# KLS Vishwanathrao Deshpande Institute of Technology

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## 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 | : | Model Question Paper - 1 |
| Date of Submission | : | 19/8/2024 |

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# Model Question Paper-1 with effect from 2022-23 (CBCS Scheme)

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

## Fourth Semester B.E. Degree Examination
## Subject Title Principles of Communication Systems

**TIME: 03 Hours** **Max. Marks: 100**

Note: 01. Answer any **FIVE** full questions, choosing at least **ONE** question from each **MODULE**.
02.
03.

| | | **Module -1** | ***Bloom's Taxonomy Level** | **Marks** |
|---|---|---|---|---|
| Q.01 | a | Define Probability. Illustrate the relationship between sample space, events and probability. | L2, CO5 | 06 |
| | b | Define the autocorrelation and cross-relation functions. Infer the properties of autocorrelation function. | L2, CO5 | 06 |
| | c | Develop a program to generate the probability density function of Gaussian distribution function. | L3, CO5 | 08 |
| | | OR | | |
| Q.02 | a | What is conditional probability? Prove that P(B/A) = P(A/B).P(B)/P(A) | L2, CO5 | 06 |
| | b | Outline random processes and illustrate an ensemble of sample functions with a neat diagram. | L2, CO5 | 06 |
| | c | Show that, if a Gaussian process X(t) is applied to a stable linear filter, then the random process Y(t) developed at the output of the filter is also Gaussian. | L3, CO5 | 08 |
| | | **Module-2** | | |
| Q. 03 | a | Interpret the concepts of modulation index and percentage of modulation. Write the necessary equations. | L2, CO1 | 08 |
| | b | A standard AM broadcast station is allowed to transmit modulating frequencies upto 5 KHz. If the AM station is transmitting on a frequency of 980 KHz, compute the maximum and minimum upper and lower sidebands and the total bandwidth occupied by the AM station. | L3, CO1 | 05 |
| | c | Explain high-level collector modulator with a neat block diagram. | L2, CO1 | 07 |
| | | OR | | |
| Q.04 | a | Outline a diode detector AMD modulator with necessary block diagram and waveforms. | L2, CO1 | 08 |
| | b | An AM transmitter has a carrier power of 30W. The percentage of modulation is 85 percent. Calculate (i) the total power and (ii) the power in one sideband. | L3, CO1 | 05 |
| | c | Explain a general block diagram of an FDM system. | L2, CO1 | 07 |
| | | **Module-3** | | |
| Q. 05 | a | Compare and contrast FM and AM. | L3, CO2 | 06 |
| | b | Explore with a neat diagram the concept of frequency modulation with an IC VCO. | L2, CO2 | 07 |
| | c | Draw the block diagram of a super heterodyne receiver and explain the function of each block. | L2, CO2 | 07 |
| | | OR | | |
| Q. 06 | a | Identify a method used to convert a phase-modulated (PM) signal into a frequency-modulated (FM) signal. | L3, CO2 | 06 |
| | b | Define PLL. Explain the basic block diagram of a PLL along with capture and lock ranges. | L2, CO2 | 07 |
| | c | Interpret the concept of a mixer with a neat schematic diagram. | L2, CO2 | 07 |
| | | **Module-4** | | |
| Q. 07 | a | What are the advantages of digital signals over analog signals? | L1, CO3 | 04 |

| | | | | |
|---|---|---|---|---|
| | b | State sampling theorem. Explain sampling with neat sketches and equations. What are the challenges faced with Nyquist criteria for sampling? Develop a program to display the signals and its spectrum. | L3, CO3 | 10 |
| | c | Explain the generation and detection of PPM waves with a relevant block diagram. | L2, CO3 | 06 |
| | | OR | | |
| Q. 08 | a | What is aperture effect in PAM systems? How can it be minimized? | L1, CO3 | 04 |
| | b | What is multiplexing and why is it required in communication? Explain the working of TDM with a neat block diagram. | L3, CO3 | 10 |
| | c | Explain the basic elements of a PCM system with neat diagrams. | L3, CO3 | 06 |
| | | **Module-5** | | |
| Q. 09 | a | Define Intersymbol Interference (ISI). Outline Baseband binary data transmission system with neat block diagram and equations. | L2, CO4 | 08 |
| | b | Develop a code to generate and plot eye diagram. | L3, CO4 | 06 |
| | c | Illustrate the concept of noise in cascaded stages with a diagram. Write Friis' formula and mention its terms. | L2, CO1 | 06 |
| | | OR | | |
| Q. 10 | a | Explain the following concepts briefly:<br>(i) Nyquist criterion for distortionless transmission.<br>(ii) Baseband M-ary PAM transmission | L2, CO4 | 08 |
| | b | Develop a code to generate NRZ and RZ pulse. | L3, CO4 | 06 |
| | c | Define Signal-to-Noise Ratio (SNR). Explain the different types of external and internal noise. | L2, CO1 | 06 |

*Bloom's Taxonomy Level: Indicate as L1, L2, L3, L4, etc. It is also desirable to indicate the COs and POs to be attained by every bit of questions.

1) a) Define Probability. Illustrate the relationship betw-een Sample space, events and Probability

[Defn - 2M, Illustration - 4M] [06 Marks]

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

Let a random experiment is repeated n-times. If the event occurs $n_A$ times, then 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 occurance of event A.

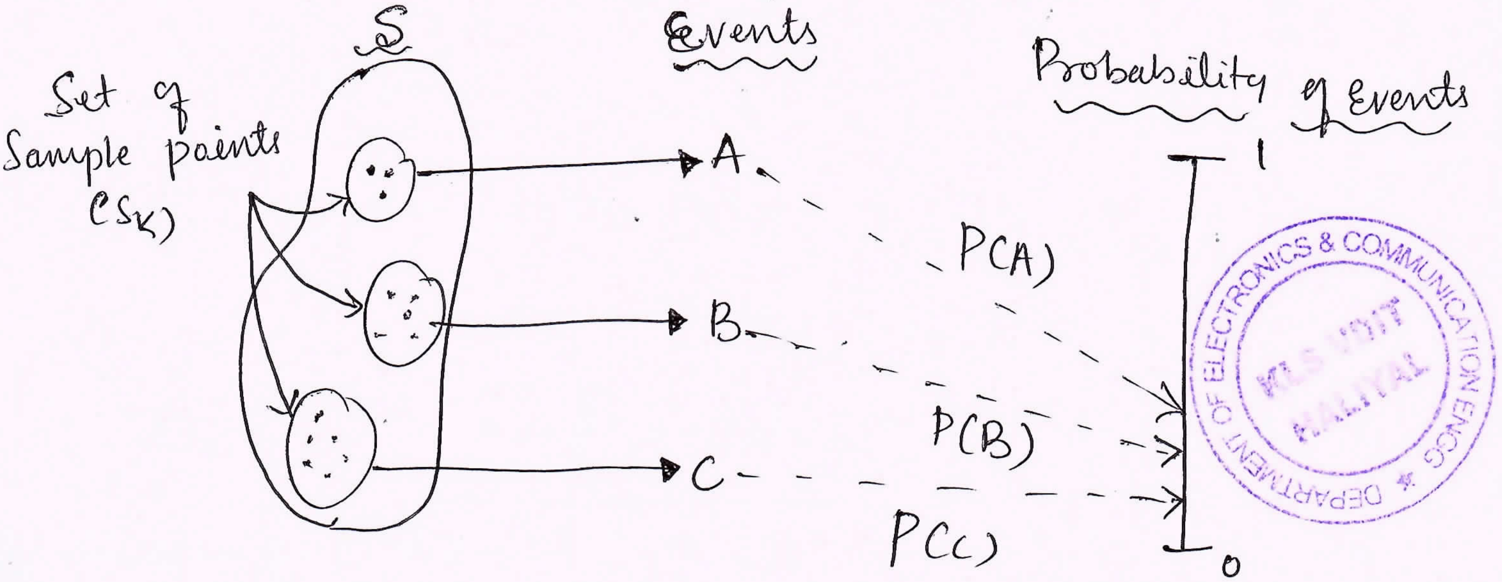



Fig: Relationship between Sample Space (S), Events and Probability

The concept of a random variable shown in above figure, where we have suppressed the events but show subsets of the sample space being mapped directly to a subset of the real line.

The probability function applies to this random variable in exactly the same manner that it applies to the underlying events.

1) b) Define the Auto correlation and cross- correlation functions. Infer the properties of auto correlation function. [06 Marks]

[Defn - 2M + 2M, Properties - 2M]

Soln: The auto correlation function of the random variables '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,

$$r_x(k,l) \text{ or } r_x(\tau)$$

i.e Auto correlation function is given by,

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



Properties of Auto Correlation function :-

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

(i) $r_x(\tau) = r_x(k-l)$ ; It is a function of time difference $(k-l)$

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

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

i.e $r_{x=0} > r_x(\tau)$ for any value of $\tau$

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

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

1) c) Develop a program to generate the probability density function of Gaussian Distribution function
[program – 08M] [08 Marks]

Soln:

```
mu=0;
Sigma=1;
x= linspace (-5, 5, 1000);
Pdf = (1/ Sigma * sqrt (2 * pi))) * exp(-0.5 *
        ((x-mu)/ Sigma) .^2);
figure;
plot (x, pdf, 'Line width', 2);
title ('Gaussian Distribution Function');
xlabel ('x');
ylabel('probability Density');
grid on;
```



2) a) what is conditional Probability? Prove that
$P(B/A) = P(A/B) \cdot P(B)/P(A)$
[06 Marks]

Soln: Conditional Probability is defined as the ratio of Joint probability to that of probability of known event.

$$P[X/Y] = \frac{P[X, Y]}{P[Y]}$$

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

Soln: We know that conditional probability

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

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

Similarly $P[A/B] = P[A/B]\ P[B]$ — (4)

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



By comparing equations (3) and (4) we get,

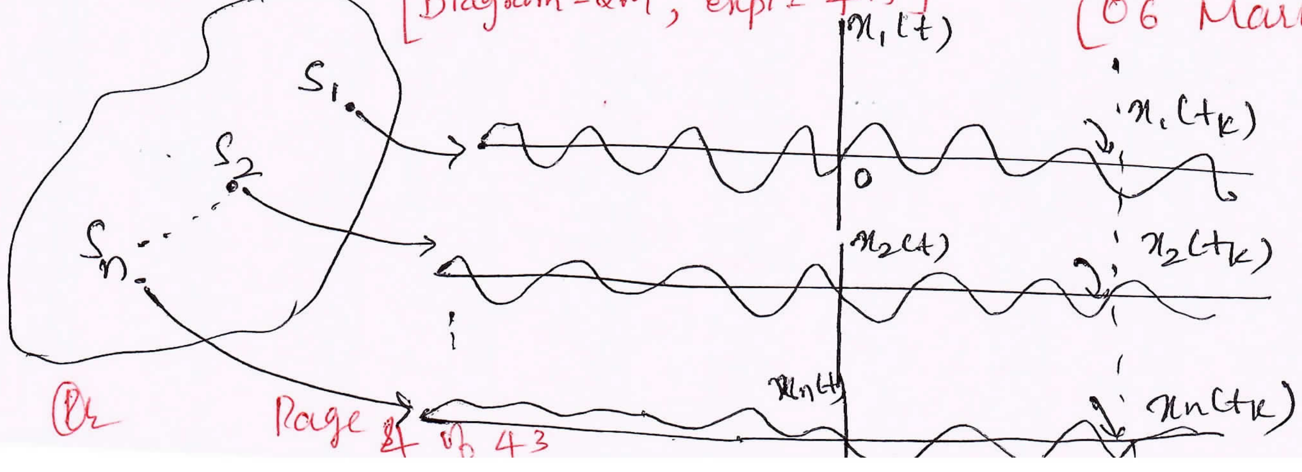
$$P[B/A]\ P[A] = P[A/B]\ P[B]$$

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

2) b) Outline random processes and illustrate an ensemble of sample functions with a neat diagram

[Diagram - 2M, Expl 2-4M] [06 Marks]

Soln:




Dr

→ Figure above illustrates a set of sample fns $\{x_j(t) \mid j = 1, 2, \ldots n\}$.

From this figure, we note that for a fixed time $t_K$ inside the observation interval, the set of numb

$$\{x_1(t_K), x_2(t_K) \ldots x_n(t_K)\}$$
$$= \{X(t_K, S_1), X(t_K, S_2), \ldots, X(t_K, S_n)\}$$



Constitutes a random variable.

Thus we have an indexed ensemble of random variables $\{X(t, S)\}$, which is called a random process.

→ To simplify the notation, the customary practice is to suppress the S and simply use $X(t)$ to denote a random process.

→ We can then define a random process $X(t)$ as an ensemble of time functions together with a probability rule that assigns a probability to any meaningful event associated with an observation of one of the sample functions of the random process.

2) c) Show that, if a Gaussian process $X(t)$ is applied to a stable linear filter, then the random process $Y(t)$ developed at the output of the filter is also Gaussian

[Statement - 3M, Proof - 5M] [08 Marks]

PL

Soln: It is a property of Gaussian process

This property is derived by using the defi-tion of Gaussian process.




Consider the figure above, where we have a linear time invariant filter of impulse response $h(t)$, with the random process $x(t)$ as input and the random process $Y(t)$ as o/p.

→ we assume that $x(t)$ is a Gaussian process. The random processes $Y(t)$ and $x(t)$ are related by. convolutional integral

$$Y(t) = \int_0^T h(t-\tau)\, x(\tau)\, d\tau \quad , 0 \le t \le \infty$$

→ we assume that the impulse response $h(t)$ is such that the mean square value of the o/p random process $Y(t)$ is finite for all $t$ in the range $0 \le t \le \infty$ for which $Y(t)$ is defined.

If we defined the random variable

$$Z = \int_0^{\infty} g_Y(t) \cdot \int_0^T h(t-\tau)\, x(\tau)\, d\tau\, dt$$

then Z must be a Gaussian random variable for every function $g_y(t)$, such that the mean square value of Z is finite.

Interchanging the order of integration in above equation, we get,

$$Z = \int_0^T g(\tau) X(\tau) \, d\tau$$

where

$$g(\tau) = \int_0^\infty g_y(t) \, h(t-\tau) \, dt$$



∴ if i/p $X(t)$ to a linear filter is a Gaussian process, then the o/p $Y(t)$ is also a Gaussian process.

3) a) Interpret the concepts of modulation index and percentage of modulation. Write the necessary equations. [MI - 2M, % Mod$^n$ - 2M, Eq$^n$ - 4M] [08 Marks]

Soln: For undistorted Amplitude Modulation to occur the modulating signal voltage, $V_m$, must be less than the carrier voltage $V_c$.

Therefore, the relationship between the amplitude of the modulating signal and the amplitude of the carrier signal is important.

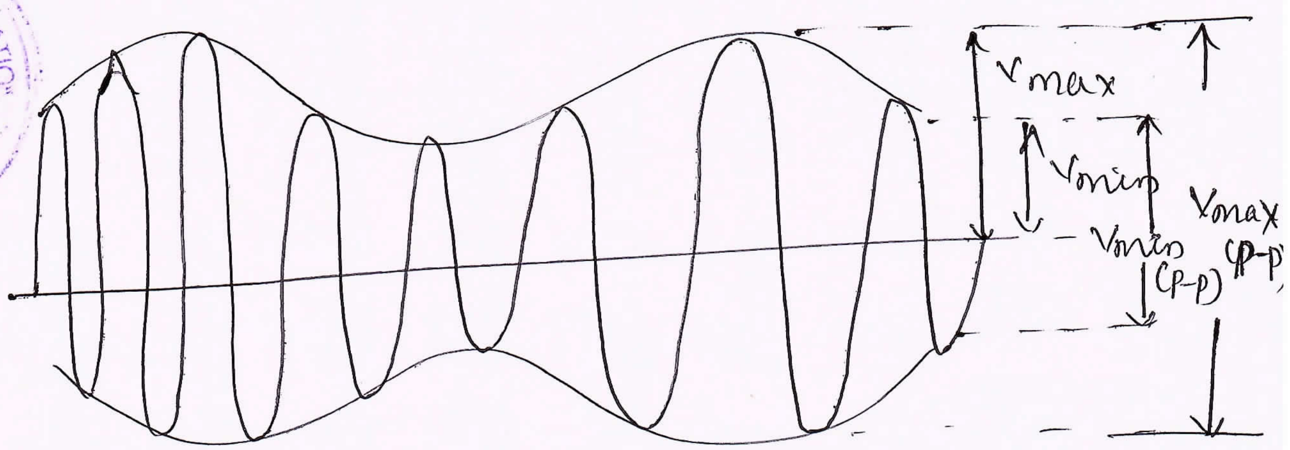
$$\boxed{\text{modulation index, } m = \frac{V_m}{V_c}}$$

Multiplying the modulation index by 100 gives the percentage of modulation.

## Percentage of Modulation

→ The modulation index can be determined by measuring the actual values of the modulation voltage and the carrier voltage and computing the ratio.

→ When the AM signal is displayed on an CRO, the modulation index can be computed from $V_{max}$ and $V_{min}$ as shown in fig below.




The peak value of modulating signal $V_m$ is one half of the difference of the peak and trough values,

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

As shown in figure, $V_{max}$ is the peak value of signal during modulation and $V_{min}$ is the lowest value or trough of modulated wave

$V_{max}$ is the one half the peak to peak value of the AM signal or

$$V_{max} = \frac{V_{max}(p-p)}{2}$$

→ Subtracting $V_{min}$ from $V_{max}$ produces the peak to peak value of the modulating signal.

→ The peak value of carrier signal, $V_c$ is the average value of $V_{max}$ and $V_{min}$ values

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

The modulation index is given by,

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

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

5

3) a) A standard AM broadcast station is allowed to transmit modulating frequencies upto 5 KHz. If the AM station is transmitting on a frequency of 980 KHz, compute the maximum upper and lower sidebands and the total bandwidth occupied by the AM station.

[05 Marks]

Soln: $f_{USB} = 980 + 5 = 985$ KHz — (1)

$f_{LSB} = 980 - 5 = 975$ KHz — (1)

$BW = f_{USB} - f_{LSB}$

$= 985 - 975$

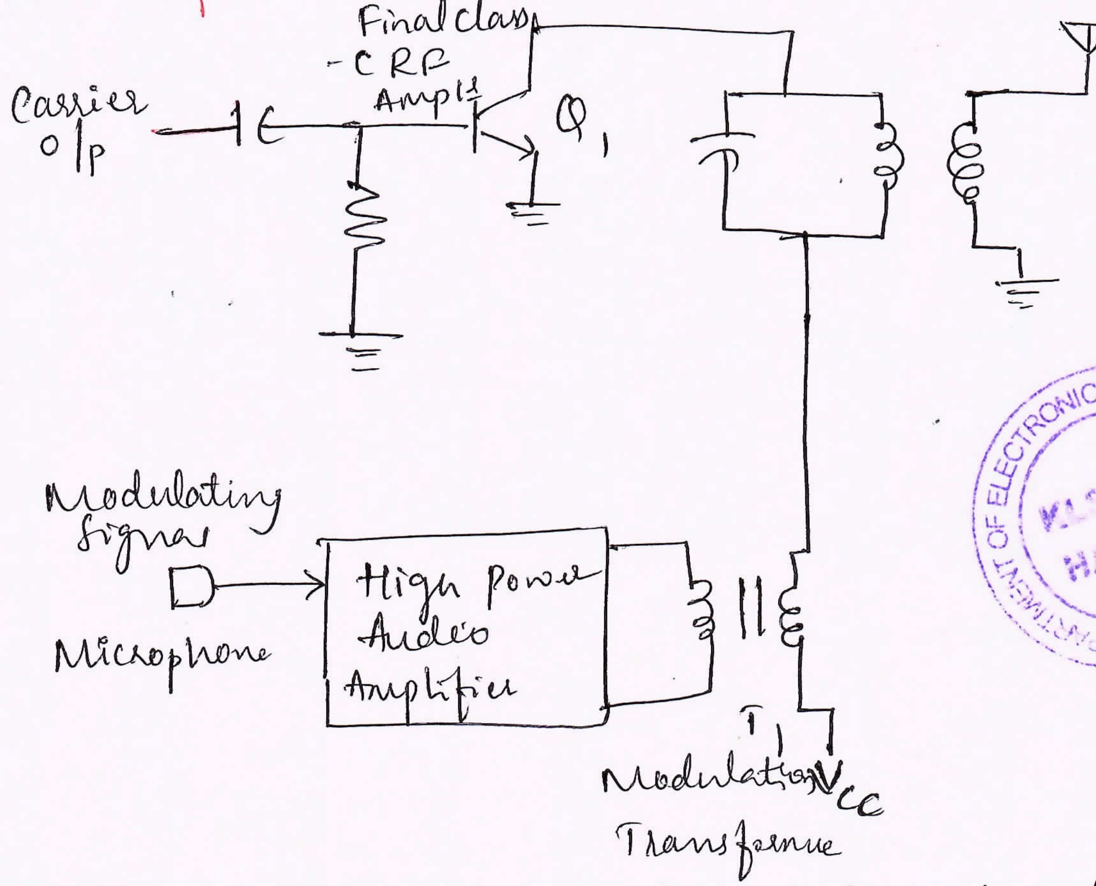
$= 10$ KHz — (2)

$BW = 2(5 \text{ KHz}) = 10$ KHz — (1)

3) c) Explain high-level collector modulator with a neat block diagram. [07 Marks]

[Block dgm - 3M
Expln - 4M]

Soln:-




In the above figure, the o/p stage of the transmitter is a high power class c amplifier.

→ class c amplifiers conducts for only a portion of the Positive half cycles of their i/p signal.

The collector current pulses cause the tuned circuit to oscillate at the desired o/p frequency. The tuned circuit therefore reproduces the negative portion of the carrier signal.

The modulator is a linear power amplifier that takes the low level modulating signal and amplifies it to a high power level.

→ The secondary winding of the modulation transformer is connected in series with the collector supply voltage Vcc of the class C amplifier.

→ With a zero modulation i/p signal, there is zero modn voltage across the secondary of $T_1$, the collector supply voltage is applied directly to the class C amplr and the o/p carrier is a steady sine wave.

→ 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 modulation signal goes negative, it subtracts from the collector supply voltage, decreasing it.

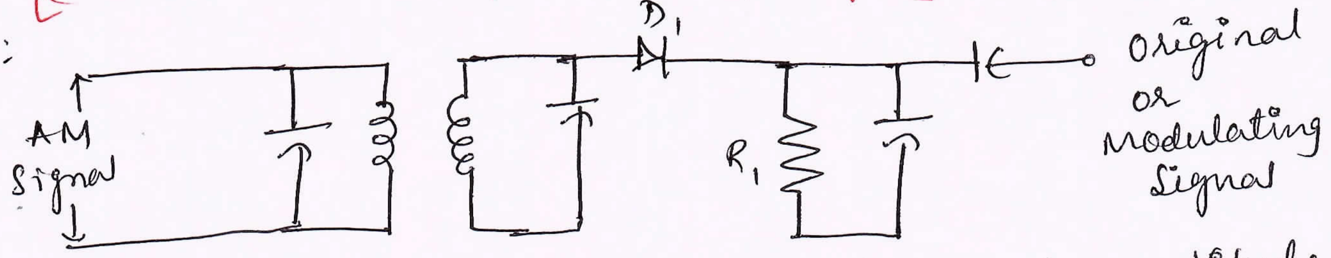
→ For that reason, the class C amplr current pulses are smaller, resulting in a lower amplitude carrier o/p.

→ When the positive peak occurs, the voltage applied to the collector is twice the collector supply voltage.

→ When the negative peak is equal to the supply voltage the effective voltage applied to the collector of $Q_1$ is zero, producing zero carrier.

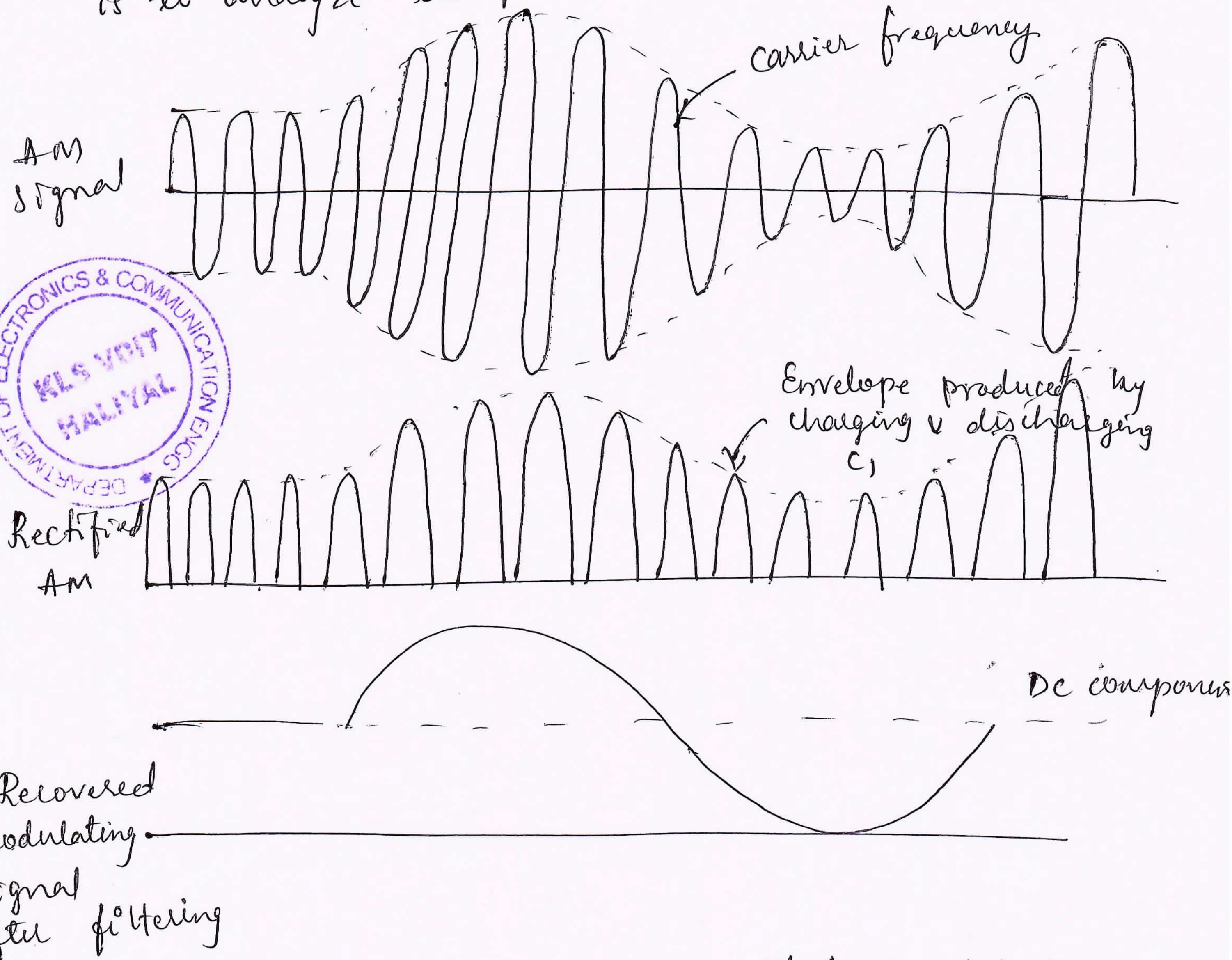
4) a) Outline a diode detector AMD modulator with necessary block diagram and waveforms.
[Circuit - 2M, waveforms - 2M, Expln - 4M] [08 Marks]

Soln:




The simplest and most widely used amplitude demodulator is the diode detector as shown in fig above.

→ One way to look at the operation of a diode detector is to analyze its operation in the time domain.






The waveforms shown above illustrate detection.

→ On each positive alternation of AM signal, the capacitor charges quickly to the peak value of the pulses passed by the diode.

* when the pulse voltage drops to zero, the capacitor discharges into resistor $R_1$. The time constant of $C_1$ and $R_1$ is choosen to be long compared to the period of the carrier.

* As a result, the capacitor discharges only slightly during the time that the diode is not conducting.

* when the next pulse comes along, the capacitor again charges to its peak value.

* when diode cuts off, the capacitor again discharges a small amount into the resistor. The resulting waveform across the capacitor is a close approximation to the original modulating signal.

→ Because the capacitor charges and discharges the recovered signal has a small amount of ripple on it, causing distortion of the modulating signal.

→ As shown the AM signal is usually transformer coupled and applied to a basic half wave rectifier circuit consisting of $D_1$ and $R_1$.

→ The diode conducts when the positive half cycles of the AM signals occur. During the negative half cycles, diode is reverse biased and no current flows through it.

4)b) An AM transmitter has a carrier power of 30W. The percentage of modulation is 85%. Calculate i) the total power ii) the power in one side band.

[05 Marks]

[ $P_c$ – 3M, $P_{USB}$ – 2M ]



Soln:

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

$= 30\left[1+\frac{(0.85)^2}{2}\right]$

$= 30\left[1+\frac{0.7225}{2}\right]$

$\boxed{P_T = 40.8W}$

ii) $P_{SB}$ (Both) $= P_T - P_C$

$= 40.8 - 30 = 10.8W$

$P_{SB}$ (one) $= \frac{P_{SB}}{2} = \frac{10.8}{2} = 5.4W$

4)c) Explain a general block diagram of an FDM system.
[Block dgm – 3M, Expl$^n$ – 4M] [07 Marks]

Soln: In freq division multiplexing, multiple signals share the bandwidth of a common channel.

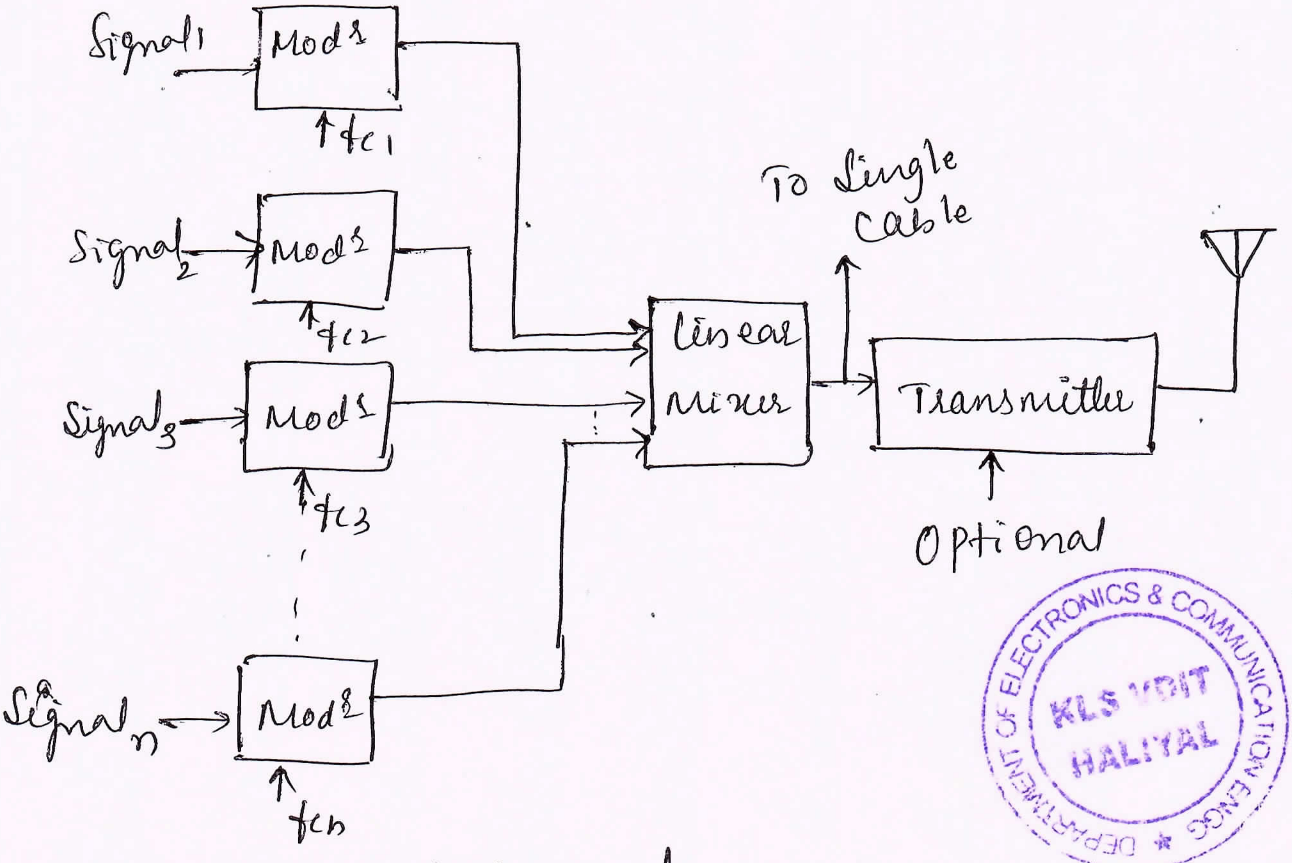



Fig: Transmitting end of an FDM system



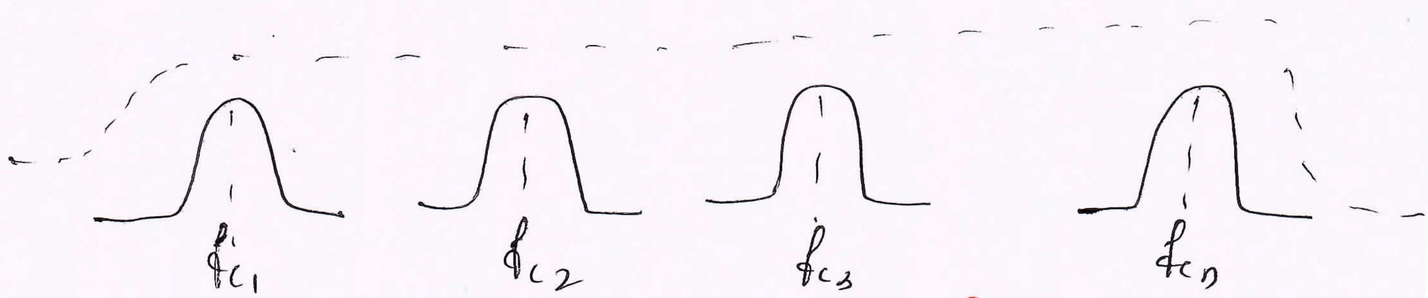



fig: Spectrum of FDM frequency →

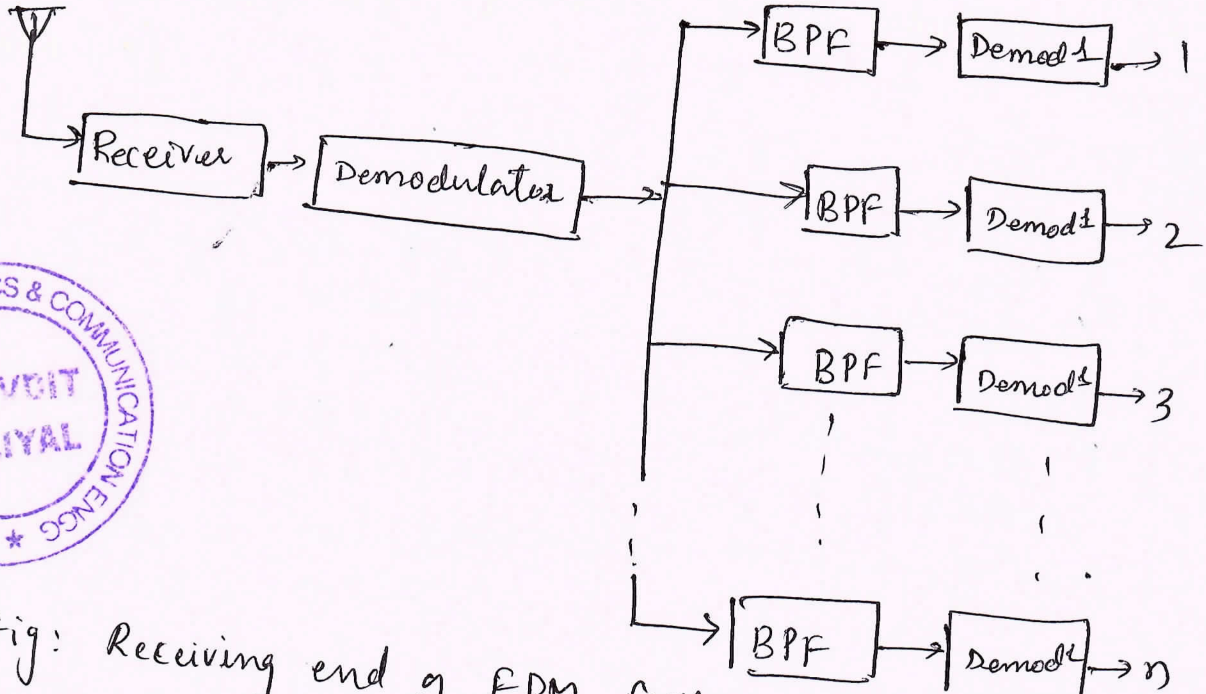





Fig: Receiving end of FDM System

→ A Co-axial Cable has a bandwidth of about 1GHz. The bandwidth of radio channels vary and are usually determined by FCC regulation and the type of radio service involved.

→ The figure shows a block diagram of an FDM System. Each Signal to be transmitted feeds a modulator circuit. The carrier frequencies are usually equally spaced from one another over a Specific frequency range. These carriers are referred to as subcarriers.

→ Each i/p Signal is given a portion of a bandwidth. The System is shown in the Spectrum.

→ In the receiving end of an FDM System a receiver Picks up the signal and demodulates it, ~~Picks up~~ the Composite signal. This is sent to a group of band pass filters each centered on one of the carrier frequencies.

→ A channel demodulator then recovers each original i/p Signal.

5) a) compare and contrast FM and AM.

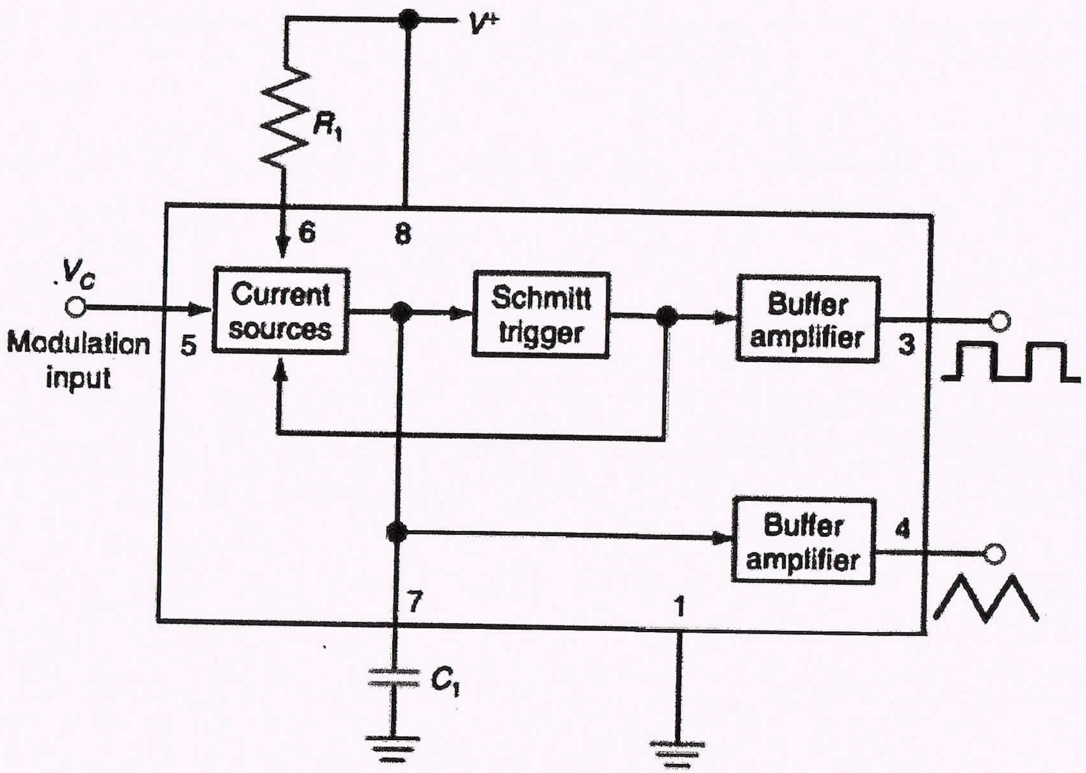
[6 difference - 1 M each] [06 Marks]

Soln:

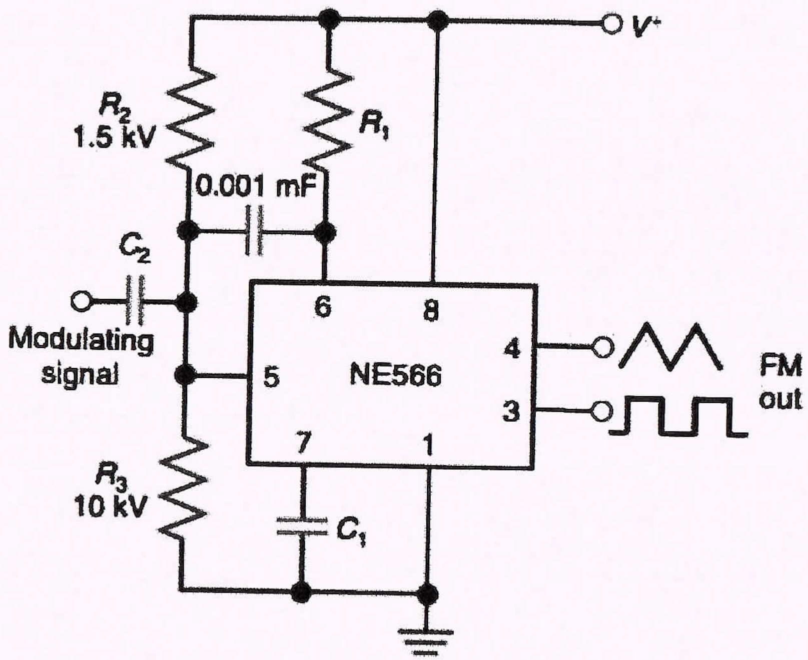
| FM | AM |
| --- | --- |
| 1) Amplitude of FM wave is Constant. | 1) Amplitude of AM wave will change with the modulating voltage. |
| 2) Transmitted power remains constant | 2) Transmitted power depends on modulation index. |
| 3) FM receivers are immune to noise | 3) AM receivers are not immune to noise |
| 4) noise can be decreased by increasing deviation | 4) This feature is absent in AM. |
| 5) $BW = 2[\Delta f + f_m]$ | 5) $BW = 2f_m$ |
| 6) BW is large. Hence wide channel is required | 6) BW is much less than FM. |

5) b) Explore with a neat diagram the concept of frequency modulation with an IC VCO.

[Diagram - 3M, Expln - 4M] [07 Marks]


V+
R1
6
8
Vc
Current sources
Schmitt trigger
Buffer amplifier
Modulation input
5
3
Buffer amplifier
4
7
1
C1

(a)


V+
R2
1.5 kV
R1
0.001 mF
C2
Modulating signal
6
8
NE566
5
4
3
FM out
7
1
R3
10 kV
C1

(b)

MD

Fig a above is a block diagram of 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.

→ Schmitt trigger 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 saw tooth of voltage is developed across the capacitor by the current source. This is buffered by an amplifier and made available at pin 3.

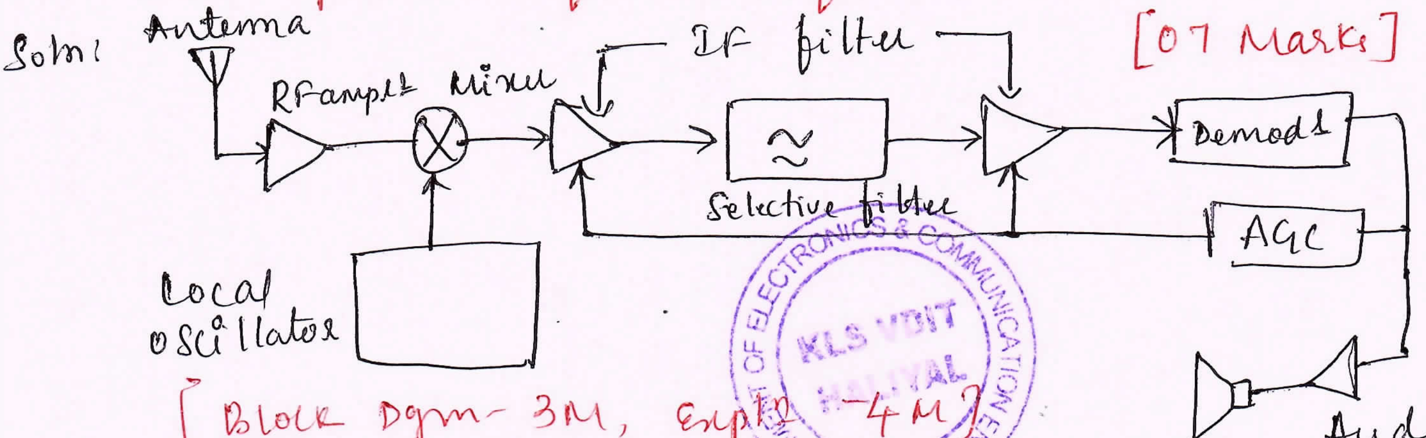
→ From fig(b), the current sources are biased with a voltage divider made up of $R_2$, $R_3$.

→ The modulating signal is applied through $C_2$ to the voltage divider at pin 5.

The deviation is linear with respect to the i/p amplitude over the entire range.

5) c) Draw the block diagram of a super heterodyne receiver and explain the function of each block. [07 Marks]

Soln:




[Block Dgm - 3M, Expln - 4M]

→ Super heterodyne receivers convert all incoming Signals to a lower frequency known as the intermediate frequency at which a single set of amplifiers and filters is used to provide a fixed level of Sensitivity and Selectivity.

→ The Key circuit is the mixer, which acts as simple amplitude modulator to produce sum and difference frequencies.

→ Figure above shows a general block diagram of a Superheterodyne receiver.

RF amplifier:

The antenna picks up the weak signal and feeds it to the RF amplifier, also called low-noise amplifier.

Mixers and Local Oscillators:

The o/p of RF amplifier is applied to the i/p of the mixer. The mixer also receives an i/p from a local oscillator or freq synthesizer.

IF amplifiers

The o/p of mixer is an IF signal containing the same modulation that appeared on the o/p RF signal. This signal is amplified by one or more IF amplifier stages.

Demodulators:

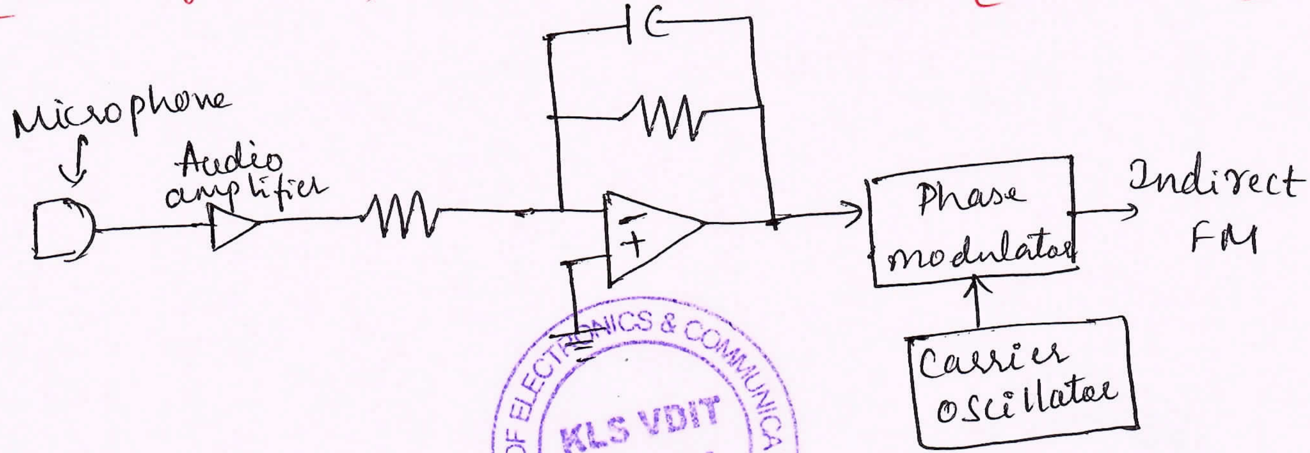
A highly amplified IF signal is finally applied to the demodulator or detector, which recovers the original modulating information.

Automatic Gain Control:

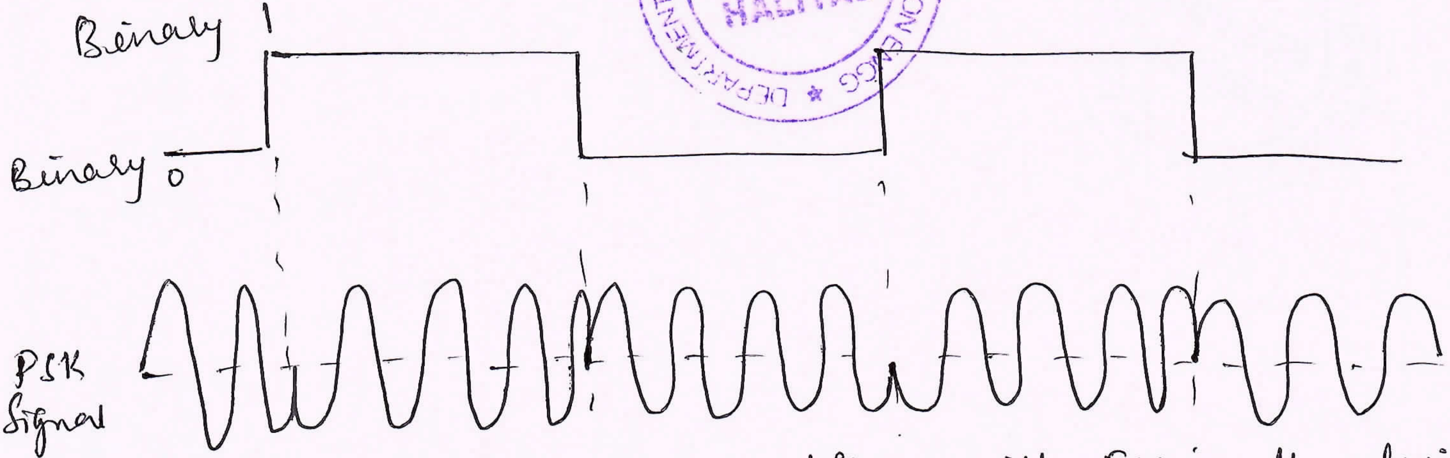
The o/p of a demodulator is usually the original modulating signal, the amplitude of which is directly proportional to the amplitude of the received signal.

5)a) Identify a method used to convert a phase modulation (PM) signal into a frequency modulated signal. [Block dgm - 2M, w/f - 2M, Expl^n - 2M] [06 Marks]

Soln:






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 in figure above

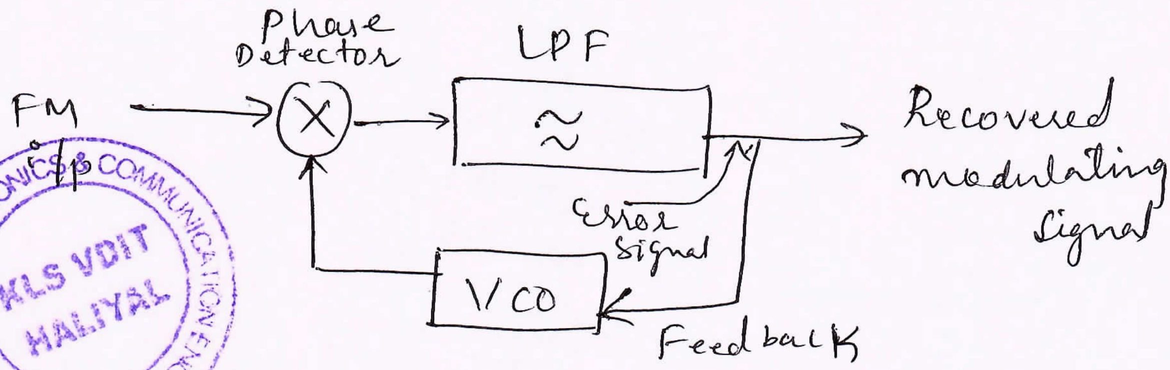
→ This low pass filter called a frequency correcting network, predistorter or 1/f filter, causes the higher modulating frequencies to be attenuated.

→ The predistorter compensates for the excess freq 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.

6) b) Define PLL. Explain the basic block diagram of a PLL along with capture and lock ranges.
[Defn - 1M, Block dgm - 2M, capture & lock - 2M, Graph - 1M] [07 Marks]

Soln: A phase locked loop (PLL) is a frequency or phase sensitive feedback control circuit used in frequency demodulation, freq synthesizers, and various filtering and signal detection applications.




All phase locked loops have the three basic elements as shown above.

1) Phase detector is used to compare the FM i/p, some times referred to as the reference signal to the o/p of a VCO.

2) The VCO frequency is varied by the dc o/p voltage from a low pass filter.

3) 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 i/p signals and generate an o/p signal that, when filtered, will control the VCO.

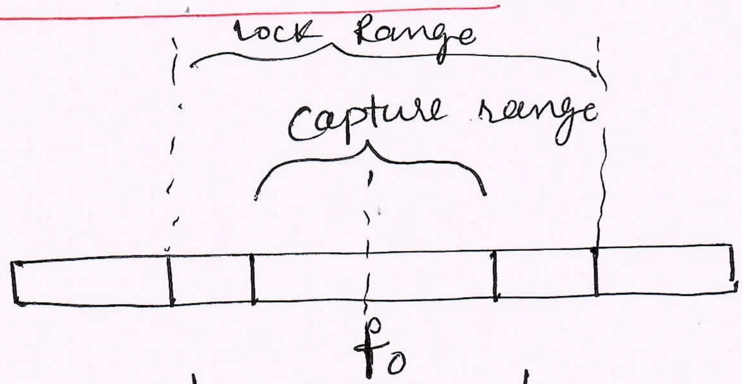
→ If there is a phase or frequency difference between the FM i/p and VCO signals, the phase detector o/p varies in proportion to the difference

The filtered O/P adjusts the VCO freq in an attempt to correct for the original freq or phase difference. 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 is determined by internal frequency-determining components.

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

## Capture and Lock ranges




$f_0$ = VCO free running frequency

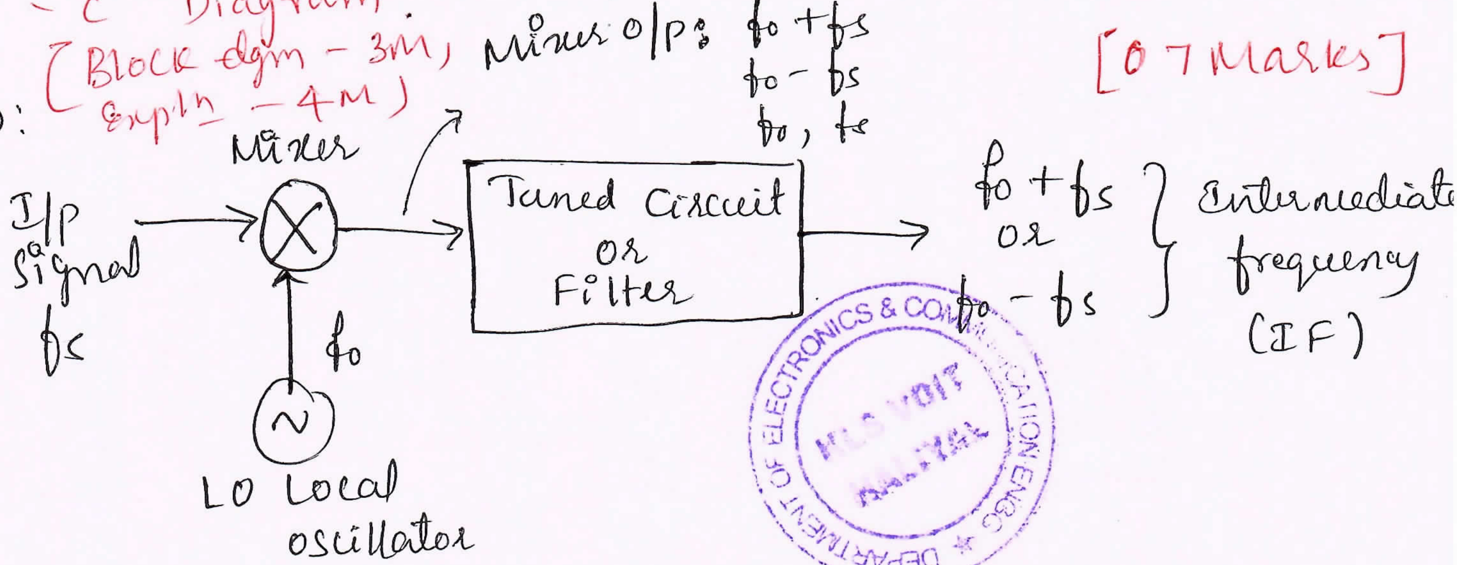
**Lock Range:** The range of frequencies over which a PLL can track the i/p signal and remains locked is known as lock range.

**Capture Range:**

The range of frequencies over which a PLL will capture an i/p signal, known as the capture range.

6) c) Interpret the concept of a mixer with a Schematic Diagram. [07 Marks]

Soln: [Block dgm - 3M, Expln - 4M)




The function performed by the mixer is called heterodyning.

Figure above is a schematic diagram of a mixer circuit. Mixers accept two i/ps. The signal $f_s$ which is to be translated to another frequency, is applied to one input and the sinewave from a local oscillator $f_o$ is applied to the other input.

→ A mixer performs a mathematical multiplication of its two i/p signals according to the principles. The oscillator is the carrier and the signal to be translated is the modulating signal. The o/p contains not only the carrier signal but also sidebands formed when the local oscillator and input signal are mixed. The o/p of the mixer, therefore, consists of signals $f_s$, $f_o$, $f_o + f_s$ and $f_o - f_s$ or $f_s - f_o$.

7) a) What are the advantages of digital signal over analog signals?

[04 Marks]

Soln: The advantages of Digital Signal Over Analog Signal are

1) Digital signals are less sensitive to noise th Analog.

2) With digital systems, it is easier to integrate different Services, for example, Video and accompany-ng Sound track, into the same transmission scheme

3) The transmission Scheme can be relatively independ-ent of the Source.



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

5) Digital Signals are Simpler to characterize and typically do not have the same amplitude range and ~~typically do not have the sa~~ variability as analog signals.

7) b) State Sampling theorem. Explain Sampling with neat Sketches and equations. What are the challenges faced with Nyquist Criteria for Sampling? Develop a program to display signals and its spectrum

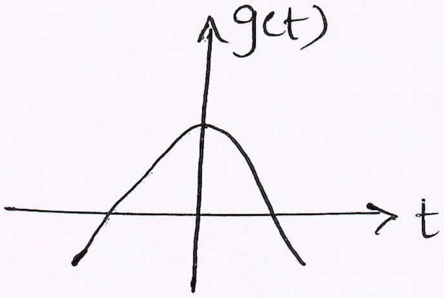
[Statement - 2M, Expln - 4M, Program - 4M] [10 Marks]

Soln: Sampling theorem states that any continuous time Signal can be completely represented in its samples and recovered back if the sampling frequency is greater than or equal to twice the highest frequency component of baseband Signal.

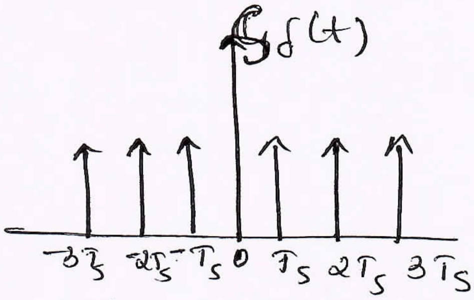
i.e $f_s \geq 2W$ or $f_s \geq 2f_m$

→ This condition is also called Nyquist condition for sampling process.

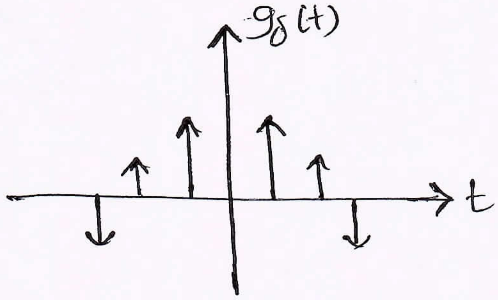
→ Consider an arbitrary signal g(t) of finite energy which is specified for all time. A segment of signal g(t) is shown in figure.




(a) Analog signal




(b) Periodic signal





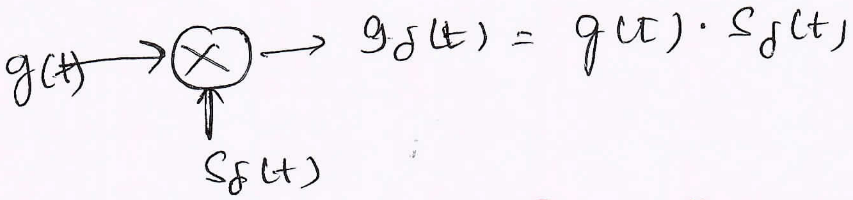

(c) Sampled signal

Suppose that we sample the signal g(t) instantaneously and at a uniform rate, once every $T_s$ seconds.

* Consequently we obtain an infinite sequence of samples spaced $T_s$ seconds apart and denoted by $\{g(nT_s)\}$ where 'n' takes an all positive integer values.

* We refer to '$T_s$' as the sampling period and to its reciprocal $f_s = 1/T_s$ as sampling rate.

This form of sampling is called instantaneous sampling.




$$g_\delta(t) = g(t) \cdot \delta_\delta(t)$$

* Let $g_\delta(t)$ denote the signal obtained by individually weighting the elements of a periodic sequence spaced $T_s$ seconds. Therefore sampled o/p $g_\delta(t)$ is given by,

$$g_\delta(t) = g(t) * s_\delta(t) \quad \text{—} (1)$$

Let $s_\delta(t)$ denote the periodic impulse train and is represented as

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

Substituting eqn (2) in eqn (1) we get

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

using shifting property of impulse function,

$$g(t) * \delta(t - nT_s) = g(nT_s)\,\delta(t - nT_s) \quad \text{—} (3)$$

For frequency domain,

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

Taking fourier Transform on both sides

$$G_\delta(f) = G(f) * S_\delta(f) \quad \text{—} (4)$$

where $S_\delta(f) = f_s \sum_{n=-\infty}^{\infty} \delta(f - nf_s) \quad \text{—} (5)$

Substituting eqn (5) in eqn (4) we get,

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

From convolution property of impulse function, WKT

$$G(f) * \delta(f - nf_s) = G(f - nf_s)$$

$$\therefore G_\delta(f) = f_s \sum_{n=-\infty}^{\infty} G(f - nf_s) \quad \text{—} (6)$$

Eqn (6) can be written as,

$$G_\delta(f) = f_s G(f) + f_s \sum_{\substack{n=-\infty \\ n \neq 0}}^{\infty} G(f - nf_s) \quad \text{—} (7)$$

When the spectrum of $G_\delta(f)$ is passed through an LPF, the second term in the RHS of eqn 7 is eliminated

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

or $$G_\delta(f) = \frac{1}{T_s} G(f)$$

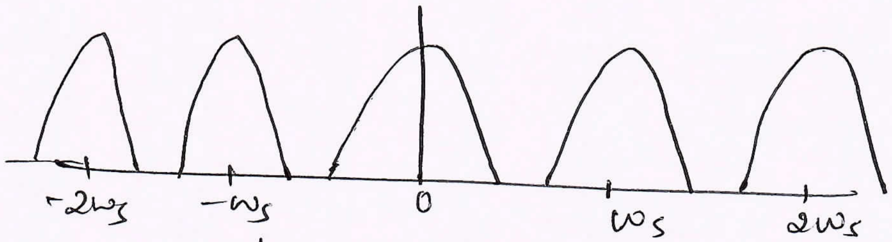
where $f_s = 2W$

$\therefore$ The Nyquist rate is $f_s \geq 2f_m$

**Challenges faced with Nyquist Criteria are:**

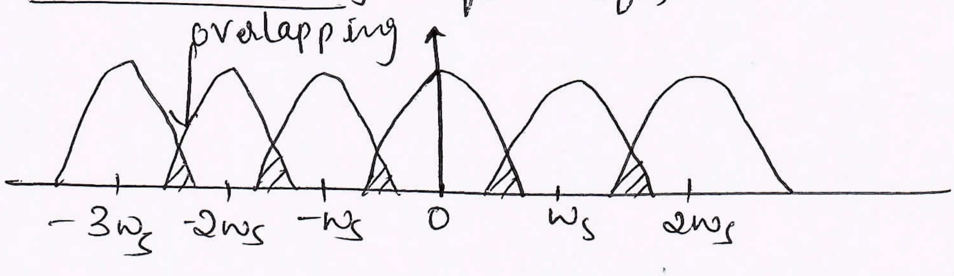
1) Over sampling
2) Under sampling

1) **Over sampling:** $f_s > 2f_m$




→ No overlapping of successive cycles, but forms a gap between them. So forms distortion

2) **Under sampling:** $f_s < 2f_m$



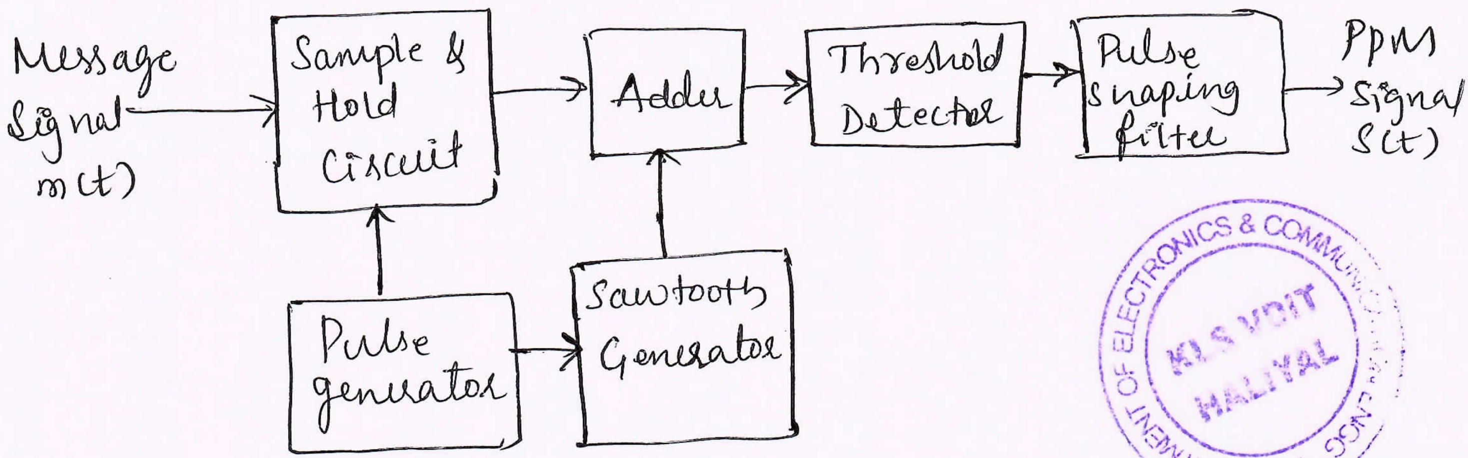

Because of overlapping, when passed through LPF, there is chance of distortion.. Due to aliasing effect, it is not possible to recover original signal $x(t)$ by LPF.

Program to display Signals and its Spectrum :-

```
fs = 1000;
t = 0:1/fs : 1-1/fs ;
f1 = 50;
f2 = 120;
Signal = 0.7 * Sin (2 * pi * f1 *t) + Sin (2 * pi * f2 * t)
Subplot (2,1,1);
Plot (t, Signal);
title ( 'Time Domain Signal');
Xlabel ('Time (s)');
y Label ('Amplitude');
n = length (signal );
f = (0: n-1) * (fs/n);
Spectrum = fft (signal );
Subplot (2, 1, 2);
Plot (f, abs (spectrum)/n );
title ('Frequency Spectrum');
Xlabel ('Frequency (Hz)');
y label ('Magnitude');
Xlim ( [0 fs/2 ]);
```

7)c) Explain the generation and Detection of PPM waves with a relevant block diagram.
(Block dgm - 2M, w/f - 2M, Expl'n - 2M) [06 Marks]

Soln: <u>Generation of PPM wave:</u>






PPM Equation :-

$$s(t) = \sum_{n=-\infty}^{\infty} g(t - nT_s - k_p m(nT_s))$$ may be generated using the above block diagram

→ The message signal m(t) is first converted into a PAM signal by means of a sample and hold circuit, generating a staircase waveform u(t).

→ Note that pulse duration T of the sample and hold circuit is the same as the sampling duration $T_s$.

→ From the figure below, we can say that combined signal v(t) is applied to a threshold detector that produces a very narrow pulse each time v(t) crosses a zero in the negative going direction.

→ Finally the PPM signal s(t) is generated by using this sequence of impulses to excite a filter whose impulse response is defined by the standard pulse g(t).

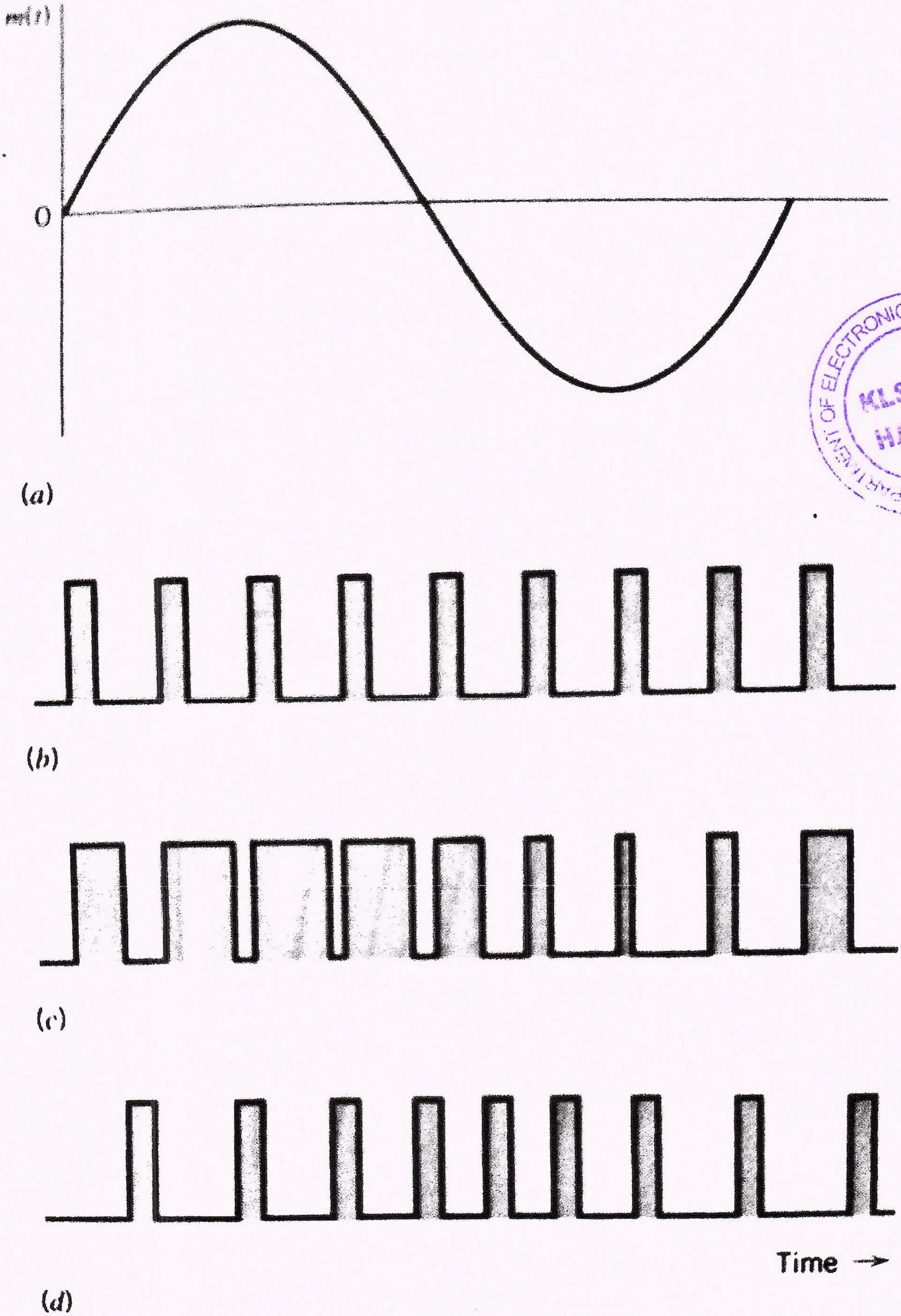





**FIGURE 7.10** Illustrating two different forms of pulse-time modulation for the case of a sinusoidal modulating wave. (*a*) Modulating wave. (*b*) Pulse carrier. (*c*) PDM wave. (*d*) PPM wave.

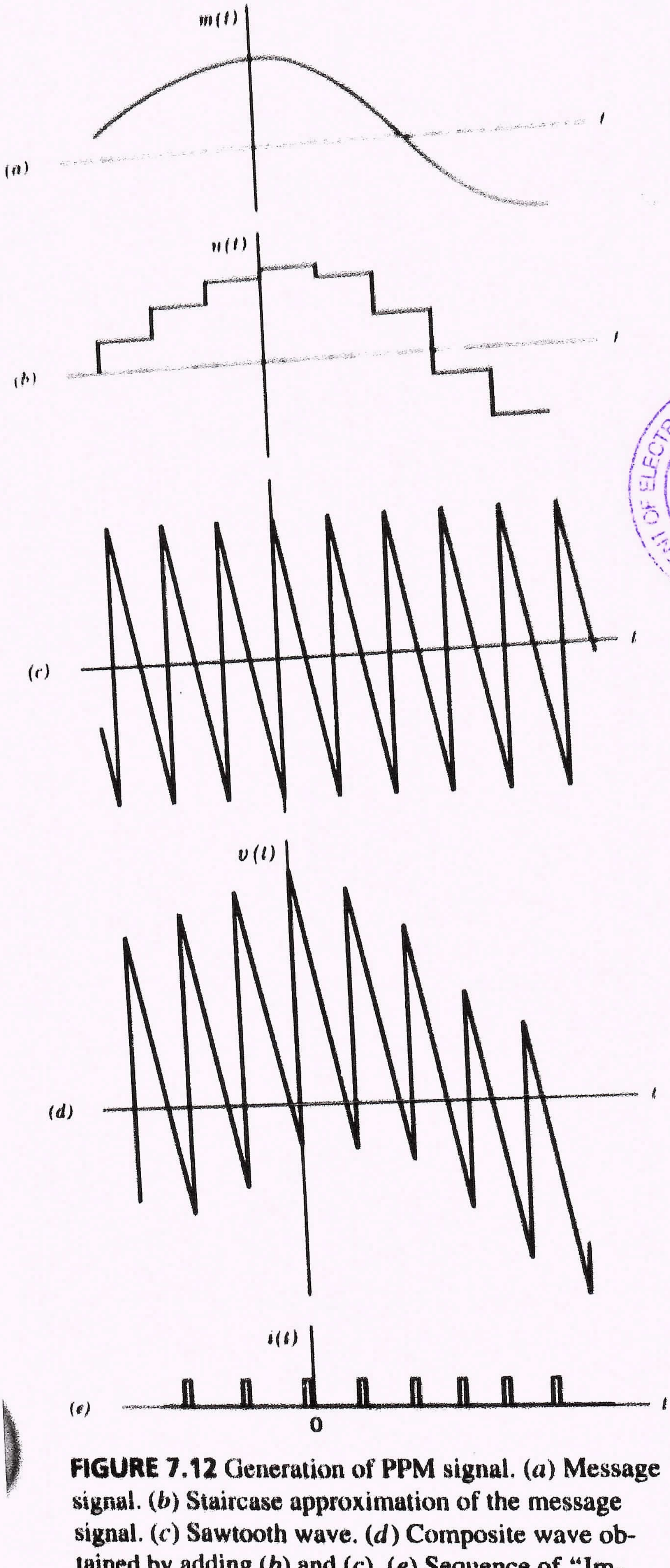



**FIGURE 7.12** Generation of PPM signal. (*a*) Message signal. (*b*) Staircase approximation of the message signal. (*c*) Sawtooth wave. (*d*) Composite wave obtained by adding (*b*) and (*c*). (*e*) Sequence of "Impulses" used to generate the PPM signal.

## Detection of PPM waves

The operation of one type of PPM receiver may proceed as follows.

i) Convert the received PPM wave into a PDM or PWM wave with same modulation.

ii) Integrate this PDM wave using a device with a finite integration time, thereby computing the area under each pulse of PDM wave

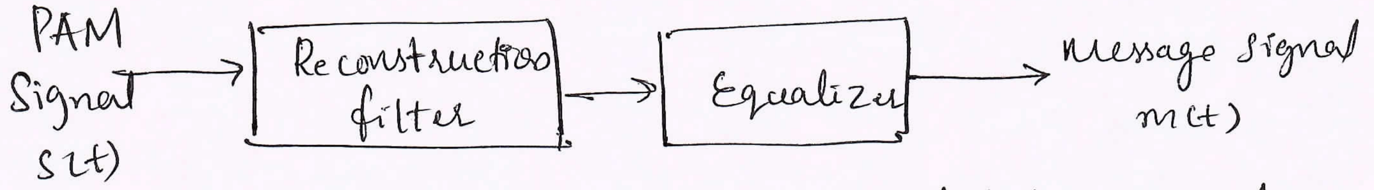
iii) Sample the o/p of the integrator at a uniform rate to produce a PAM wave, whose pulse amplitudes are proportional to the signal samples $m(nT_s)$ of the original PPM wave $s(t)$.

iv) Finally demodulate the PAM wave to recover the message signal $m(t)$

8) a) What is aperture effect in PAM systems? How can it be minimized?

[Aperture – 2M, Remedy – 2M] [04 Marks]

Soln: The distortion caused by the use of pulse amplitude modulation to transmit an analog information bearing signal is referred to as the Aperture effect.




This distortion may be corrected by connecting an equalizer in cascade with the low pass reconstruction filter as shown above.

* Equalizer has the effect of decreasing the in band loss of the reconstruction filter as the frequency increases in such a manner as to compensate for the aperture effect.

Amplitude response of Equalizer is given by

$$\frac{1}{|H(f)|} = \frac{1}{T\ \mathrm{Sinc}(T)} = \frac{\pi f}{\sin(\pi f T)}$$



8) b) What is multiplexing? and why it is required in Communication? Explain the working of TDM with a neat block diagram.

(Defn – 1, Need – 2, TDM-Block – 3, Expln – 4M) [10 Marks]

Soln: Multiplexing is a method by which multiple signals are combined into one signal over a shared medium.

Need for Multiplexing:

1) to enable network devices to communicate with each other

2) to better utilize scarce or expensive network resources.

## Time division Multiplexing (TDM) Systems :-

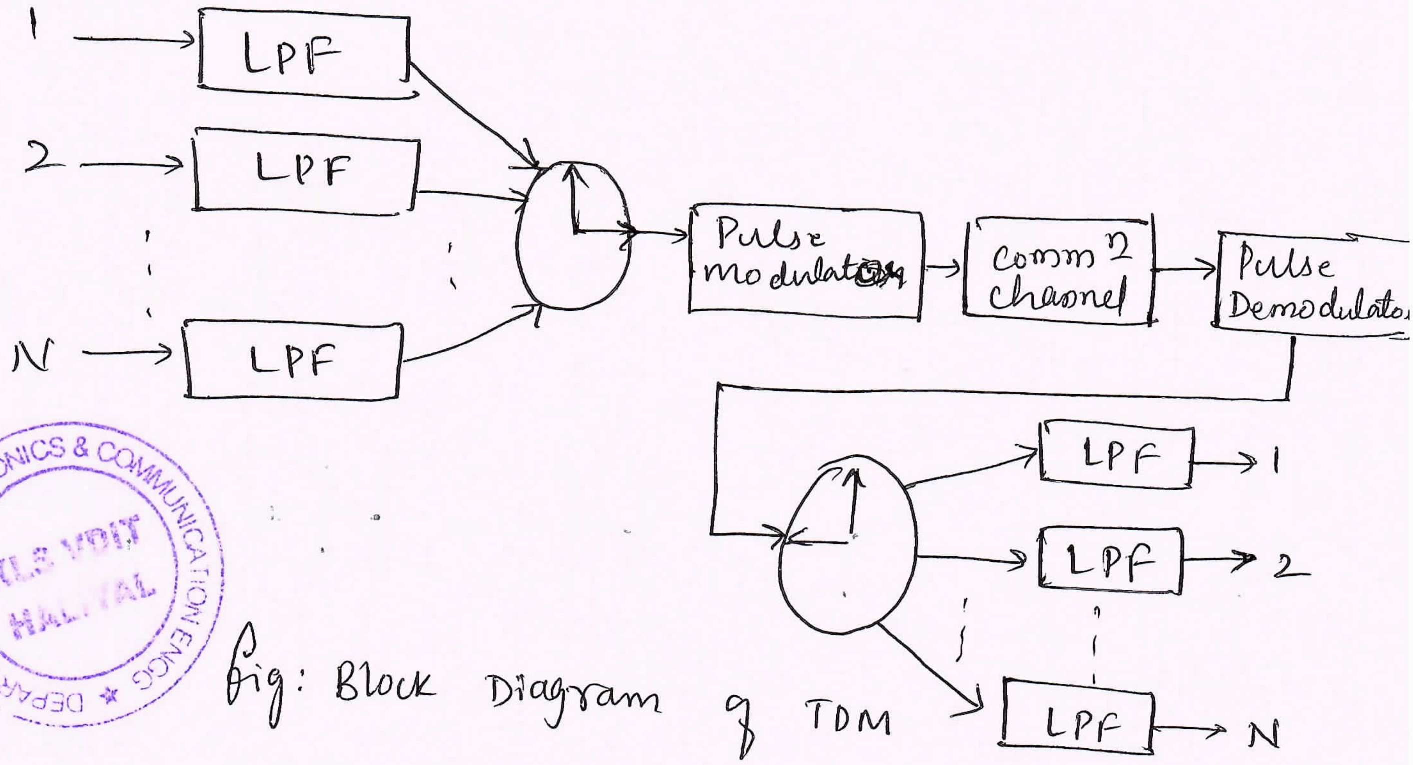





Fig: Block Diagram of TDM

→ Each i/p message signal is first restricted in band-width by a LPF to remove the frequencies that are non essential to an adequate signal representation

→ The low pass filter O/ps are then 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

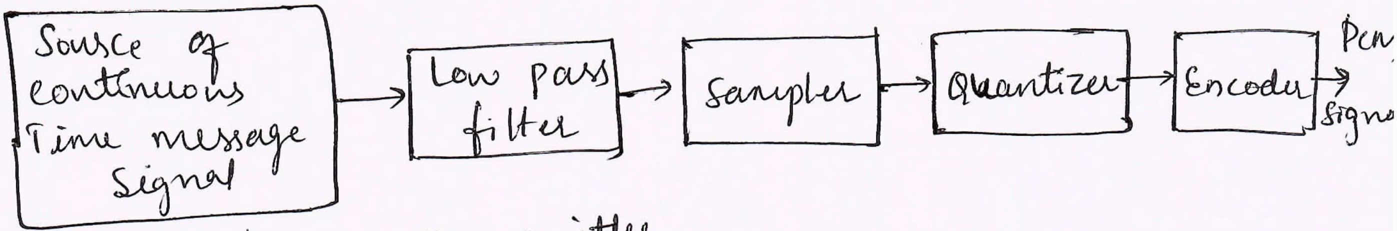
→ The narrow samples produced at the pulse demodulator are distributed to the appropriate LPF by means of a decommutator, which operates in synchronism with the commutator in the transmitter.

→ This synchronization is essential for satisfactory operation of the system.

8) c) Explain the basic elements of a PCM system with neat diagrams. [06 Marks]

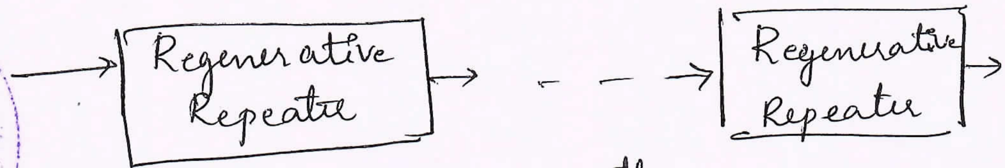
[Block dgm – 3M, Expln 3M]

Soln:

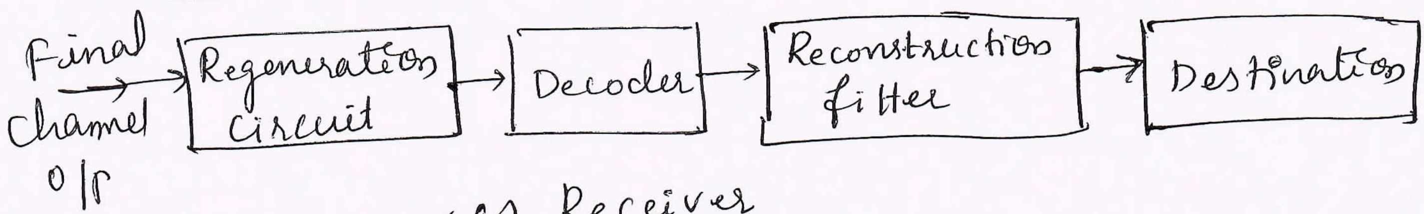



(a) Transmitter






(b) Transmission path




(c) Receiver

The block diagram shows the process of Pulse Code Modulation (PCM).

The basic elements of a PCM system are:

Sampling –

The incoming message signal is sampled with a train of narrow rectangular pulses so as to closely approximate the instantaneous sampling process.

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.



Encoding:

In combining the process of sampling and quantizing, the specifications of a continuous message signal becomes limited to a discrete set of values, but not in the form best suited for transmission over a line.

Decoding -

The first operation in the receiver is to regenerate the received pulses one last time. These clean pulses are then regrouped into codewords and decoded into a quantized PAM signal.

Filtering:

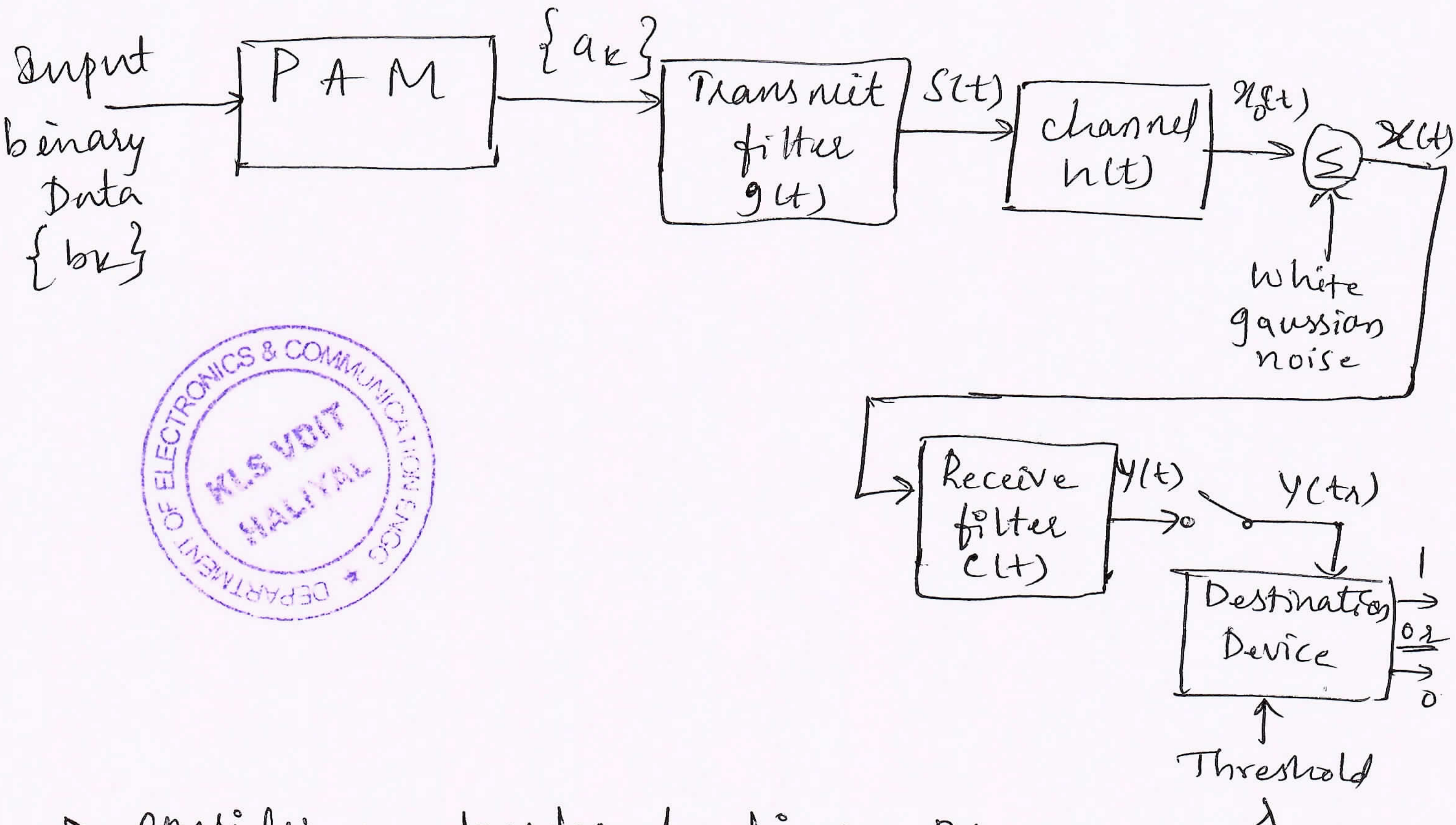
The final operation in the receiver is to recover the message signal wave by passing the decoder O/P through a low pass reconstruction filter whose cut off frequency is equal to the message signal bandwidth, W.

Multiplexing:

In applications of 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.

9) a) Define Intersymbol Interference (ISI). Outline baseband binary data transmission system with neat block diagram and equations. [08 Marks]
[Defn – 2M, Block – 3M, Expln – 3M]

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




→ Consider a baseband binary PAM System, a generic form of which is shown in the above above figure.

→ The incoming binary sequence $\{b_k\}$ consists of symb-ols 1 and 0, each of duration $T_b$.

→ 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

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

The sequence of short pulses so produced is applie 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 tran -mission 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 resultin filter o/p $y(t)$ is sampled synchronously with the transmitter, with the sampling instants being determ -ined by clock or timing signal that is usually extracted from the receive filter. o/p.

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

→ If threshold $\lambda$ is exceeded, a decision is made in favor of symbol 1. If threshold $\lambda$ is not exceeded, a decision is made in favour of symbol 0.

)b) Develop a code to generate and plot eye diagram? (Program - 6M) [06 Marks]

oln:

```
numBits = 1000;
bitRate = 1e3;
SamplesPerBit = 100;
data = randi([0, 1], 1, numBits);
nrzSignal = repelem(data, SamplesPerBit);
noisySignal = nrzSignal + 0.1 * randn(1, length(nrzSignal));
numUI = 2;
WindowSize = numUI * SamplesPerBit;
```

```
figure;
hold on;
for i = 1:windowSize:(length(noisySignal) - windowSize)
Plot (noisySignal(i:i+WindowSize -1));
end.
title ('Eye Diagram');
xlabel('Time (samples)');
ylabel('Amplitude');
grid on;
hold off;
```

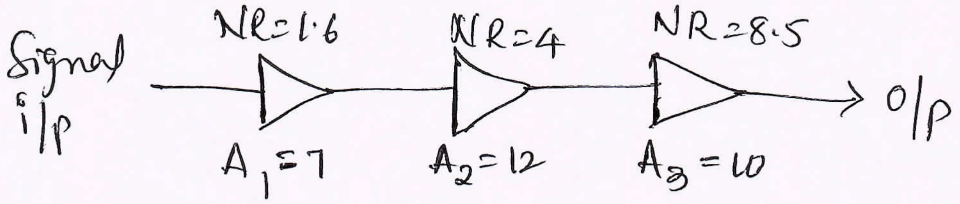


9)c) Illustrate the concept of noise in cascaded stages with a diagram. Write Frii's formula and mention its terms.
(Noise - 4M Friis - 2M] [06 Marks]

Soln: Noise has its greatest effect at the input to a receiver simply because that is the point at which the signal level is lowest.

* The noise ~~temperature~~ performance of a receiver is invariably 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.




NR = 2.12 NF = 3.26 dB

The formula used to calculate the overall noise Performance of a receiver or of multiple Stages of RF amplification, called Frii's formula.

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

$$\therefore NR = 1.6 + \frac{4-1}{7} + \frac{8.5-1}{7 \times 12}$$

$$\boxed{NR = 2.12}$$

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Noise figure, $NF = 10 \log NR$

$= 10 \log 2.12$

$NF = 3.26\,dB$

10) a) Explain the following concepts briefly.

i) Nyquist criterion for distortionless transmission.

ii) Baseband M-ary PAM transmission.

[(i) — 4 M (ii) — 4 M] [08 Marks]

Ans: i) Nyquist Criterion for distortion less transmission

In practice, we typically find that the transfer function of a channel and the transmitted pulse shape are specified, and the problem is to determine the transfer function of the transmit and receive filters so as to reconstruct the original binary data sequence $\{b_k\}$.

→ The receiver does this by extracting and then decoding the corresponding sequence of co-efficients $\{a_k\}$ from o/p $y(t)$.

The decoding requires that the weighted pulse contribution $a_k P(iT_b - KT_b)$ for $K=i$ be free from ISI due to the overlapping tails of all other weighted pulse contributions represen by $K \neq i$.

→ We control the overall pulse $p(t)$, as shown by,

$$P(iT_b - KT_b) = \begin{cases} 1 & i = K \\ 0 & i \neq K \end{cases}$$

→ $y(t_i)$ satisfies the eqn $y(t_i) = \mu a_i$ for all $i$

Consider the sequence of samples $\{p(nT_b)\}$, where $n = 0, \pm 1, \pm 2, ----$

$$P_\delta(f) = R_b \sum_{n=-\infty}^{\infty} P(f - nR_b)$$

where $R_b = 1/T_b$

→ $$P_\delta(f) = \int_{-\infty}^{\infty} \sum_{m=-\infty}^{\infty} [p(mT_b)\,\delta(t - mT_b)] \exp(-j2\pi f t)\,dt$$

Let $m = i - K$

when $i = K$

$m = 0$.

So $$P_\delta(f) = \int_{-\infty}^{\infty} p(0)\,\delta(t) \exp(-j2\pi f t)\,dt$$

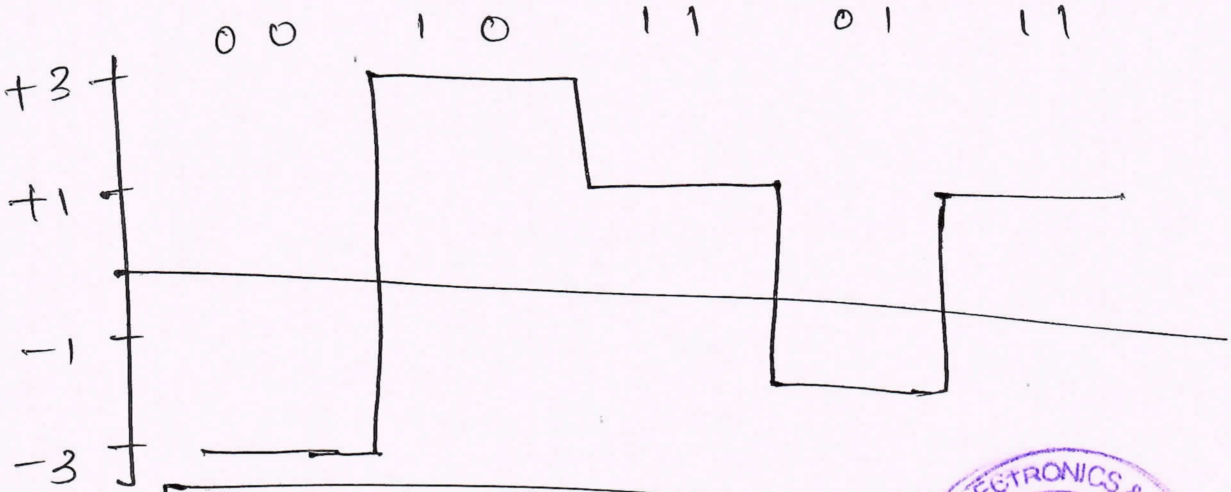
$$= P(0)$$

where we have made use of the shifting property

$$\sum_{n=-\infty}^{\infty} P(f - nR_b) = T_b \quad \text{— (1)}$$

## ii) Baseband M-ary PAM Transmission

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 |
| 11 | +1 |
| 10 | +3 |



→ On the other hand, in a baseband M-ary PAM system, 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 0010110 -111.

→ 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 inform at the rate of $1/T_b$ bits per second.

→ In an M-ary PAM system, one baud is equal to $\log_2 M$ bits per second and the symbol duration T of the M-ary PAM system is related to the bit duration $T_b$ of the equivalent binary PAM system as

$$T = T_b \log_2 M.$$

10) b) Develop a code to generate NRZ and RZ pulse. [Program - 06 Marks]. [06 Marks]

Soln:

```
numBits = 10;
bitRate = 1e3;
SamplesPerBit = 100;
data = randi ([0,1], 1, numBits);
t = 0:1/SamplesPerBit : numBits - 1/SamplesPerBit;

% NRZ Signal
  nrzSignal = repelem (data, SamplesPerBit);
% RZ Signal
  rzSignal = zeros (1, numBits * SamplesPerBit);
```

```
for (i = 1 : numBits
    rzSignal ((i-1) * samplesPerBit + 1 : (i-1) *
    samplesPerBit + samplesPerBit /2) = data (i);
end.
Subplot (2,1,1);
Stairs (t, nrzSignal, 'linewidth', 2);
ylim ([-0.5 1.5]);
title ('NRZ line code');
xlabel ('Time');
ylabel ( 'Amplitude');
grid on;

% Subplot RZ
 Subplot (2, 1, 2);
Stairs (t, rzsignal, 'Linewidth',2);
ylim ([-0.5 1.5]);
title ('RZ line code');
xlabel ( 'Time');
ylabel ('Amplitude');
grid on;
```



10) c) Define Signal to Noise Ratio (SNR). Explain the different types of external and internal noise [Defn - 2M, External - 2M, Internal - 2M] [06 Marks]

Soln: Signal to noise ratio (SNR) indicates relative strengths of the signal and the noise in a communication system.

The stronger the signal, weaker the noise, the higher the S/NR.

## External noise :-

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

i) Industrial noise
ii) Atmospheric noise
iii) Extra terristrial noise.



## Industrial noise

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

## Extraterristrial noise

Extra terristrial, solar and cosmic comes from source in space.

~~Thermal noise~~

## Internal noise :

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

## Thermal noise

The most internal noise is caused by a phenomenon known as 'thermal agitation', a random movement of free electrons in a conductor or caused by heat.

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Semiconductor noise:

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

Intermodulation noise:

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