CBCS SCHEME

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BEC402

Fourth Semester B.E./B.Tech. Degree Examination, June/July 2025
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.	Define Probability. Illustrate the relationship between sample space, events and probability.	6	L2	CO5
	b.	What are moments? Determine the characteristic function of a Gaussian random variable with a given mean and variance.	6	L2	CO5
	c.	Analyze the Gaussian process with Gaussian distribution curve. Infer the properties of a Gaussian process.	8	L2	CO5
		OR			
Q.2	a.	Define a random process. Interpret mean and covariance function with respect to stationary random process.	6	L2	CO5
	b.	What is Autocorrelation function? State and prove the properties of Autocorrelation function.	6	L2	CO5
	c.	Analyze the PDF and CDF of a random experiment in which three coins are tossed and condition to get random variable is getting head.	8	L3	CO5
		Module – 2			
Q.3	a.	Define Amplitude modulation. Derive an expression for Amplitude Modulation in time domain with necessary waveforms.	8	L2	CO1
	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 side bands and the total bandwidth occupied by the AM station.	5	L3	CO1
	c.	Outline the block diagram of FDM transmitter. List the applications of FDM.	7	L2	CO1
		OR			
Q.4	a.	Develop a code to generate Amplitude Modulation Waveforms and display its spectrum.	8	L3	CO1
	b.	Apply the concept of side bands to explain DSB and SSB, draw the relevant waveforms.	5	L2	CO1
	c.	Explain with diagrams, the working principle of Lattice-type balanced modulator.	7	L2	CO1
		Module – 3			
Q.5	a.	Identify a method used to convert a Phase Modulated (PM) signal into a Frequency-Modulated (FM) signal.	6	L2	CO3
	b.	The input to an FM receives has S/N of 2.8. The modulating frequency is 1.5 kHz. The maximum permitted deviation is 4 kHz. Determine (i) The frequency deviation caused by the noise and (ii) The improved output S/N.	6	L3	CO2

	c.	Interpret with a neat circuit diagram, the working principle of frequency modulation of a crystal oscillator with a Voltage Variable Capacitor (VVC).	8	L2	CO2
		OR			
Q.6	a.	Define Modulation. Identify any five differences between Frequency Modulation and Amplitude Modulation.	6	L2	CO2
	b.	Why Pre-emphasis and de-amphasis are required? Explain how they are implemented?	6	L2	CO2
	c.	Draw the block diagram of a super heterodyne receiver and explain the function of each.	8	L2	CO2
		Module – 4			
Q.7	a.	State and prove sampling theorem. Write a program for sampling and reconstruction of low pass signals and display the signals and its spectrum.	10	L3	CO3
	b.	Infer the working of TDM system with a neat block diagram.	5	L2	CO3
	c.	Explain briefly the block diagram of PPM generator.	5	L2	CO3
		OR			
Q.8	a.	Identify and explain the basic elements of a PCM system with neat diagrams. For the data stream [0 1 1 0 1 0 0 1], draw the following line code waveforms : (i) Unipolar NRZ (ii) Polar NRZ (iii) Unipolar RZ (iv) Bipolar RZ (v) Manchestor code	10	L3	CO3
	b.	Infer the advantages of digital signals over analog signals.	5	L2	CO3
	c.	Explain briefly the midtread and midrise Quantizers with relevant figures.	5	L2	CO3
		Module – 5			
Q.9	a.	What is Intersymbol Interference (ISI)? With a neat block diagram outline the baseband binary data transmission system and write the necessary equations?	8	L2	CO4
	b.	Define SNR. Summarize the different types of external and internal noise.	7	L2	CO4
	c.	Illustrate the concept of Noise in cascaded stages with a diagram. Write Friis formula and mention its terms.	5	L2	CO4
		OR			
Q.10	a.	What is Baseband digital transmission? Explain the following concepts briefly : (i) Nyquist criterion for distortionless transmission. (ii) Baseband M-ary PAM transmission.	8	L2	CO4
	b.	Define Noise. Classify the different types of semiconductor noise.	7	L2	CO4
	c.	What is Noise Factor and Noise Figure? An RF amplifier has an S/N ratio of 8 at the input and an S/N ratio of 6 at the output. Calculate the Noise factor and Noise figure.	5	L2	CO4

* * * * *

1a) Define probability illustrate the relationship between sample space events and probability?

Definition of Probability

Probability is a measure of the likelihood that a particular event will occur. It is a number between 0 and 1, where:

- 0 indicates an **impossible** event.
- 1 indicates a **certain** event.

Mathematically, the probability of an event **E** occurring is given by:

P(E)=Number of favorable outcomesTotal number of outcomes in the sample spaceP(E) = \frac{\text{Number of favorable outcomes}}{\text{Total number of outcomes in the sample space}}P(E)=Total number of outcomes in the sample spaceNumber of favorable outcomes

Sample Space, Events, and Probability – Relationship

1. **Sample Space (S)**:
   The **sample space** is the set of **all possible outcomes** of an experiment.
   Example: In tossing a coin once,
   S={H,T}S = \{H, T\}S={H,T}
2. **Event (E)**:
   An **event** is any **subset of the sample space**, including single outcomes or combinations.
   Example: Getting a head when tossing a coin →
   E={H}E = \{H\}E={H}
3. **Probability of an Event**:
   The **probability** of an event is calculated using the ratio:

   P(E)=n(E)n(S)P(E) = \frac{n(E)}{n(S)}P(E)=n(S)n(E)

   Where:

   - n(E)n(E)n(E): Number of favorable outcomes (size of event set)
   - n(S)n(S)n(S): Number of outcomes in sample space

Illustration Example

Experiment: Roll a fair 6-sided die

- Sample Space:
  S={1,2,3,4,5,6}S = \{1, 2, 3, 4, 5, 6\}S={1,2,3,4,5,6}
- Event: Getting an even number
  E={2,4,6}E = \{2, 4, 6\}E={2,4,6}
- Probability:

  P(E)=n(E)n(S)=36=0.5P(E) = \frac{n(E)}{n(S)} = \frac{3}{6} = 0.5P(E)=n(S)n(E)=63
  =0.5

b. What are moments? Determine the characteristic function of a Gaussian random variable with given mean and variance?

What are Moments?

In probability theory and statistics, **moments** are quantitative measures related to the **shape of a probability distribution**. They provide information such as central tendency, dispersion, skewness, and kurtosis.

Types of Moments:

1. **First Moment (Mean):**

$$\mu = E[X]$$

Represents the **average** value or central location.

2. **Second Central Moment (Variance):**

$$\mathrm{Var}(X) = E[(X - \mu)^2]$$

Measures the **spread or dispersion** around the mean.

3. **Third Central Moment (Skewness):**

$$E[(X - \mu)^3]$$

Describes the **asymmetry** of the distribution.

4. **Fourth Central Moment (Kurtosis):**

$$E[(X - \mu)^4]$$

Measures the "**tailedness**" of the distribution.

Characteristic Function of a Gaussian Random Variable

Let $X \sim \mathcal{N}(\mu, \sigma^2)$

(i.e., X is a Gaussian random variable with mean μ and variance σ^2).

Characteristic Function Definition:

The characteristic function $\phi_X(t)$ of a random variable X is defined as:

$$\phi_X(t) = E[e^{itX}]$$

For a Gaussian Random Variable:

$$\phi_X(t) = \exp\left(it\mu - \frac{1}{2}\sigma^2 t^2\right)$$

This function uniquely determines the distribution of X and can be used to derive all moments of the distribution by taking derivatives.

Example:

If $X \sim \mathcal{N}(2, 4)$ (i.e., mean = 2, variance = 4), then:

$$\phi_X(t) = \exp\left(it \cdot 2 - \frac{1}{2} \cdot 4 \cdot t^2\right) = \exp\left(2it - 2t^2\right)$$

c. Analyze the Gaussian process with Gaussian distribution curve infer the properties of a Gaussian process.

Analysis of Gaussian Process with Gaussian Distribution Curve

✅ 1. Gaussian (Normal) Distribution Curve

A **Gaussian distribution**, also called a **normal distribution**, is a symmetric bell-shaped curve defined by:

$$f(x) = \frac{1}{\sqrt{2\pi\sigma^2}} \exp\left(-\frac{(x-\mu)^2}{2\sigma^2}\right)$$

Where:

- μ = mean (center of the curve),
- σ^2 = variance (controls the spread),
- The curve is symmetric about the mean.

✅ 2. Gaussian Process (GP)

A **Gaussian Process** is a collection of random variables, any finite number of which have a **joint multivariate Gaussian distribution.**

It is not just a distribution over variables — it's a **dist tion over functions.**

Definition:

A process $\{X(t) : t \in T\}$ is called a **Gaussian process** if for every finite set $t_1, t_2, \ldots, t_n \in T$, the random vector $(X(t_1), \ldots, X(t_n))$ follows a **multivariate normal distribution.**

Properties of a Gaussian Process

Property	Description
Mean Function	$m(t) = E[X(t)]$
Covariance Function	$k(t, s) = E[(X(t) - m(t))(X(s) - m(s))]$
Fully Characterized	A Gaussian process is completely defined by its mean and covariance functions.
Linear Operations	Any linear transformation of a Gaussian process is also Gaussian.
Stationarity	If the covariance function depends only on the time difference (
Smoothness	The smoothness of the ple functions depends on the covariance kernel. For example, the squared exponential kernel gives very smooth functions.

Visual Interpretation Using the Gaussian Curve

- At a fixed point t, $X(t) \sim \mathcal{N}(\mu, \sigma^2)$
- As you consider many points $t_1, t_2, \ldots$, the joint distribution across those points forms a **multivariate normal**.
- Each curve generated from a Gaussian process is a **random function** sampled from this distribution.

Inference from the Gaussian Distribution Curve

- The curve's **peak** represents the most probable value (mean).
- The **spread** shows uncertainty (variance).
- **Confidence intervals** (e.g., 68%, 95%, 99%) correspond to ±1σ, ±2σ, and ±3σ from the mean in the curve.

Application Areas

- Machine learning: Gaussian Process Regression (GPR)
- Signal processing and control systems
- Time series modeling
- Geostatistics (Kriging)

2a. Define random process. Interpret mean and covariance function with respect to stationary random process?

Definition of Random Process

A **random process** (also called a **stochastic process**) is a collection of random variables indexed by time or space. It describes how a random variable evolves over time.

$$\{X(t),\ t \in T\}$$

Where:

- $X(t)$: Random variable at time t
- T: Index set (e.g., time)

Mean Function of a Random Process

The **mean function** of a random process $X(t)$ is the expected value of the process at time t:

$$m_X(t) = E[X(t)]$$

This function gives the **average behavior** of the process at each time instant.

Covariance Function of a Random Process

The covariance function (or autocovariance function) between times t_1 and t_2 is defined as:

$$C_X(t_1, t_2) = E\left[(X(t_1) - m_X(t_1))(X(t_2) - m_X(t_2))\right]$$

It measures the degree of linear dependence between the values of the process at different times.

Interpretation with Respect to a Stationary Random Process

A stationary random process is one whose statistical properties do not change with time.

For a **strictly stationary** process:

- The joint distributions are invariant under time shifts.

For a **wide-sense stationary (WSS)** process:

- Mean is constant:

$$m_X(t) = \mu \quad (\text{a constant})$$

- Covariance depends only on time difference $\tau = t_2 - t_1$:

$$C_X(t_1, t_2) = C_X(\tau)$$

Summary of WSS Properties

Function	Property
Mean	Constant over time: $m_X(t) = \mu$
Covariance	Depends only on lag: $C_X(t_1, t_2) = C_X(\tau)$
Variance	Also constant: $\sigma^2 = C_X(0)$

Example

Let $X(t)$ be a random process representing noise in an electrical signal:

- If it's WSS:
  - The average noise level is constant: $m_X(t) = 0$
  - The similarity between signal values at different times depends only on how far apart the times are, not their absolute values.

B. what is autocorrelation function? state and prove the properties of autocorrelation function?

✅ What is the Autocorrelation Function?

The **autocorrelation function** of a random process measures the **correlation** between the values of the process at different times. It indicates how the process is related to itself over time.

Mathematical Definition:

For a random process $X(t)$, the **autocorrelation function** is defined as:

$$R_X(t_1, t_2) = E[X(t_1) \cdot X(t_2)]$$

For a **wide-sense stationary** (WSS) process, the autocorrelation depends only on the **time difference** $\tau = t_2 - t_1$:

$$R_X(\tau) = E[X(t) \cdot X(t+\tau)]$$

✅ Properties of the Autocorrelation Function

Let's now **state and prove** the main properties of the autocorrelation function $R_X(\tau)$:

◆ 1. **Symmetry:**

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

Proof:

$$R_X(\tau) = E[X(t) \cdot X(t+\tau)] = E[X(t+\tau) \cdot X(t)] = R_X(-\tau)$$

◆ 2. **Maximum at Zero Lag:**

$$|R_X(\tau)| \le R_X(0)$$

Proof:

Using the Cauchy–Schwarz inequality:

$$|E[X(t)X(t+\tau)]| \le \sqrt{E[X^2(t)] \cdot E[X^2(t+\tau)]}$$

Since the process is WSS:

$$E[X^2(t)] = E[X^2(t+\tau)] = R_X(0) \Rightarrow |R_X(\tau)| \le R_X(0)$$

◆ 3. **Even Function (for WSS):**

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

(Same as Symmetry – applies only for wide-sense stationary processes.)

◆ 4. **Non-negative Definite:**

For any real numbers $a_1, a_2, \ldots, a_n$ and time shifts $t_1, t_2, \ldots, t_n$:

$$\sum_{i=1}^{n}\sum_{j=1}^{n} a_i a_j R_X(t_i - t_j) \geq 0$$

This is a more advanced property that ensures the **autocorrelation matrix** is positive semi-definite.

◆ 5. **R_X(0) = Variance (for Zero-Mean Process):**

If $E[X(t)] = 0$, then:

$$R_X(0) = E[X^2(t)] = \mathrm{Var}(X(t))$$

✅ **Example:**

For white noise $X(t)$, where values at different times are uncorrelated:

$$R_X(\tau) = \begin{cases} \sigma^2, & \tau = 0 \\ 0, & \tau \neq 0 \end{cases}$$

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c. Analyze the PDF and CDF of random experiment in which three coins are tossed and condition to get random variable is getting head?

✅ 1. Random Experiment

- Tossing **three fair coins**
- Sample space $S = \{HHH, HHT, HTH, HTT, THH, THT, TTH, TTT\}$
- Total outcomes = $2^3 = 8$

✅ 2. Random Variable Definition

Let X = number of **Heads** obtained in one trial.

Then possible values of X:

$$X \in \{0, 1, 2, 3\}$$

✅ 3. Probability Distribution Function (PDF)

X (Heads)	Outcomes	Count	P(X = x)
0	TTT	1	$\frac{1}{8}$
1	HTT, THT, TTH	3	$\frac{3}{8}$
2	HHT, HTH, THH	3	$\frac{3}{8}$
3	HHH	1	$\frac{1}{8}$

This is the **Probability Mass Function (PMF)** for a **Binomial Distribution** with:

$$X \sim B(n = 3, p = \frac{1}{2})$$

So:

$$P(X = x) = \binom{3}{x} \left(\frac{1}{2}\right)^x \left(\frac{1}{2}\right)^{3-x}$$

✅ 4. Cumulative Distribution Function (CDF)

CDF $F(x) = P(X \leq x)$

x	F(x) = $P(X \leq x)$
< 0	0
0	$\frac{1}{8}$
1	$\frac{1}{8} + \frac{3}{8} = \frac{4}{8}$
2	$\frac{4}{8} + \frac{3}{8} = \frac{7}{8}$
3	$\frac{7}{8} + \frac{1}{8} = 1$
> 3	1

✅ 5. Summary

- **PDF:** Discrete probabilities at 0, 1, 2, 3 heads
- **CDF:** Step-wise increasing function that reaches 1 at $x = 3$
- **Distribution Type:** Binomial with parameters $n = 3, p = 0.5$

Q3. (a) Amplitude Modulation (AM) is a modulation technique where the amplitude of a high-frequency carrier wave is varied in proportion to the instantaneous amplitude of the message (modulating) signal, while the frequency and phase of the carrier remain constant.

- In amplitude modulation, the amplitude of the carrier wave is varied in proportion to that of the message signal being transmitted.
- Amplitude modulation is used when spectral efficiency and simplicity are most important.

Limitations: Noise sensitivity and lower power efficiency.

$$\boxed{v_m = Vm \sin 2\pi f_m t}$$

$$\boxed{v_C = Vc \sin 2\pi f_C t}$$

$$v_1 = V_c + v_m$$

$$= V_c + V_m \sin 2\pi f_m t$$

$$v_2 = v_1 \sin 2\pi f_c t$$

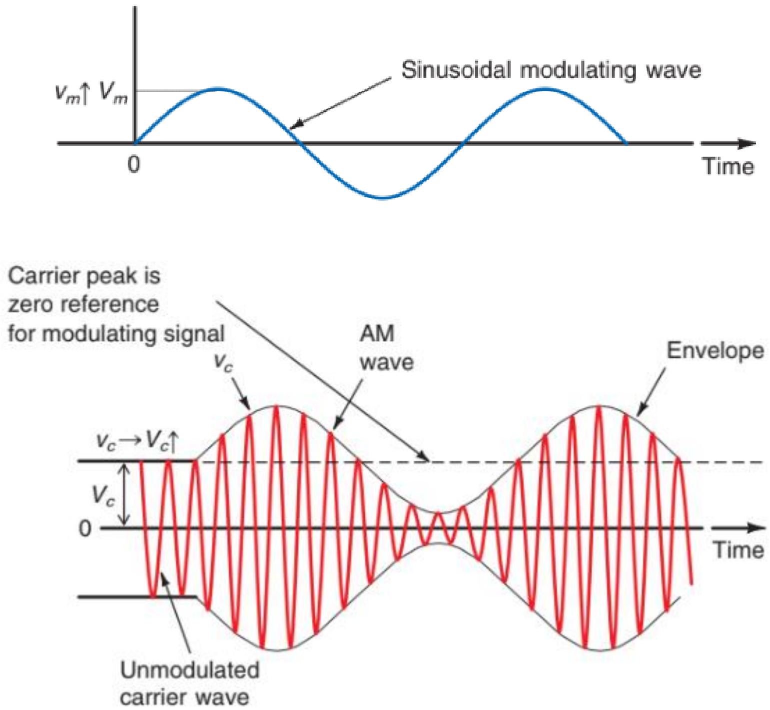
$$= V_c \sin 2\pi f_c t + (V_m \sin 2\pi f_m t)\,(\sin 2\pi f_c t)$$

$$v_{\mathrm{AM}} = V_c \sin 2\pi f_c t + \frac{V_m}{2} \cos 2\pi t(f_c - f_m) - \frac{V_m}{2} \cos 2\pi t(f_c + f_m)$$

Modulation Index μ determines depth of modulation:

- $0<\mu<1$: under-modulated
- $\mu=1$: fully modulated
- $\mu>1$: over-modulated (distortion)

 AM Bandwidth = 2fm



(b) Given fm=5 KHz

Fc=980 KHz

(i) Minimum USB: fc+0=980 KHz

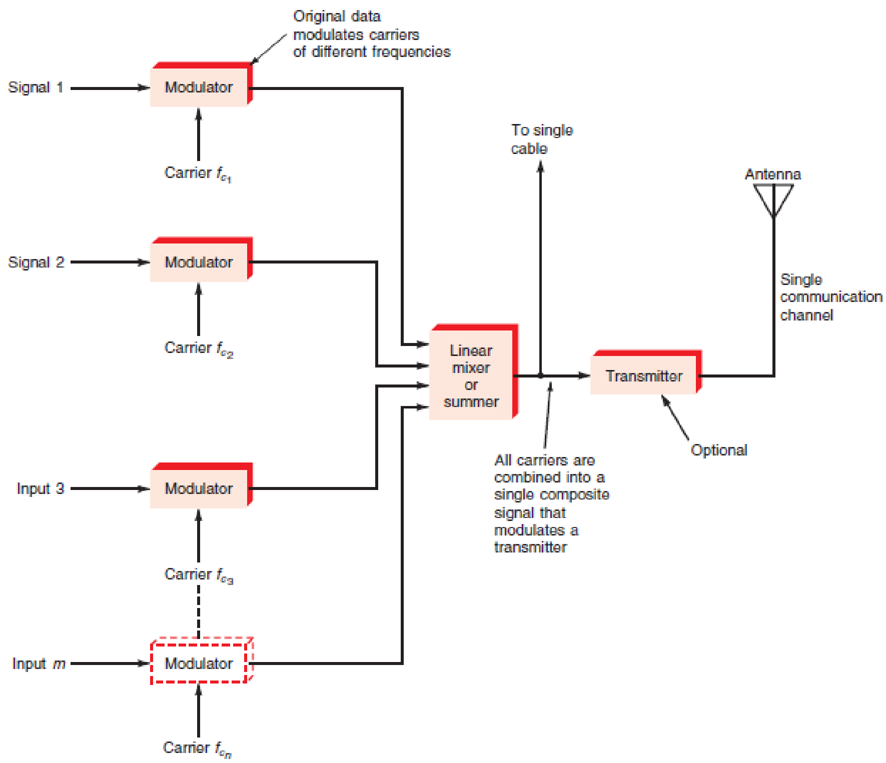
Maximum USB: fc+5=985 KHz

(ii) Minimum LSB: fc-5=975 KHz

Maximum LSB: fc-0=980 KHz

(iii) Total bandwidth=2*fm=2*5=10 KHz

(c) Block Diagram of FDM transmitter



Applications:

Frequency Division Multiplexing (FDM) is a technique where multiple signals are transmitted simultaneously over a common communication channel by allocating different frequency bands to each signal.

- Radio Broadcasting
- Television Broadcasting
- Aircraft and Marine Communication:
- Satellite Communication

Q. 4 (a)

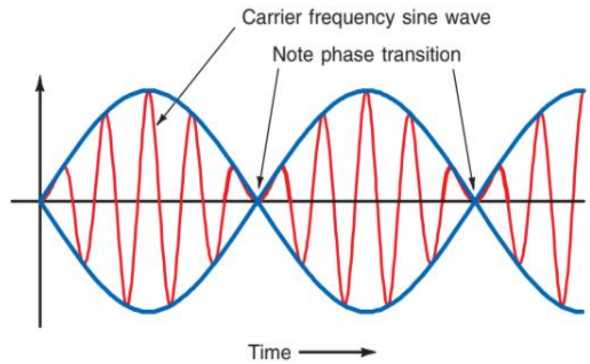
```
Ac = 2;          % Carrier amplitude
fc = 10;         % Carrier frequency (Hz)
Am = 1;          % Message amplitude
fm = 2;          % Message frequency (Hz)
Fs = 100;        % Sampling frequency (Hz)
duration = 1;    % Duration of the signal (seconds)
t = 0:1/Fs:duration; % Time vector

%% Generate Message Signal
mt = Am*cos(2*pi*fm*t);

%% Generate Carrier Signal
ct = Ac*cos(2*pi*fc*t);

%% Generate Amplitude Modulated Signal
modulated_signal = (1 + 0.5*mt).*ct;

%% Plot the signals
figure;
subplot(3,1,1);
plot(t, mt);
title('Message Signal');
xlabel('Time (s)');
ylabel('Amplitude');
grid on;

subplot(3,1,2);
plot(t, ct);
title('Carrier Signal');
xlabel('Time (s)');
ylabel('Amplitude');
grid on;

subplot(3,1,3);
plot(t, modulated_signal);
title('Amplitude Modulated Signal');
xlabel('Time (s)');
ylabel('Amplitude');
grid on;

%% Calculate and Plot the Spectrum
N = length(modulated_signal);
frequencies = Fs*(-N/2:N/2-1)/N;
spectrum = fftshift(fft(modulated_signal))/N;
```

```
figure;
plot(frequencies, abs(spectrum));
title('Magnitude Spectrum of Modulated Signal');
xlabel('Frequency (Hz)');
ylabel('Magnitude');
xlim([-2*fc 2*fc]); % Focus on relevant frequency range
grid on;
```

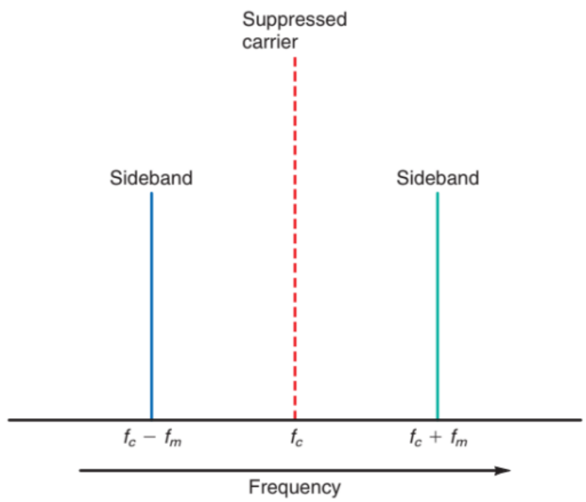
(b) **DSB Signals**

- The first step in generating an DSB signal is to suppress the carrier, leaving the upper and lower sidebands called as DSSC or DSB.
- Benefit is no power is wasted on the carrier.
- DSB signal is a sine wave at the carrier frequency, varying in amplitude.
- A unique characteristic of the DSB signal is the phase transitions that occur at the lower-amplitude portions of the wave.
- There are two adjacent positive-going half-cycles at the null points in the wave.

A time-domain display of a DSB AM signal.



A frequency-domain display of DSB signal.



SSB Signals

- In DSB transmission, since the sidebands are the sum and difference of the carrier and modulating signals, the information is contained in both sidebands.
- no reason to transmit both sidebands in order to convey the information.
- One sideband can be suppressed; the remaining sideband is called a single- sideband suppressed carrier (SSSC or SSB) signal.
- In DSB transmission, since the sidebands are the sum and difference of the carrier and modulating signals, the information is contained in both sidebands.
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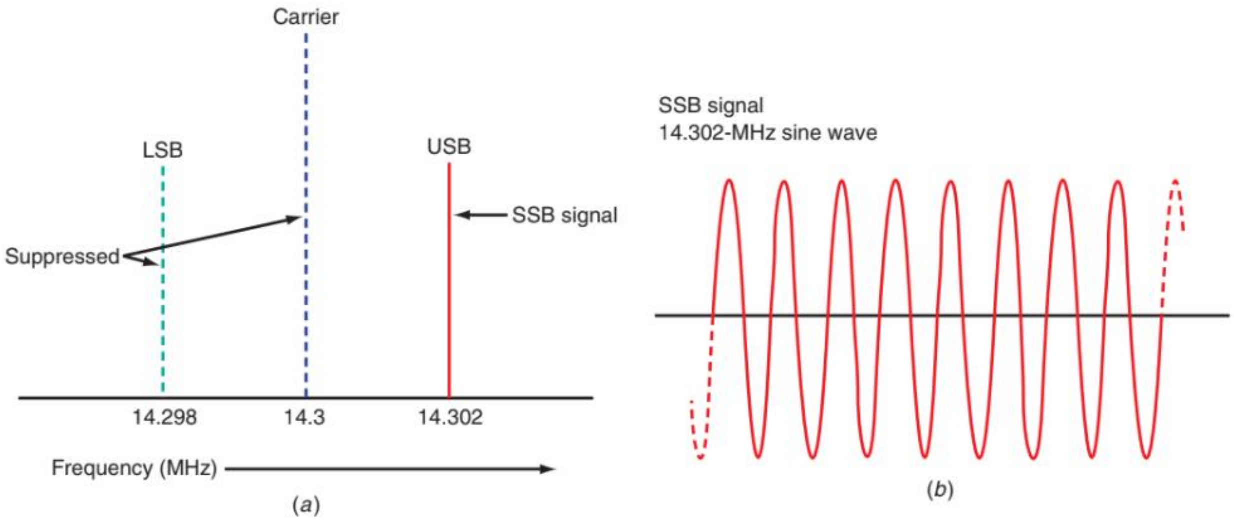
An SSB signal produced by a 2-kHz sine wave modulating a 14.3-MHz sine wave carrier.



(c) Lattice Type Balanced Modulator

- *Lattice modulator*: consists of an input transformer *T*1, an output transformer *T*2, and four diodes connected in a bridge circuit.
- The carrier signal is applied to the center taps of the input and output transformers, and the modulating signal is applied to the input transformer *T*1.
- The output appears across the secondary of the output transformer T2.
- 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 carrier turns the diodes off and on at a high 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.
- When the polarity of the carrier is positive, diodes *D*1 and *D*2 are forward-biased.
- At this time, *D*3 and *D*4 are reverse-biased and act as open circuits.
- As you can see, current divides equally in the upper and lower portions of the primary winding of *T*2.
- The current in the upper part of the winding produces a magnetic field that is equal and opposite to the magnetic field produced by the current in the lower half of the secondary.
- The magnetic fields thus cancel each other out. No output is induced in the secondary, and the carrier is effectively suppressed.
- Lattice modulator : consists of an input transformer T1, an output transformer T2, and four diodes connected in a bridge circuit.

- The carrier signal is applied to the center taps of the input and output transformers, and the modulating signal is applied to the input transformer T1.
- The output appears across the secondary of the output transformer T2.
- 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 carrier turns the diodes off and on at a high rate of speed, and the diodes act as switches that connect the modulating signal at the secondary of T1 to the primary of T2.
- When the polarity of the carrier is positive, diodes D1 and D2 are forward-biased.
- At this time, D3 and D4 are reverse-biased and act as open circuits.
- As you can see, current divides equally in the upper and lower portions of the primary winding of T2.
- The current in the upper part of the winding produces a magnetic field that is equal and opposite to the magnetic field produced by the current in the lower half of the secondary.
- The magnetic fields thus cancel each other out. No output is induced in the secondary, and the carrier is effectively suppressed.

Figure 4-24 Operation of the lattice modulator.

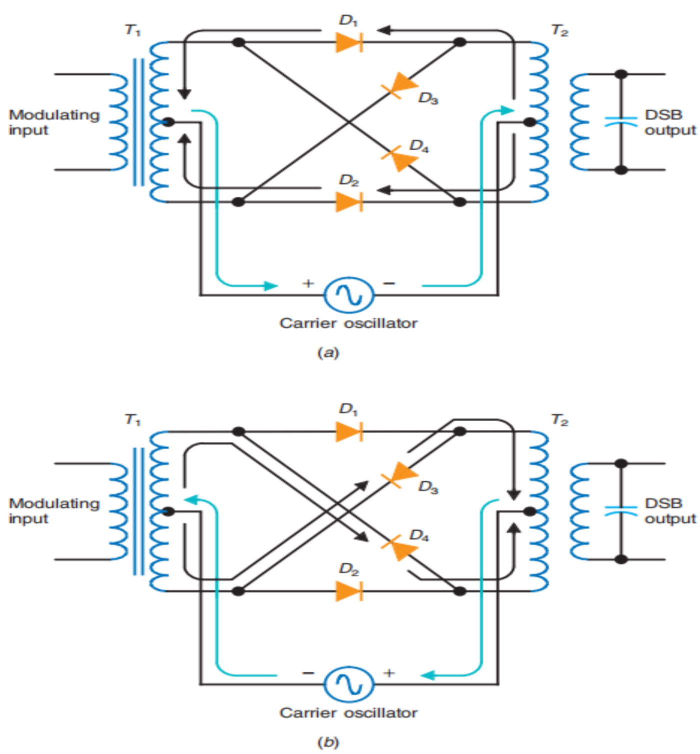
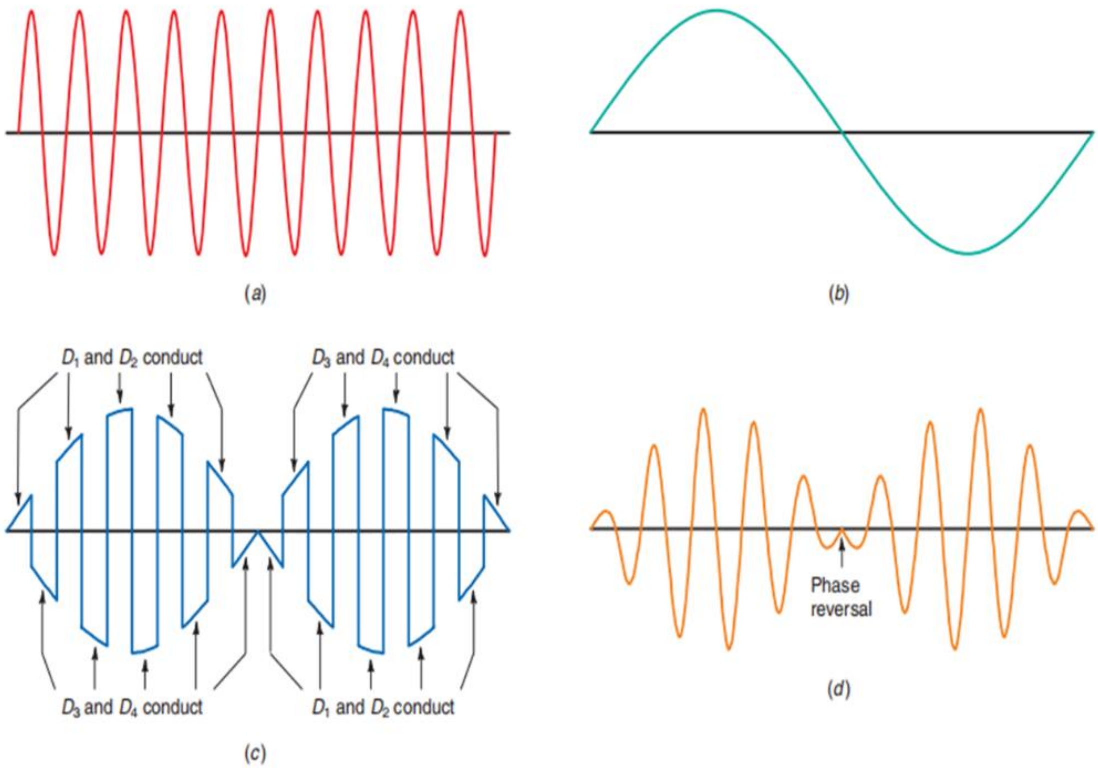


Figure 4-25 Waveforms in the lattice-type balanced modulator. (*a*) Carrier. (*b*) Modulating signal. (*c*) DSB signal—primary T_2. (*d*) DSB output.



Q5. (a)

A standard frequency-modulated wave is written in the form:

$$S_{FM}(t) = A\ cos\left(2\pi f_c t + 2\pi k_f \int_0^t m(t).\,dt\right)$$

where k_f is in Hz/V

A standard phase-modulated wave is written in the form:

$$S_{PM}(t) = A\ cos\,(2\pi f_c t + k_p m(t))$$

where k_p is in rad/V

Suppose, x(t) = $\int_0^t m(t).\,dt$

When x(t) is applied at the phase modulator we will get the following signal at the output:

$$S(t) = A\ cos\,(2\pi f_c t + k_p x(t))$$

putting the value of x(t), we get

$$S(t) = A\ cos\left(2\pi f_c t + k_p(\int_0^t m(t).\,dt)\right)$$

This signal represents a standard FM signal with

$$2\pi k_f = k_p$$

(b)

The input to an FM receiver has an S/N of 2.8. The modulating frequency is 1.5 kHz. The maximum permitted deviation is 4 kHz. What are (*a*) the frequency deviation caused by the noise and (*b*) the improved output S/N?

a. $\phi = \sin^{-1}\frac{N}{S} = \sin^{-1}\frac{1}{2.8} = \sin^{-1}0.3571 = 20.92^\circ \quad \text{or} \quad 0.3652 \text{ rad}$

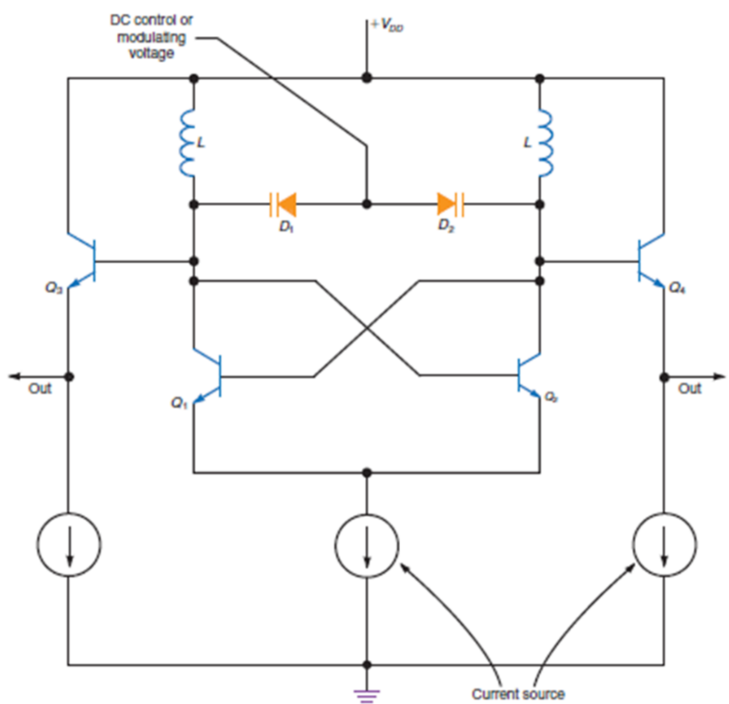
$$\delta = \phi(f_m) = (0.3652)(1.5 \text{ kHz}) = 547.8 \text{ Hz}$$

b. $\frac{N}{S} = \frac{\text{frequency deviation produced by noise}}{\text{maximum allowed deviation}} = \frac{547.8}{4000}$

$$\frac{N}{S} = 0.13695$$

$$\frac{S}{N} = \frac{1}{N/S} = 7.3$$

(c)



- Oscillators whose frequencies are controlled by an external input voltage are generally referred to as *voltage-controlled oscillators (VCOs)*.
- This circuit uses silicon-germanium (SiGe) bipolar transistor to achieve an operating frequency centered near 10 GHz.

- The oscillator uses cross - coupled transistors *Q*1 and *Q*2 in a multivibrator or flip-flop type of design.
- The signal is a sine wave whose frequency is set by the collector inductances and varactor capacitances.
- The modulating voltage, usually a binary signal to produce FSK, is applied to the junction of *D*1 and *D*2.
- Two complementary outputs are available from the emitter followers *Q*3 and *Q*4.
- In this circuit, the inductors are actually tiny spirals of aluminum (or copper) inside the chip, with inductance in the 500- to 900-pH range.
- The varactors are reverse-biased diodes that function as variable capacitors.
- The tuning range is from 9.953 to 10.66 GHz.
- CMOS type of circuit also uses a cross coupled *LC* resonant circuit design and operates in the 2.4- to 2.5-GHz range.
- Variations of it are used in Bluetooth transceivers and wireless LAN applications controlled over a wide range by an ac or dc input voltage.
- These VCOs typically have an operating range of less than 1 Hz to approximately 1 MHz.
- The output is either a square or a triangular wave rather than a sine wave.

6(a) Define Modulation and Differences Between FM and AM

Modulation is the process of varying a carrier signal in accordance with the information (message) signal. It enables transmission of signals over long distances by shifting their frequency range.

Five differences between Frequency Modulation (FM) and Amplitude Modulation (AM):

Feature	Amplitude Modulation (AM)	Frequency Modulation (FM)
Parameter varied	Amplitude of carrier	Frequency of carrier
Noise immunity	Poor	Better
Bandwidth	2 × message frequency	2 × (message frequency + deviation)
Signal quality	Lower (affected by noise)	Higher (less distortion)
Power efficiency	Less efficient	More efficient

6(b) Why Pre-emphasis and De-emphasis are Required

Pre-emphasis and De-emphasis are used in FM systems to improve the signal-to-noise ratio (SNR) of high-frequency components.

Why Required:

- - Noise in FM is more prominent at higher frequencies.
- - Human ears are sensitive to high-frequency noise.
- - Pre-emphasis boosts high frequencies before transmission.
- - De-emphasis attenuates them at the receiver to suppress noise.

Implementation:

- - Pre-emphasis: High-pass filter ($H(s) = 1 + sRC$)
- - De-emphasis: Low-pass filter ($H(s) = 1 / (1 + sRC)$)

Typical time constants: 75 µs (USA), 50 µs (Europe/India)

6(c) Block Diagram of Superheterodyne Receiver and Explanation

Block Diagram:

Antenna → RF Amplifier → Mixer → IF Filter → Demodulator → Audio Amplifier → Speaker

↓
Local Oscillator

Functions of Each Block:

- - Antenna: Captures the incoming RF signal.
- - RF Amplifier: Amplifies the weak RF signal.
- - Mixer: Combines RF and oscillator signals to produce IF.
- - Local Oscillator: Generates frequency for frequency conversion.
- - IF Filter & Amplifier: Selects and amplifies the IF signal.
- - Demodulator: Extracts baseband signal from IF.
- - Audio Amplifier: Amplifies demodulated signal.
- - Speaker: Converts signal to sound.

7 a. Define Modulation. Identify any five differences between Frequency Modulation and Amplitude Modulation.

Modulation is the process of varying one or more properties of a carrier signal (such as amplitude, frequency, or phase) in accordance with the information signal to be transmitted.

Differences between Frequency Modulation (FM) and Amplitude Modulation (AM):

S.No	Frequency Modulation (FM)	Amplitude Modulation (AM)
1	Frequency of carrier varies with the message signal.	Amplitude of carrier varies with the message signal.
2	More immune to noise and interference.	More affected by noise and interference.
3	Complex and costly receivers.	Simple and low-cost receivers.
4	Bandwidth is higher than AM.	Requires less bandwidth.
5	Used for high-fidelity applications like FM radio.	Used for broadcasting like AM radio.

b. Why Pre-emphasis and De-emphasis are required? Explain how they are implemented.

Pre-emphasis and de-emphasis are techniques used in FM systems to improve the signal-to-noise ratio for high-frequency components.

High-frequency signals are more affected by noise. To counter this, pre-emphasis boosts the amplitude of high-frequency components at the transmitter, and de-emphasis attenuates them at the receiver to restore the original signal.

Implementation:

- Pre-emphasis is done using a high-pass filter before modulation.
- De-emphasis is done using a low-pass filter after demodulation.

c. Draw the block diagram of a superheterodyne receiver and explain the function of each.

A superheterodyne receiver converts the received RF signal to a fixed intermediate frequency (IF) to simplify filtering and amplification.

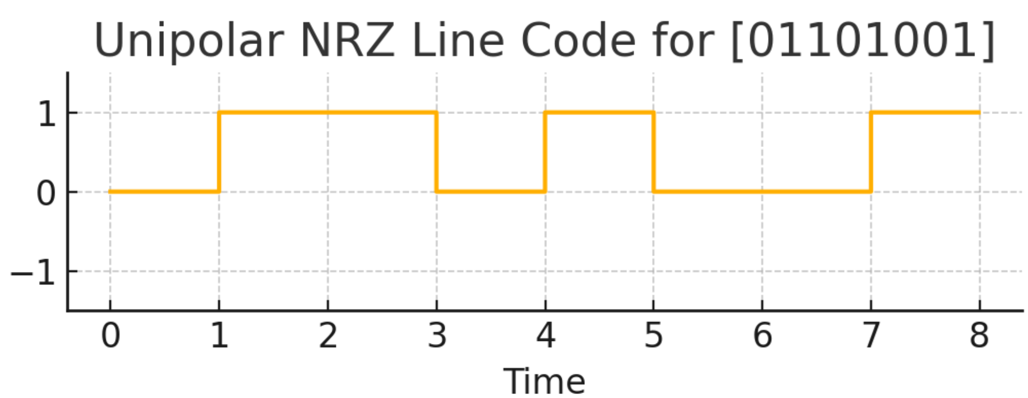
Block Diagram and Functions:

1. Antenna – Captures the RF signal.
2. RF Amplifier – Amplifies weak RF signals.
3. Mixer – Mixes RF signal with local oscillator to produce IF.
4. Local Oscillator – Generates a frequency that mixes with RF.
5. IF Amplifier – Amplifies intermediate frequency signal.
6. Detector/Demodulator – Extracts audio or baseband signal.
7. Audio Amplifier – Amplifies audio signal for output.
8. Speaker – Converts audio signal to sound.

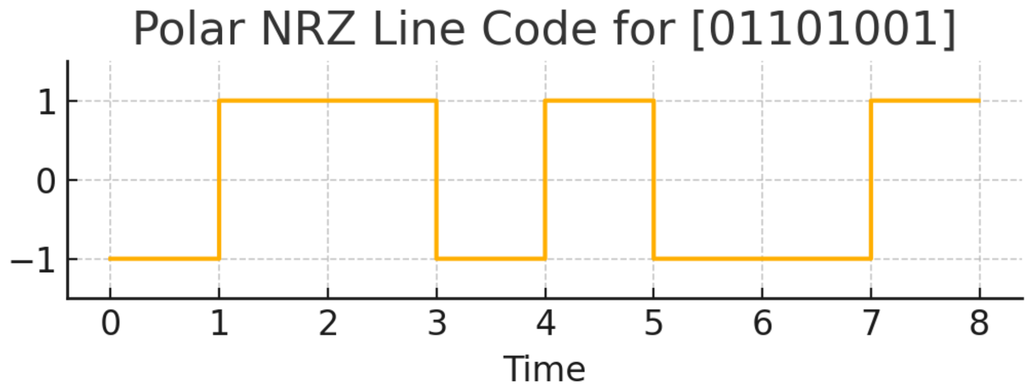
8. Basic Elements of PCM System

1. Sampler: Converts the analog signal into a series of samples at regular intervals.
2. Quantizer: Converts each sample to the nearest value from a finite set of levels.
3. Encoder: Assigns a binary code to each quantized level.
4. Transmission: The encoded binary data is transmitted over the channel.
5. Regenerator: Regenerates the clean digital signal.
6. Decoder: Reconstructs the quantized signal from binary data.
7. Reconstruction Filter: Converts the discrete signal back into an analog signal.

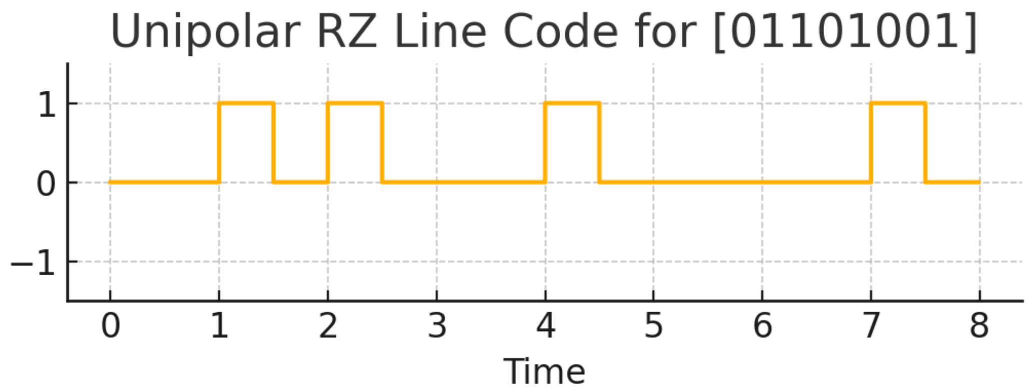
Unipolar NRZ



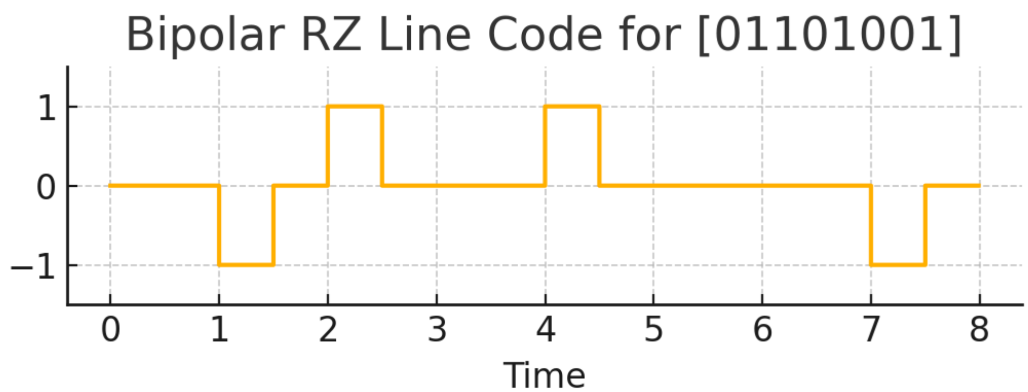
Polar NRZ



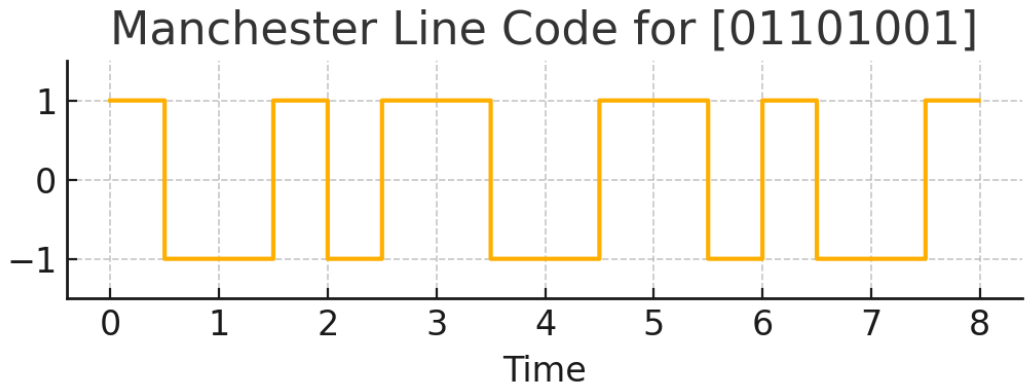
Unipolar RZ



Bipolar RZ



Manchester



Module – 5

Q.9	~~a.~~	What is Intersymbol Interference (ISI)? With a neat block diagram outline the baseband binary data transmission system and write the necessary equations?	8	L2	CO4

One source of bit errors in a baseband-pulse transmission system that we wish to study is
intersymbol interference (ISI), which arises when the communication channel is dispersive.

• Dispersive channel :the channel has a frequency dependent amplitude spectrum.

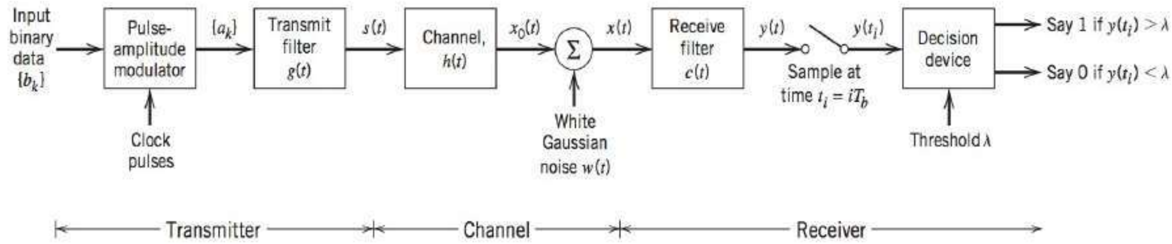


FIGURE Baseband binary data transmission system.

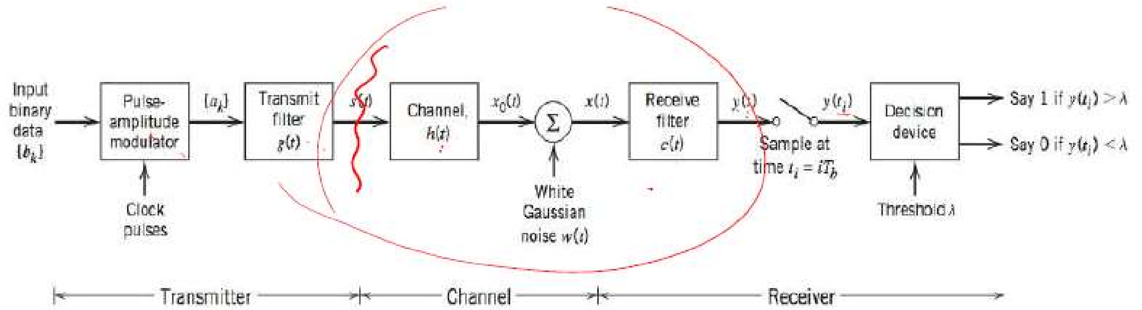


FIGURE Baseband binary data transmission system.

Consider a baseband PAM System.

Incoming binary sequences $\{b_k\}$ consists of symbols 1 and 0 each of duration T_b.

The pulse amplitude modulator transforms this binary sequence into new sequence of short pulses with amplitudes $\{a_k\}$ in polar form.

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

The sequence of short pulses are introduced to the transmit filter $g(t)$ resulting in $s(t)$.

$$s(t) = \sum_{k=-\infty}^{\infty} a_k \, g(t - kT_b) \quad \text{——— (2)}$$

From the block diagram

$$y(t) = s(t) * h(t) * c(t) \quad \text{——— (3)}$$

- The scaled pulse μ $p(t)$ is obtained by a <u>double convolution</u> 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, as shown by

$$\mu p(t) = g(t) * \underline{h(t)} * c(t)$$ | Pulse function.

Frequency Domain → $\mu P(f) = G(f) \cdot H(f) \cdot C(f)$

→ $$y(t) = \sum_{k=-\infty}^{\infty} a_k \, \mu p(t - kT_b) + n(t) \quad \text{——— (4)}$$

| $n(t)$ is the noise

μ - scaling factor.

we assume $p(0) = 1$ for normalization.

$$y(t) = \sum_{k=-\infty}^{\infty} a_k \, \mu \, p(t - kT_b) + n(t)$$

For sampling, take $t = t_i$ at the switch

$t_i = i T_b$

Eqn ④ $\rightarrow$ $$y(t_i) = \mu_i \sum_{k=-\infty}^{\infty} a_k p(iT_b - kT_b) + n(t_i)$$

$$\rightarrow y(t_i) = \mu_i \sum_{k=-\infty}^{\infty} a_k p((i-k)T_b) + n(t_i) \quad \text{— ⑤}$$

Considering a_i for $i=k$,

$$y(t_i) = \underbrace{\mu_i a_i}_{\text{required term}} + \underbrace{\sum_{\substack{k=-\infty \\ i \neq k}}^{\infty} \mu a_k p((i-k).T_b)}_{ISI} + \underbrace{n(t_i)}_{\text{noise}} \quad \text{— ⑥}$$

$$y(t_i) = \mu_i a_i + \mu_i \sum_{\substack{k=-\infty \\ k \neq i}}^{\infty} a_k p((i-k)T_b) + n(t_i)$$

Standard Equation of baseband transmission

→ The first term μa_i represents contribution of i^{th} transmitted bit

→ The second term represents the residual effect of other transmitted bits on the decoding of i^{th} bit.

→ The residual effect due to the occurrence of pulses before and after the sampling time instant t_i is called inter symbol interference (ISI)

		equations?			
	b.	Define SNR. Summarize the different types of external and internal noise.	7	L2	CO4

- The *signal-to-noise (S/N) ratio,* also designated SNR, indicates the relative strengths of the signal and the noise in a communication system.
- The stronger the signal and the weaker the noise, the higher the *S/N* ratio. If the signal is weak and the noise is strong, the *S/N* ratio will be low and reception will be unreliable.
- Communication equipment is designed to produce the highest feasible *S/N* ratio.

External Noise

• External noise comes from sources over which we have little or no control—industrial, atmospheric, or space.

• Regardless of its source, noise shows up as a random ac voltage and can be seen on an oscilloscope.
• The amplitude varies over a wide range, as does the frequency.
• The noise in general contains all frequencies, varying randomly. This is generally known as white noise.
• Atmospheric noise and space noise are a fact of life and simply cannot be eliminated.
• Some industrial noise can be controlled at the source, but because there are so many sources of this type of noise, there is no way to eliminate it.
• The key to reliable communication, then, is simply to generate signals at a high enough power to overcome external noise

Industrial Noise: Industrial noise is produced by manufactured equipment, such as automotiveignition systems, electric motors, and generators. Any electrical equipment that causes high voltages or currents to be switched produces transients that create noise. Noise pulses of large amplitude occur whenever a motor or other inductive device is turned on or off.

• Atmospheric Noise: The electrical disturbances that occur naturally in the earth's atmosphere are another source of noise. Atmospheric noise is often referred to as static. Static usually comes from lightning, the electric discharges that occur between clouds or between the earth and clouds.
• Extraterrestrial Noise. Extraterrestrial noise, solar and cosmic, comes from sources in space. One of the primary sources of extraterrestrial noise is the sun, which radiates a wide range of signals in a broad noise spectrum. The noise intensity produced by the sun varies with time. In fact, the sun has a repeatable 11-year noise cycle. During the peak of the cycle, the sun produces an awesome amount of noise that causes tremendous radio signal interference and makes many frequencies
unusable for communication.

Internal Noise

• Electronic components in a receiver such as resistors, diodes, and transistors are major sources of internal noise.
• Internal noise, although it is low level, is often great enough to interfere with weak signals.
• The main sources of internal noise in a receiver are thermal noise, semiconductor noise, and intermodulation distortion.
• Since the sources of internal noise are well known, there is some design control over this type of Noise

c.	Illustrate the concept of Noise in cascaded stages with a diagram. Write Friis formula and mention its terms.	5	L2	CO4

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

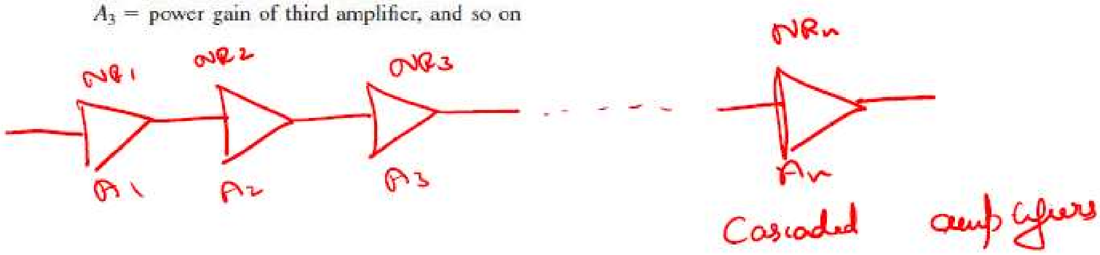
NR_2 = noise ratio of second amplifier

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



Q.10	a.	What is Baseband digital transmission? Explain the following concepts briefly : (i) Nyquist criterion for distortionless transmission. (ii) Baseband M-ary PAM transmission.	8	L2	CO4

Baseband transmission of digital data therefore requires the use of a low-pass channel with a bandwidth large enough to accommodate the essential frequency content of the data stream.

• In general, channel is dispersive with its frequency response deviates from that of an ideal low-pass filter.

• The result of data transmission over such a channel is that each received pulse is affected somewhat by adjacent pulses, thereby giving rise to a common form of interference called intersymbol interference (ISI).

• To correct for it, control has to be exercised over the pulse shape in the overall system.

• Another source of bit errors in a baseband data transmission system is the ubiquitous receiver noise (channel noise).

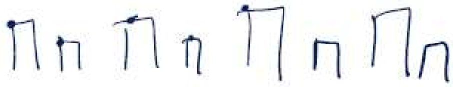
NYQUIST'S CRITERION FOR DISTORTIONLESS TRANSMISSION

- The receiver obtains the transfer function of channel by *extracting* and then *decoding* the corresponding sequence of coefficients
- $\{ak\}$, from the output $y[t]$.
- The *extraction* involves sampling the output $y(t)$ at time $t_i = iTb$.
- The *decoding* requires that the weighted pulse contribution $a_k p(iTb - kTb)$ for $k = i$ *be free* from ISI due to the overlapping tails of all other weighted pulse contributions represented by $k \neq i$.
- This, in turn, requires that we *control* the overall pulse $p(t)$, as shown by

$$p(iT_b - kT_b) = \begin{cases} 1, & i = k \\ 0, & i \neq k \end{cases} \qquad 1$$

- the receiver output $y(t_i)$ will be

$$y(t_i) = \mu a_i \qquad \text{for all } i$$



mT_b

which implies zero intersymbol interference.

Hence, the condition $p(iT_b - kT_b) = \begin{cases} 1, & i = k \\ 0, & i \neq k \end{cases}$ ensures *perfect reception in the absence of noise.*

$$\rightarrow p(t - mT_b) = \sum p(mT_b)\, \delta(t - mT_b) \quad \text{--- (2)}$$

- Consider then the sequence of samples $\{ p(nT_b) \}$, where $n = 0, \pm 1, \pm 2, \ldots\ldots$
- We know that sampling in the time domain produces periodicity in the frequency domain which implies

$$P_\delta(f) = R_b \sum_{n=-\infty}^{\infty} P(f - nR_b) \quad \text{--- (3)} \qquad R_b = \frac{1}{T_b}$$

- $R_b = 1/T_b$ is the *bit rate* in bits per second (b/s); $P_\delta(f)$ is the Fourier transform of an infinite periodic sequence of delta functions of period T_b, whose areas are weighted by the respective sample values of $p(t)$.

We know that $g_\delta(t) = \sum_{n=-\infty}^{\infty} g(t)\, \delta(t - nT)$

$$g_\delta(t) \overset{F.T}{\rightleftharpoons} f_s \sum_{n=-\infty}^{\infty} G(f - n f_s) \qquad f_s = \frac{1}{T_s}$$

$$p_\delta(t) = \sum p(mT_b)\, \delta(t - mT_b) \quad \text{--- (4)}$$

F.T of eqn (4)

$$\rightarrow P(f) = \frac{1}{T_b} \sum P\left(f - \frac{m}{T_b}\right) \qquad \frac{1}{T_b} = R_b \text{ bit rate}$$

$$P(f) = R_b \sum P(f - mR_b) \quad \text{--- (5)}$$

Consider $m = i - k \Rightarrow P_\delta(f) = R_b \sum P(f - (i-k) R_b)$

At an instant, $i = k$ $\quad P(f) = R_b \sum P(f) \quad \text{--- (6)}$

$$\sum P(f - mR_b) = 1/R_b$$

$$\boxed{\sum_{m=-\infty}^{\infty} P(f - mR_b) = T_b}$$

: Nyquist criterion for Distortionless channel.

→ The frequency function $P(f)$ eliminates intersymbol interference for samples taken at intervals T_b provided that it satisfies below equation $\sum_{n=-\infty}^{\infty} P(f - nR_b) = T_b$.

BASE BAND M-ARY PAM TRANSMISSION

- In a *baseband M-ary PAM system,* the pulse-amplitude modulator produces one of *M* possible amplitude levels
- In an 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 output corresponds to a distinct symbol, so that there are M distinct amplitude levels to be transmitted.
- Consider then an M-ary PAM system with a signal alphabet that contains M equally likely and statistically independent symbols, with the symbol duration denoted by *T* seconds
- *1 /T* is the *signaling rate* of the system, which is expressed in *symbols per second* or *bauds.*

M= 2 → Binary 0, 1
M = 3 → Ternary 00, 01, 10
M = 4 → Quaternary 00 01 11 10

b.	Define Noise. Classify the different types of semiconductor noise.	7	L2	CO4

Noise

- *Noise* is an electronic signal that is a mixture of many random frequencies at many amplitudes that gets added to a radio or information signal as it is transmitted from one place to another or as it is processed.
- Noise is *not* the same as interference from other information signals.
- The noise level in a system is proportional to temperature and bandwidth, and to the amount of current flowing in a component, the gain of the circuit, and the resistance of the circuit.
- Increasing any of these factors increases noise.
- Therefore, **low noise is best obtained by using low-gain circuits, low direct current, low resistance values, and narrow bandwidths.**

Semiconductor Noise

- Electronic components such as diodes and transistors are major contributors of noise.
- In addition to thermal noise, semiconductors **produce shot noise, transit-time noise, and flicker noise.**
- The most common type of ***semiconductor noise*** is ***shot noise.***
- Current flow in any device is not direct and linear. The current carriers, electrons or holes, sometimes take random paths from source to destination, whether the destination is an output element, tube plate, or collector or drain in a transistor.
- It is this random movement that produces the shot effect. Shot noise is also produced by the random movement of electrons or holes across a PN junction.
- Even though current flow is established by external bias voltages, some random movement of electrons or holes will occur as a result of discontinuities in the device

c.	What is Noise Factor and Noise Figure? An RF amplifier has an S/N ratio of 8 at the input and an S/N ratio of 6 at the output. Calculate the Noise factor and Noise figure.	5	L2	CO4

$SNR_{in} = 8$

$SNR_{out} = 6$

NR = Noise ratio $= 8/6 = \underline{\underline{1.33}}$

Noise figure $NF = 10 \log NR = \underline{\underline{1.23 \text{ dB}}}$

- **Noise Factor and Noise Figure.** The noise factor is the ratio of the S/N power at the input to the S/N power at the output.
- The device under consideration can be the entire receiver or a single amplifier stage.

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

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

$$NF = 10 \log NR \quad \text{dB}$$