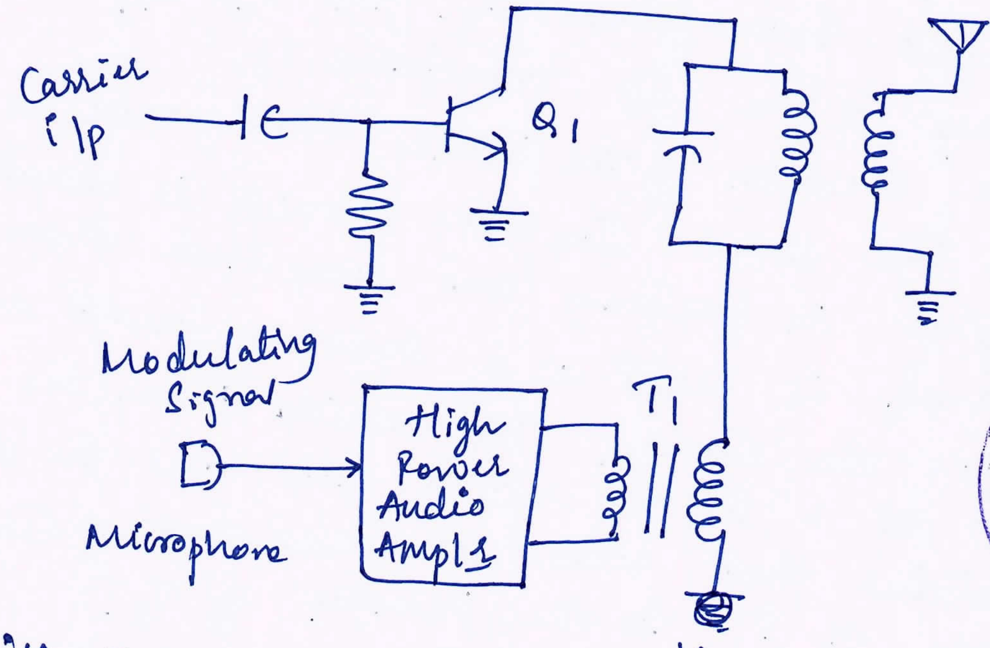
The operation of the lattice transformer is relatively simple. The carrier sine wave which is usually considerably higher in frequency and amplitude than the modulating signal, is used as a source of forward and reverse bias for the diodes. The carriers turns the diodes off and on at a higher rate of speed, and the diodes act as switches that connect the modulating signal at the secondary of T_1 to the primary of T_2.

4)c) With neat diagrams, explain high level collector modulator [Circuit - 3M, Expln - 4M] [07 Marks]



The above figure shows a collector modulator. In this, the o/p stage of the transmitter is a high power class c amplifier. class c amplifiers conduct for only a portion of the positive half cycle of their input signal.

The collector current pulses cause the tuned circu to oscillate at the desired o/p frequency. The tuned ci therefore, reproduces the negative portion of the carrie signal.

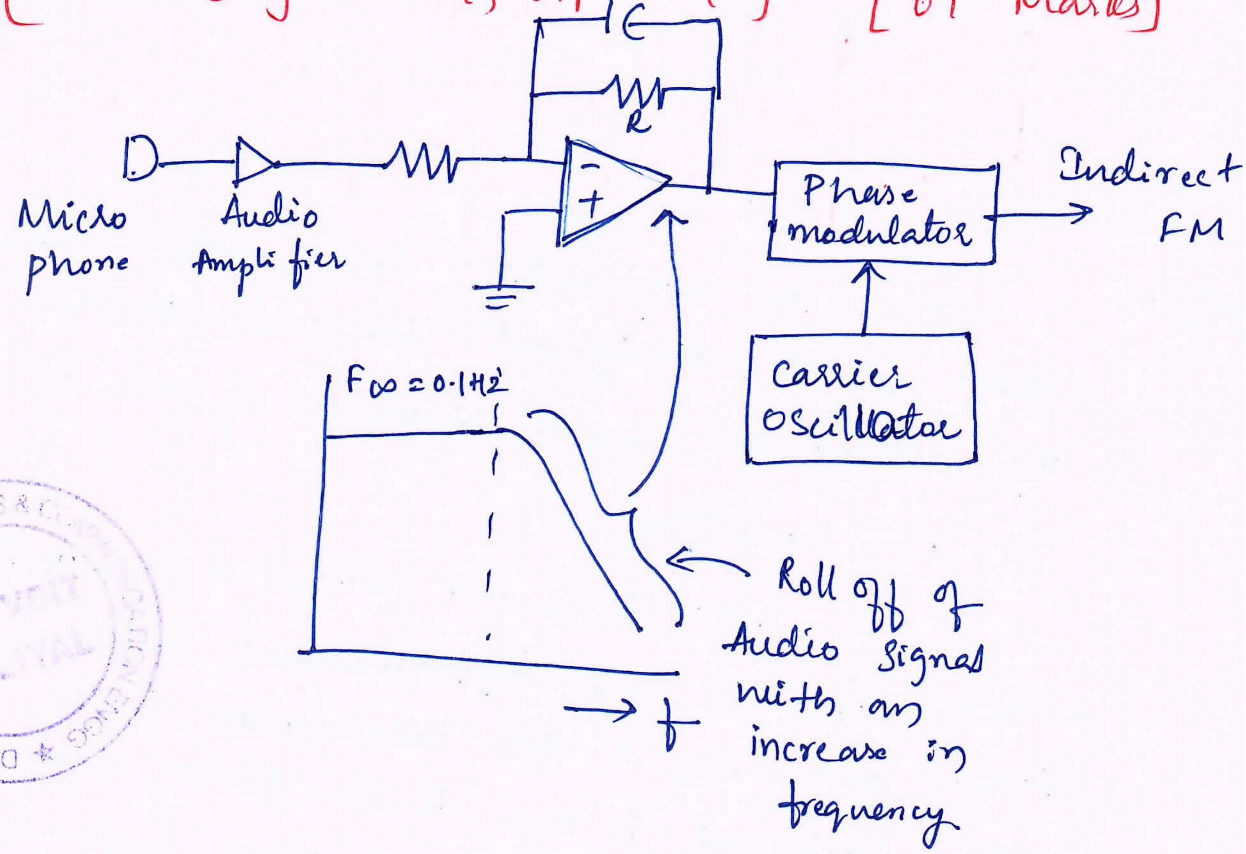


With zero modulation signal, there is zero modula-tion voltage across the secondary of T_1, the collector supply voltage is applied directly to the class C amplifier, and the o/p carrier is a steady sine wave.

→ When 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 amplifie causing the amplitude of the carrier wave varies in acco-rdance with the modulated signal.

5)a) With a neat diagrams, explain converting a phase modulated signal into a frequency modulated signal
[Circuit diagram – 3M, Expⁿc – 4M] [07 Marks]

Ans:



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 shown above. This low pass filter is called a frequency correcting network, predistorter or 1/f filter, causes the higher modulating frequencies to be attenuated. Higher modulating frequencies produce a great rate of change and thus a greater frequency deviation, this is offset by the lower deviation.

The predistorter compensates for the excess frequency deviation caused by higher modulating frequencies. The result is an o/p that is the same as an FM signal. The FM produced by a phase modulator is called 'Indirect FM'

5) 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)
[Exprn derivation – 6M] [06 Marks]

Soln:

Given the modulation index, the number and amplitudes of the significant sidebands can be determined by solving the basic equation of an FM signal.

The FM equation,

$$V_{FM} = V_c \sin(2\pi f_c t + m_f \sin 2\pi f_m t)$$

where, V_{FM} is the instantaneous value of the FM signal and m_f is the modulation index. The term whose coefficient is m_f is the phase angle of the carrier. This equation expresses the phase angle in terms of the sine wave modulating signal.

This equation is solved with a complex mathematical process known as Bessel functions.

$$V_{FM} = V_c \{ J_0 (\sin \omega_c t) + J_1 [\sin(\omega_c + \omega_m)t - \sin(\omega_c - \omega_m)t]$$
$$+ J_2 [\sin(\omega_c + 2\omega_m)t + \sin(\omega_c - 2\omega_m)t]$$
$$+ J_3 [\sin(\omega_c + 3\omega_m)t + \sin(\omega_c - 3\omega_m)t]$$
$$+ J_4 [\sin(\omega_c + 4\omega_m)t + \sin(\omega_c + 4\omega_m)t]$$
$$+ \ldots$$

where $\omega_c = 2\pi f_c$ = carrier frequency

$\omega_m = 2\pi f_m$ = modulating signal frequency

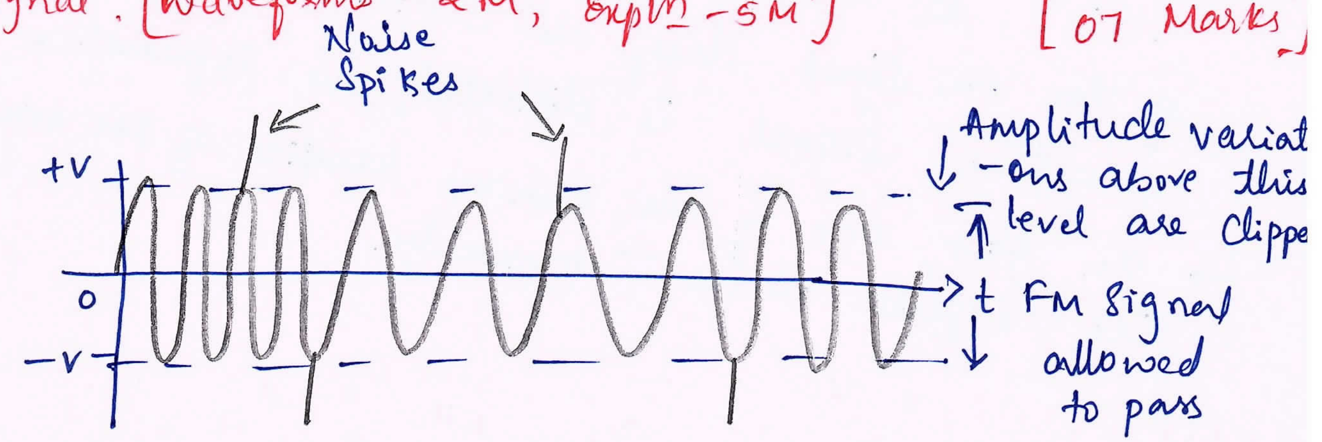
V_c = Peak value of unmodulated carrier

The amplitudes of the sidebands are determined by J_n Co-efficients, which are in turn, determined by value of the modulation index,

$$J_n(m_f) = \left(\frac{m_f}{2^n n!}\right)^n \left[1 - \frac{(m_f)^2}{2(2n+2)} + \frac{(m_f)^4}{2\cdot 4(2n+2)(2n+4)} - \frac{(m_f)^6}{2\cdot 4\cdot 6(2n+2)(2n+4)(2n+6)} + \cdots\right.$$

5) c) Identify the noise suppression of frequency modulated signal. [Waveforms - 2M, Expln - 5M] [07 Marks]

Ans:



Noise is interference signal generated by lightening motors, automotive ignition systems and any power line switching that produces transient signals. Such noise is typically narrow spikes of voltage with very high frequencies.

They add to a signal and interfere with it. The potential effect of such noise on an FM signal is shown in the above waveforms. If the noise signal were strong enought, they could completely obliterate the information signal.

FM signals, however have a constant modulated carrier amplitude, and FM receivers contain limiter circuits that deliberately restrict the amplitude of the received signal. Any amplitude variations occuring on the FM signal are effectively clipped off.

→ This does not affect the information content of the FM signal, since it is contained solely with the frequency variations of the carrier.

→ Because of the clipping action of the limiter circuit noise is almost completely eliminated. Even if the peaks of the FM signal itself are clipped or flattened and the resulting signal is distorted, no information is lost.

→ In fact, one of the primary benefits of FM over AM is its superior noise immunity.

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 Marks]
[Each value - 2M]

Soln: i) $m_f = \frac{f_d}{f_m} = \frac{30\,KHz}{5\,KHz} = 6$

No. of sidebands = 9

$BW = 2 f_m N = 2 \times 5\,KHz \times 9 = 90\,KHz$

ii) Using Carson's rule

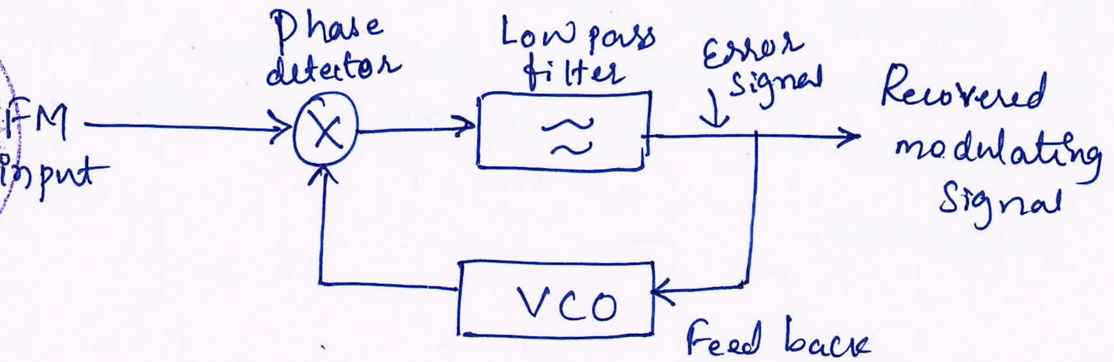
$BW = 2\,[f_{d(max)} + f_{m(max)}]$

$= 2\,[30\,KHz + 5\,KHz] = 2\,(35\,KHz)$

$BW = 70\,KHz$

6)b) Define Phase Locked Loop (PLL). Explain with neat diagram of FM demodulator using the IC 555.
[Defn - 2M, Block diagram - 2M, expln - 4M]
[08 Marks]

Ans: A phase-locked loop (PLL) is a frequency - or phase sensitive feedback control circuit used in frequency demodulation, freq synthesizers and various filters and signal detection applications. All phase locked loop have the three basic elements as shown below



i) A phase detector is used to compare the FM input, sometimes referred to as the reference signal to the O/p of a VCO.

ii) A VCO frequency is varied by the dc o/p voltage from a low pass filter

iii) The low pass filter smoothes the O/p 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 o/p 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 O/p varies in proportion to the difference.

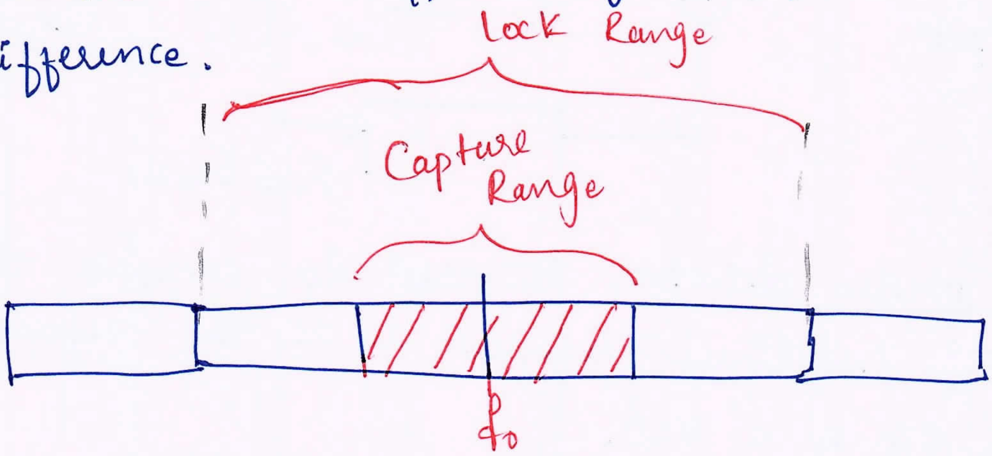
The filtered O/p adjusts the VCO frequency in an attempt to correct for the original frequency or phase difference.

Dr

This dc control voltage, called the error signal, is also the feedback in this circuit.

→ When no i/p signal is applied, the phase detector and low pass filter o/ps are zero. The VCO then operates at what is called the free running frequency, its normal operating frequency as determined by internal frequency determining components.

→ When an i/p signal close to the frequency of the VCO is applied, the phase detector compares the VCO free-running frequency to the input frequency and produces an o/p voltage proportional to the freq difference.

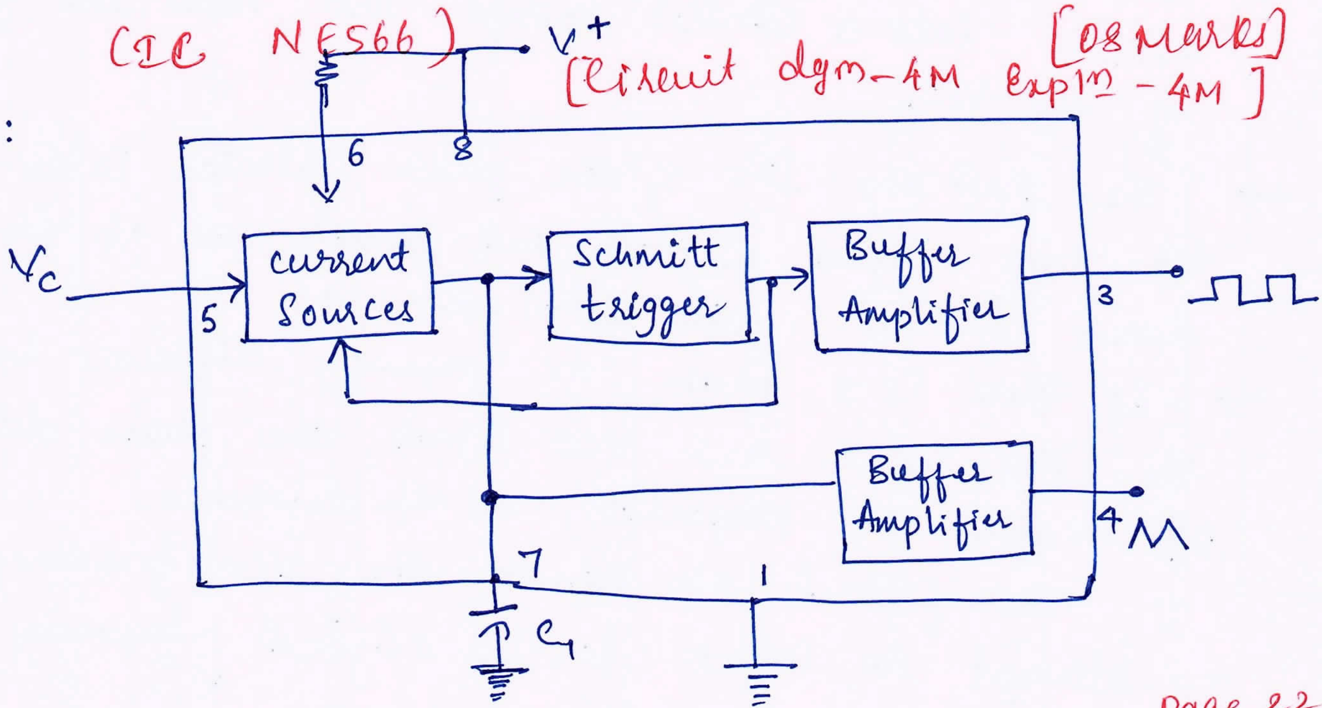


f_0 = VCO free running frequency

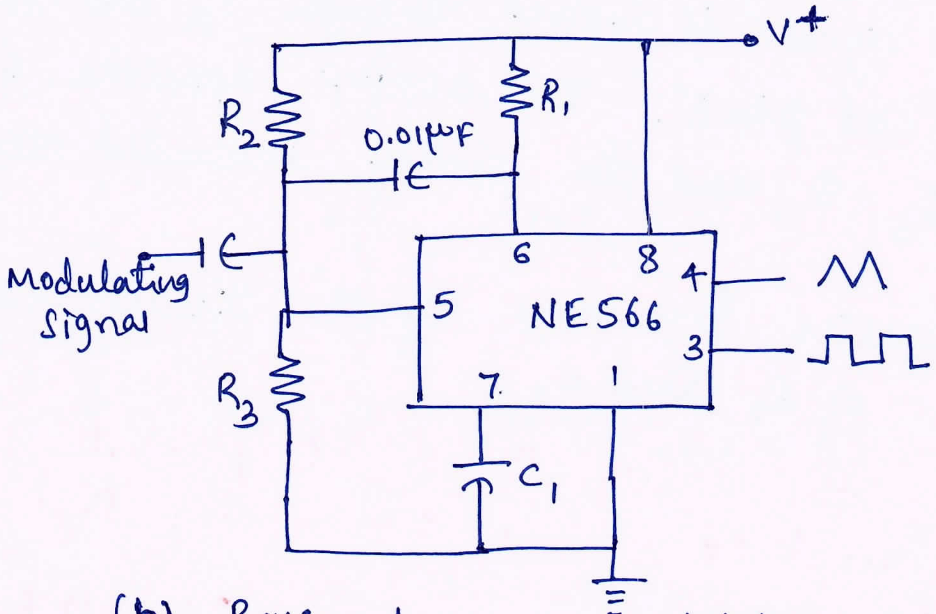
6) c) With neat diagram, explain the concept of frequency modulation with an IC voltage controlled oscillator. (IC NE566)

[08 Marks]
[Circuit dgm - 4M Expln - 4M]

Ans:



(a) Block diagram with an IC VCO.



(b) Basic frequency modulator using the NE566 VCO.

→ The above figure is a block diagram of one widely used IC VCO, the popular NE566. External resistor R_1 at pin 6 sets the value of current produced by the internal current sources. The current sources linearly charge and discharge external capacitor C_1 at pin 7.

→ An external voltage V_c applied at pin 5 is used to vary the amount of current produced by the current sources.

→ The Schmitt trigger circuit is a level detector that controls the current source by switching between charging and discharging when the capacitor charges or discharges to a specific voltage level.

→ A linear sawtooth of voltage is developed across the capacitor by the current source. This is buffered by an amplifier and made available at pin 4.

→ The Schmitt trigger o/p is a square wave at the same frequency available at pin 3.

The complete frequency modulator circuit using the NE566 is shown in fig(b). The current sources are biased with a voltage divider made up of R_2 and R_3.

The modulating signal is applied through C_2 ... Voltage divider at pin 5. The $0.001\mu F$ capacitor betn pins 5 and 6 is used to prevent unwanted oscillati -ons.

→ The center carrier frequency of the circuit is set by the values of R_1 and C_1.

)a) Why digitize the analog signals? Explain the differ-ent processes used to convert the analog signal to digital signal. [06 Marks]
[Digitize-3M, ADC-3M]

Soln:- The advantages of using digital signals are:-

a) The effects of distortion, Noise and Interference is much less in digital signals.

b) Discrete circuits are more reliable, easier to design & cheaper than analog circuits

c) With digital systems, it is easier to integrate different services

Ex: Video + Soundtrack

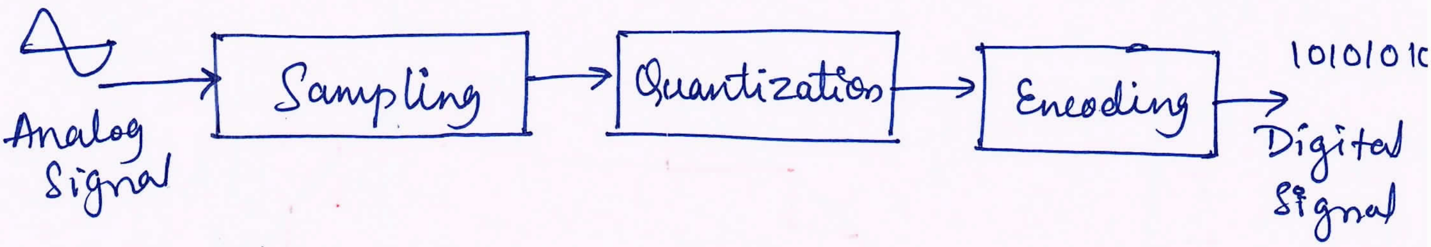
d) Digital signals are simpler to characterize and typically do not have the same amplitude range & variability as analog signals.

e) Digital circuits are less sensitive to physical effects such as vibration and Temperature.

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

g) The transmission scheme is relatively independent of the source.

PL



The Key steps involved in the conversion of Analog signal to Digital Signal are:

i) Sampling ii) Quantization iii) Encoding

* The low pass filter prior to sampling is included to prevent aliasing of the message signal.
* The Quantization and encoding operations are usually performed in the same circuit, which is called an Analog to Digital converter.

7) b) What is Quantization process? Explain the different types of quantization with their important characteristics? [Defn - 2M, Types - 4M] [07 Marks]

Soln: The Process of transforming Sampled amplitude values of a message signal into a discrete amplitude values or discrete amplitude levels is called as Quantization.

$x(nT_s) \rightarrow$ Quantizer $\rightarrow x_q(nT_s)$

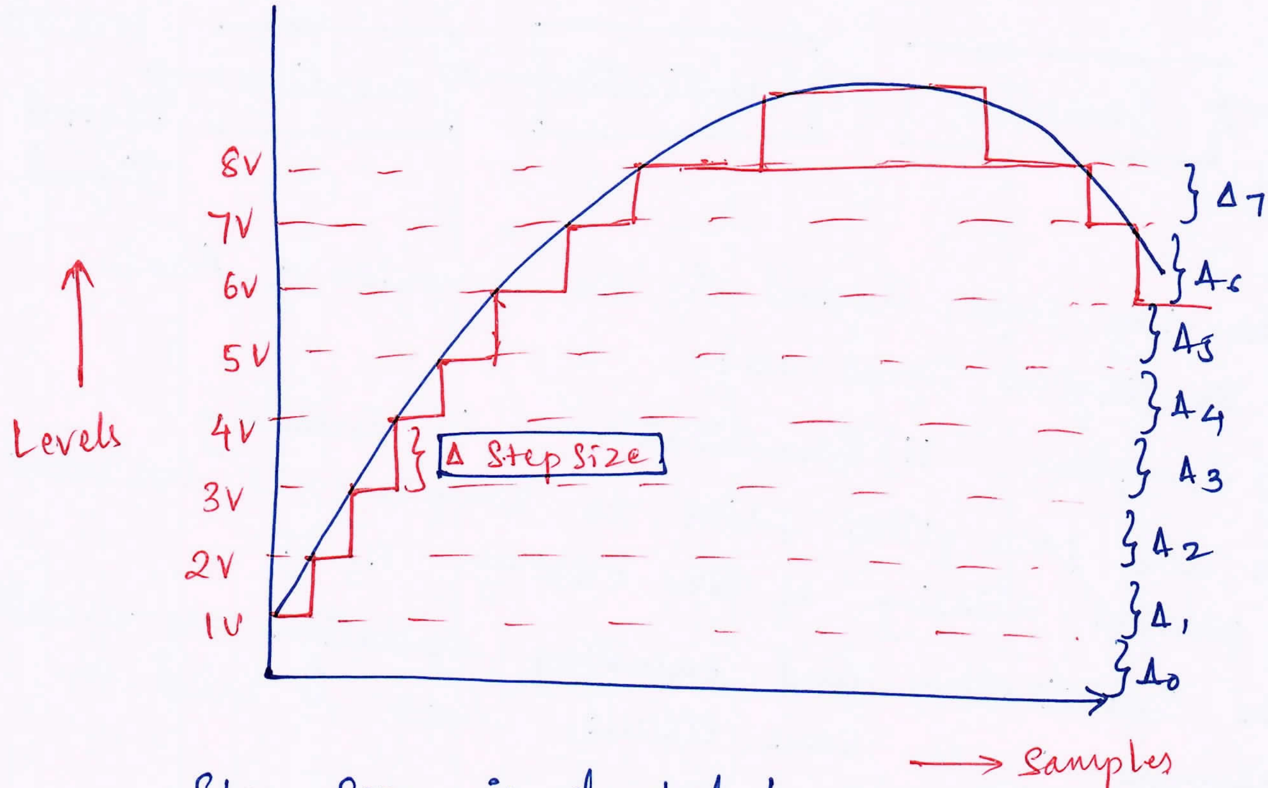
* Consider a continuous time signal $x(t)$, whose excursions are confined to the voltage range V_L to V_H as shown in figure below.

Let $x(nT_s)$ be the sampled version of $x(t)$.

* Let the no of Quantizer level be 8

$L = 2^R$

$8 = 2^3$ $\therefore R = 3 =$ No of bits



Step size is denoted by, Δ and is given by,

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

where $L = 2^R$

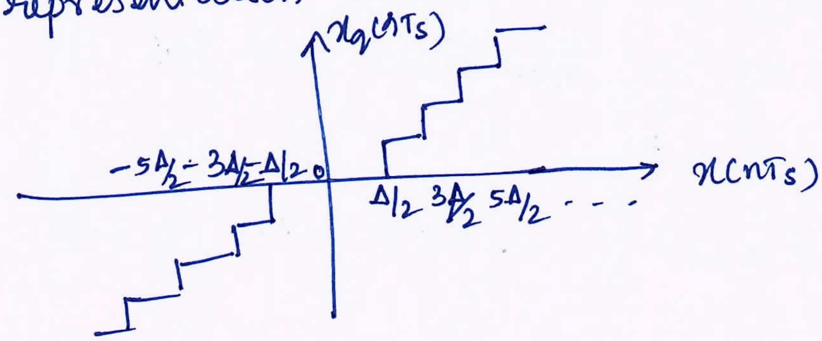
$$\therefore \text{ Step size} = \Delta = \frac{V_H - V_L}{2^R}$$

Types of quantization

1) Mid Tread type Quantization
2) Mid Rise " "

Mid Tread type quantization

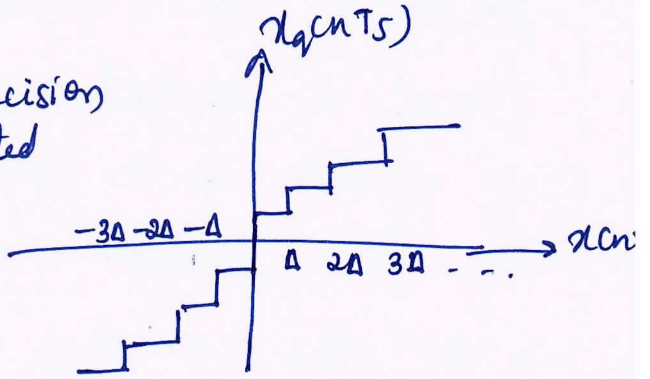
In this type of quantization, the decision threshold of the Quantizers are located at $\pm\frac{\Delta}{2}, \pm\frac{3\Delta}{2}, \pm\frac{5\Delta}{2}, \ldots$ and representation levels are located at $0, \pm\Delta, \pm 2\Delta, \ldots$



Mid rise type Quantizer

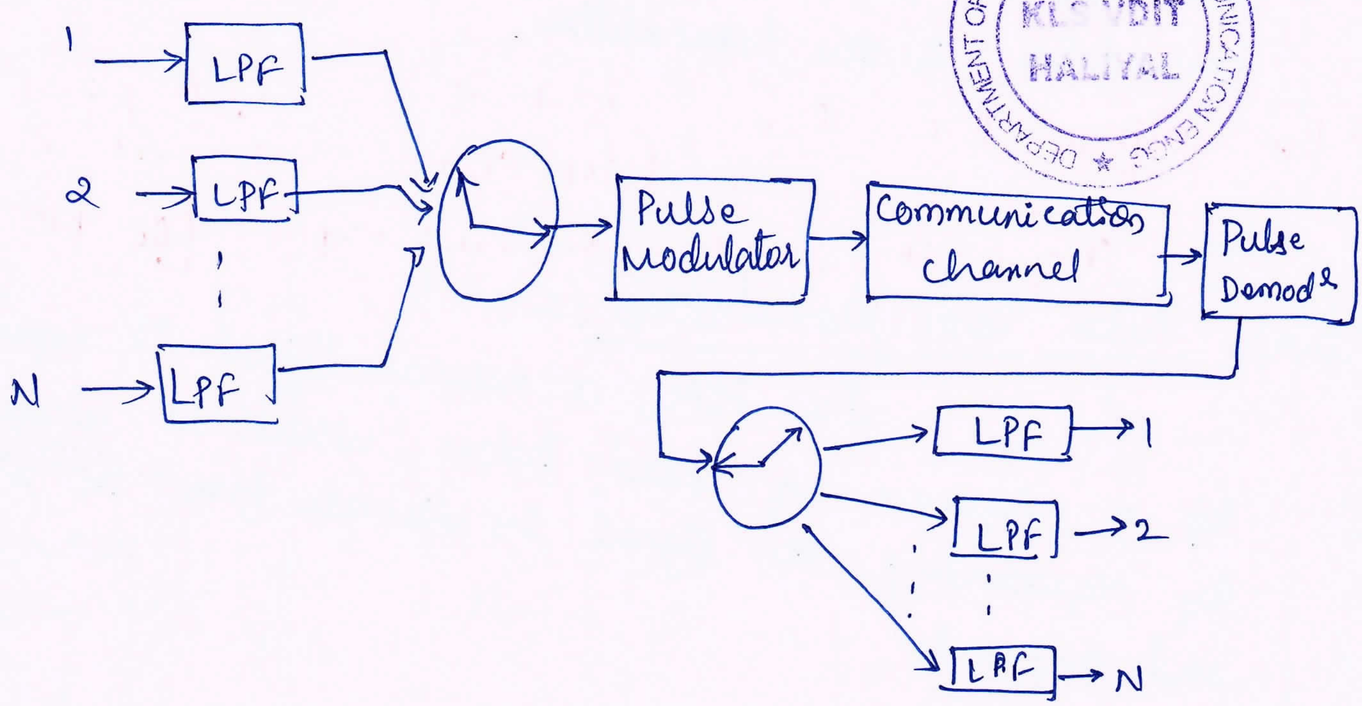
In mid riser quantizer, the decision threshold of the quantizer are located at $0, \pm\Delta, \pm 2\Delta, \pm 3\Delta \ldots$

Representation levels are at $\pm\Delta/2, \pm 3\Delta/2, \pm 5\Delta/2 \ldots$



7) c) Explain the concept of Time Division Multiplexing (TDM) with a neat block diagram.
[Block dgm - 3M, Expln - 4M] [07 Marks]

Soln: ~~In~~ ~~Time~~



In the above TDM block diagram, each input message signal is first restricted in bandwidth by a LPF to remove the frequencies that are not essential to an adequate signal representation.

→ The low pass filter o/ps are thus applied to commutator which can be implemented using electronic switching circuitry.

Following the commutation process, the multiplexed signal is applied to a pulse modulator, the purpose of which is to transform the multiplexed signal into a form suitable for transmission over a common channel.

→ At the receiving end of the system, the received signal is first applied to a pulse demodulator which performs the reverse operation of pulse modulator.

→ The narrow samples produced at the pulse demodulator are distributed to the appropriate LPF by means of decommutator, which operates in synchronism with the commutator in the transmitter.

8) a) Define PCM (Pulse Code Modulation). Explain the basic elements of a PCM system with neat diagrams.
[Defn - 2M, Block dgm - 2M, Expln - 3M] [06 Marks]

Soln: **Pulse Code Modulation :-**

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 and amplitude.

(a) Transmitter

(b) Transmission path

Final o/p of channel → Regeneration Circuit → Decoder → Reconstruction filter → Destinati

(c) Receiver

→ The basic operations performed in the transmitter a Sampling, Quantization and Encoding. The low pass fil Prior to Sampling is included to prevent aliasing of the message signal.

→ The Quantization and Encoding are performed in the same Circuit which is called as Analog to digita Converter.

→ The basic operations in the FM receiver are regeneration of impaired Signals using regeneration circui

→ Regeneration also occurs at the intermediate points along the transmission path of communication channel as shown

8) b) For the data stream 01101001, Draw the following line code waveforms.

i) Unipolar NRZ ii) Polar NRZ iii) Unipolar RZ

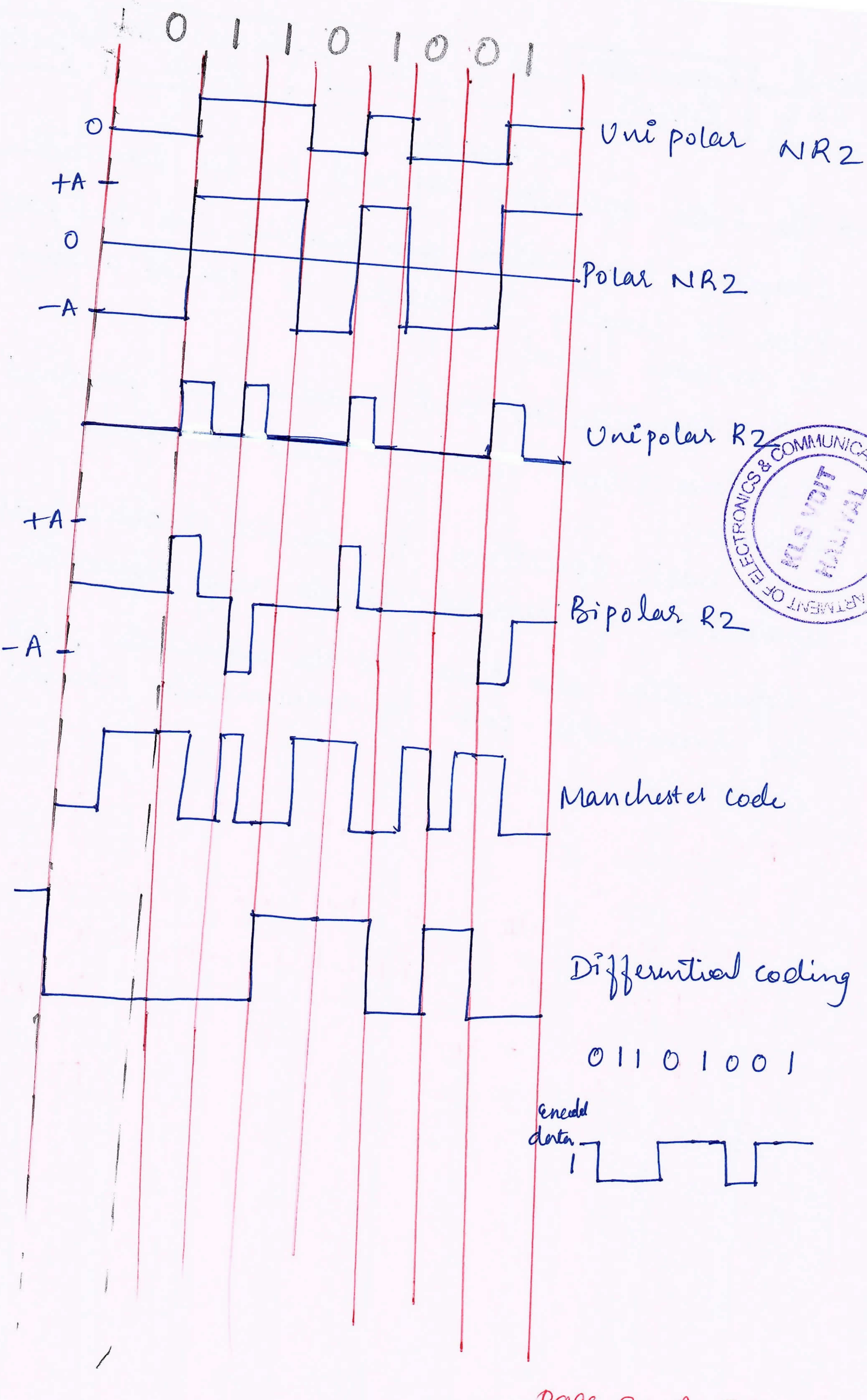
iv) Bipolar RZ v) Manchester code vi) Differential coding

[09 Marks]

[Waveforms – 9M]

(PTO)

DL


0 1 1 0 1 0 0 1
0
Uni polar NRZ
+A
0
-A
Polar NRZ
Unipolar RZ
+A
-A
Bipolar RZ
Manchester code
Differential coding
0 1 1 0 1 0 0 1
encoded data
1

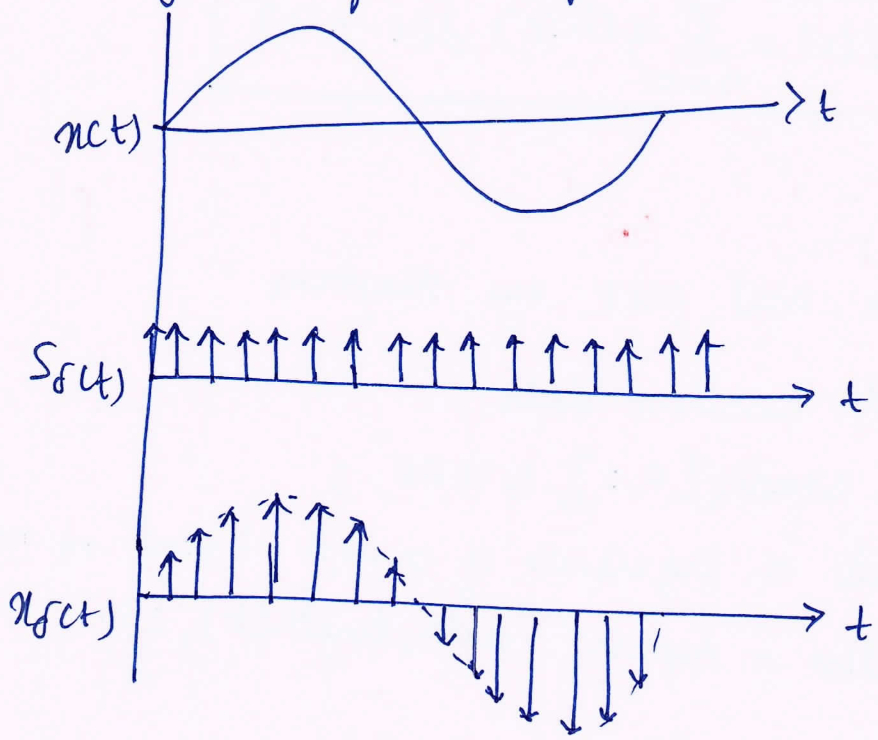
8)c) State and prove Sampling Theorem. Explain with neat sketches and equations.
[State - 2M, Proof - 2M, waveforms - 1M] [05 Marks]

Soln: Sampling theorem states that "Any continuous time base band signal can be completely represented by its samples and recovered from its samples if the sampling frequency (f_s) is greater than or equal to twice the highest frequency present in the base band signal.

i.e $f_s \geq 2W$

Proof: Consider an arbitrary signal $x(t)$ of finite energy which is specified for all time instants

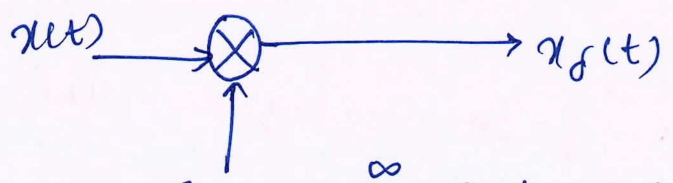
→ A segment of the signal $x(t)$ is shown below.



Let $x_\delta(t)$ denotes the signal obtained by multiplying $S_\delta(t)$ and instantaneous value of $x(t)$ at every 'T_s' seconds

∴ The sampled signal : $x_\delta(t)$ is given by

$$x_\delta(t) = x(t) \cdot S_\delta(t) \quad —①$$



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

$$S_\delta(t) = \sum_{n=-\infty}^{\infty} \delta(t - nT_s) \quad — ②$$

$$\therefore \quad x_\delta(t) = x(t) \cdot \sum_{n=-\infty}^{\infty} \delta(t - nT_s)$$

$$= \sum_{n=-\infty}^{\infty} x(t)\, \delta(t - nT_s)$$

using shifting property of impulse function

We know that,

$$x(t)\, \delta(t - nT_s) = x(nT_s)\, \delta(t - nT_s)$$

$$\therefore \quad \boxed{x_\delta(t) = \sum_{n=-\infty}^{\infty} x(nT_s)\, \delta(t - nT_s)}$$

9) a) Develop a code to generate and plot eye diagram [06 Marks]

[Program - 6M]

Soln:

```
% Generate and plot eye diagram
% Generate random data
  data = randi([0,1], 1, 1000);
% Generate a sequence of pulses based on NRZ
nrz_pulses = repmat (data, length(t), 1);
% Plot eye diagram
figure;
plot (t, nrz_pulses, 'b');
title ('Eye diagram (NRZ)');
xlabel ('Time (s)');
ylabel ('Amplitude');
grid on;
```

Q)b) Define noise factor and noise figure. Also explain noise in cascade connection
[Defn - 4M, Cascade - 2M] [06 Marks]

Soln: <u>Noise factor</u> of an any amplifier or network is defined as the ratio of signal to noise power ratio at the input and to that of signal to noise power at the output.

i.e Noise factor, $F = \dfrac{(S/N) \text{ Power ratio at the input}}{(S/N) \text{ Power ratio at the output}}$

$$F = \frac{(P_{Si} / P_{Ni})}{(P_{So} / P_{No})} = \frac{P_{Si}}{P_{Ni}} \times \frac{P_{No}}{P_{So}} \quad \text{--- (1)}$$

<u>Noise figure</u> : (F_{dB})

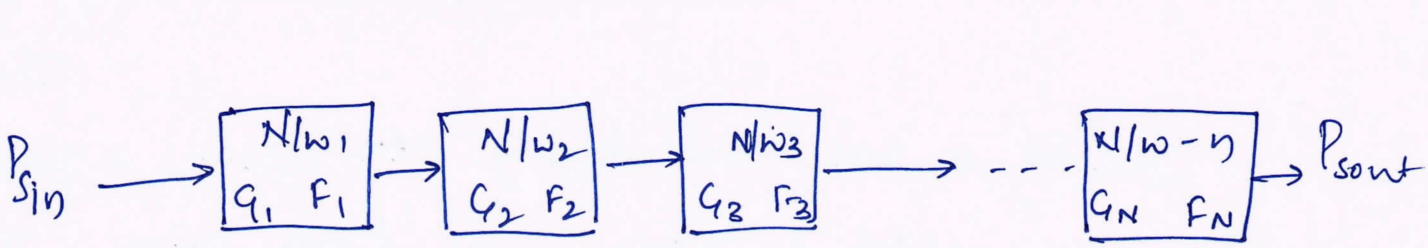
Noise factor, F expressed in dB, is known as Noise figure

$$\text{Noise figure} = 10 \log [F]$$

$$= 10 \log \left[\frac{(S/N)_{input}}{(S/N)_{output}} \right]$$

$$F_{dB} = 10 \log (S/N)_{input} - 10 \log (S/N)_{output}$$

<u>Noise in cascade connection</u> :-



where G_N is - Gain of network N

F_N - Noise factor of Network - N

The overall Noise factor, F for N-amplifiers connected in cascade is given by Friis Formula

$$F = F_1 + \frac{F_2 - 1}{G_1} + \frac{F_3 - 1}{G_1 G_2} + \cdots \quad \text{— (i)}$$

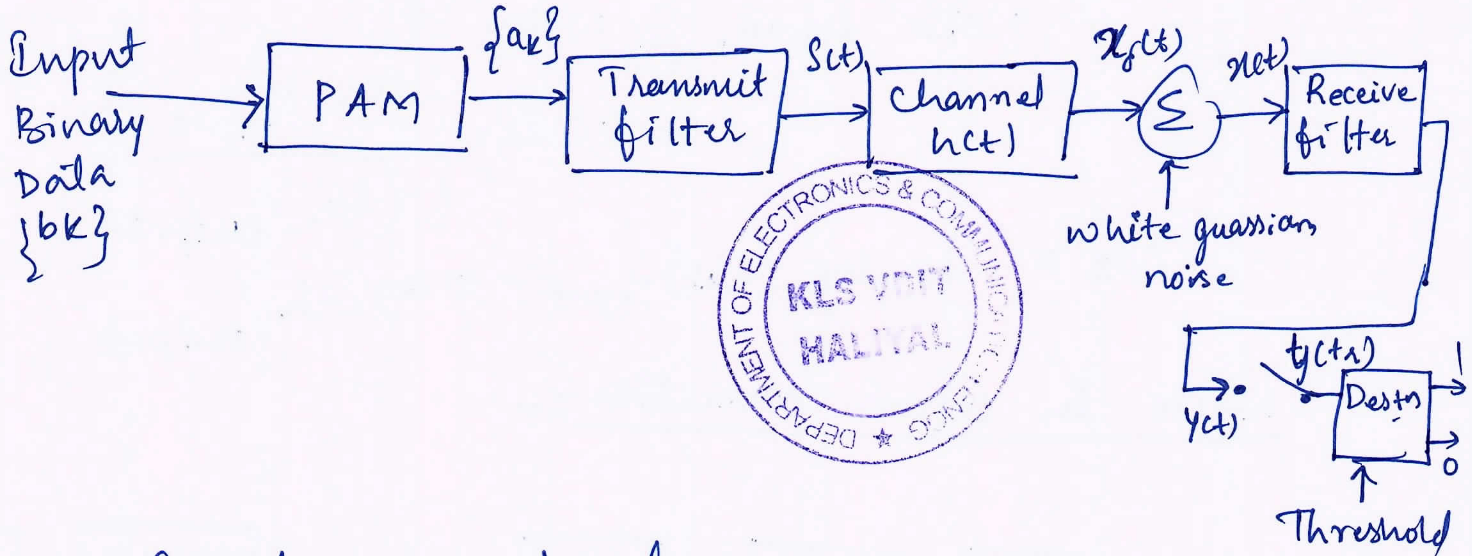
and its equivalent noise temperature is given by

$$T_e = (F-1)T$$

9)c) Define Inter Symbol Interference (ISI). Outline baseband binary data transmission System with neat block diagram and equations.

[Defn - 2M, Block diagram - 3M, Expln 3M) [08 Marks]

Soln: InterSymbol Interference (ISI) is that which arises when the communication channel is dispersive which means that channel has a frequency dependent amplitude Spectrum.



Consider a baseband binary PAM System, a generic form of which is shown in the figure above. The incoming binary sequence {bk} consists of Symbols 1 & 0, each of duration Tb.

The pulse amplitude modulator transforms this binary sequence into a new sequence of short pulses whose amplitude a_k is represented in the polar form.

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

The sequence of short pulses so produced, is applied to a transmit filter of impulse response $g(t)$, producing the transmitted signal $s(t)$.

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

→ The signal $s(t)$ is modified as a result of transmission through the channel of impulse response $h(t)$. The noisy signal $x(t)$ is then passed through a receive filter of impulse response $c(t)$. The resulting filter o/p $y(t)$ is sampled synchronously with the transmitter, with the sampling instants being determined by clock or timing signal that is usually extracted from the receive filter output.

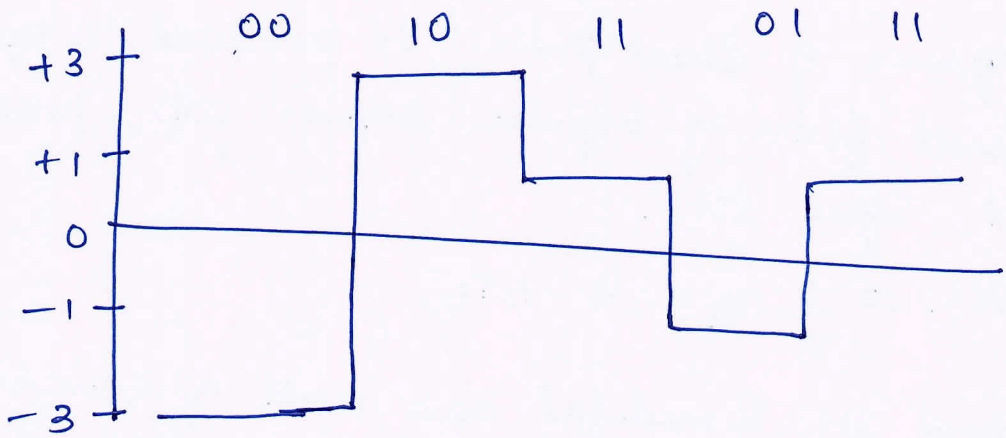
→ Finally the sequence of samples those obtained is used to reconstruct the original data sequence by means of a decision device.

→ If the threshold λ is exceeded, a decision is made in favour of symbol 1, If the threshold λ is not exceeded, a ~~decision~~ decision is made in favour of symbol 0.

②

10) a) Explain Bandwidth requirements of ISI Systems (Expln – 4M, Graph – 2M) [06 Marks]

Ans:– In Intersymbol Interference block diagram, the Pulse Amplitude Modulator produces binary pulses, that is pulse with one of two possible amplitude levels



Dibit	Amplitude
00	−3
01	−1
10	+1
11	+3

On the otherhand, in a Baseband M-ary PAM S/m, the pulse amplitude modulator produces one of M possible amplitude levels with NRZ as shown above.

In the above case of a quaternary (M=4) system and the binary data sequence 0010110111. In a M-ary System, the information source emits a sequence of symbols from an alphabet that consists of M symbols. Each amplitude level at the pulse amplitude modulator o/p corresponds to a distinct symbol, so that there are M distinct amplitude levels to be transmitted.

The binary PAM system produces information at the rate of $1/T_b$ bits per second.

10) b) Write short note on:
i) Signal to Noise Ratio — 2M
ii) External Noise — 3M
iii) Internal Noise — 3M
[08 Marks]

Soln: (i) Signal to Noise Ratio:

SNR indicates relative strength of the signal and the noise in a communication system. The stronger the signal, weaker ~~the~~ noise, Higher is the SNR.

ii) External Noise

External noise comes from sources over which we have little or no control.

a) Industrial noise b) Atmospheric noise c) Extra terristrial noise

a) ~~Industrial noise~~ Atmospheric noise

Electrical disturbances that occur naturally in the earth's atmosphere are another source of noise. These occur from lightening, discharges occur between clouds and earth.

b) Industrial noise

This noise occurs due to the movement of parts of machines.

c) Extra terristrial noise

The main source of this noise are, cosmic rays and solar radiation.

iii) Internal noise:

Electrical components in a receiver such as resistors, diodes and transistors are major sources of internal noise.

Thermal noise:

Most of the internal noise is caused by thermal noise or thermal agitation, a random movement of free electrons in a conductor caused by heat.

Semiconductor noise:

Electronic components such as diodes & transistors are major contributors of this noise.

Intermodulation noise:

Intermodulation distortion results from the generation of new signals and harmonics caused by circuit non linearities.

10) c) An RF amplifier has an S/N ratio of 8 at the i/p and S/N ratio of 6 at the output. What are the noise factor, noise figure and noise temperature?

[Each value - 2M] [06 Marks]

Soln: a) Noise factor, $F = \dfrac{(S/N) \text{ Power at the i/p}}{S/N \text{ power at the o/p}}$

$$= \frac{8}{6} = 1.33$$

b) Noise figure, $F_{dB} = 10 \log F$

$$= 10 \log 1.33$$

$$= 1.249$$

c) Noise Temperature, $T_e = (F-1)\,T$

$$T_e = (1.33 - 1)\,290$$

$$T_e = 95.7\,K$$